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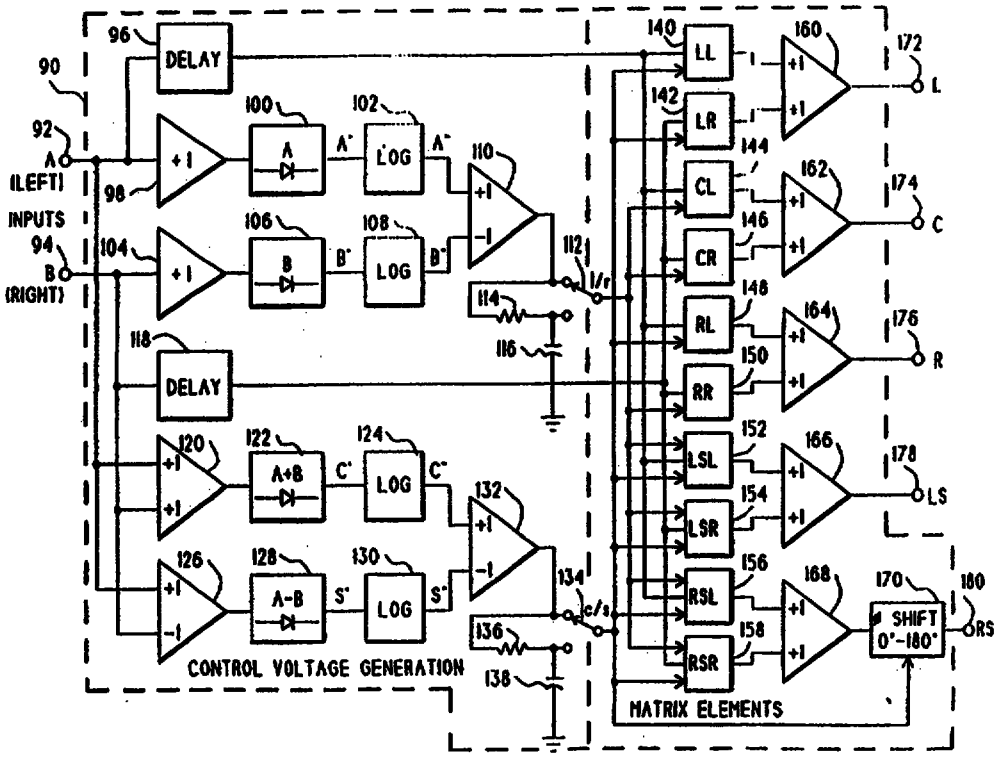
(54) Title: 5-2-5 MATRIX ENCODER AND DECODER SYSTEM

(57) Abstract

A sound reproduction system for converting stereo signals on two input channels (92, 94), at least one signal component being directionally encoded and correlated and at least one signal component that is not directionally encoded and uncorrelated in the two input channels, into signals for several ouput channels, including decoding apparatus (90) for enhancing the correlated component of the input signals in the desired direction and reducing the strength of such signals in channels not associated with the encoded direction, while preserving the separation between the respective left and right ouput channels (172, 176) and the total energy of the uncorrelated component of the input channels in each output channel, such that instruments recorded on the right input channel stay on the right side of the output channels and the instruments recorded on the left stay on the left side, and the apparent loudness of all the intruments in all the output channels stays the same regardless of the direction of the directionally encoded component of the input signals, and encoding means to encode five input channels so they will encode with correct direction and level in decoders according to the invention, and in decoders according to the current film standard.



5-2-5 MATRIX ENCODER AND DECODER SYSTEM

Cross-Reference to Related Applications

This application is based upon the U.S. Provisional Patent Application No. 60/058,169, entitled "5-2-5 Matrix Encoder and Decoder System," filed September 5, 1997, which is a continuation-in-part of U. S. Patent Application No. 08/742,460 entitled "Multichannel Active Matrix Encoder and Decoder with Maximum Lateral Separation," filed November 1, 1996, which is a continuation-in-part of U.S. Patent Application No. 08/684,948, entitled "Multichannel Active Matrix Sound Reproduction with Maximum Lateral Separation," filed July 19, 1996.

Field of the Invention

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.

More particularly, the invention concerns an improved set of design criteria and their solution to create a decoding matrix having optimum psychoacoustic performance, with high separation between left and right components of the stereo signals while maintaining non-directionally encoded components at a constant acoustic level regardless of the direction of directionally encoded components of the input audio signals.

Additionally, this invention relates to the encoding of multi-channel sound onto two channels for reproduction by decoders according to the invention. In particular, it relates to improved matrixing coefficients for a 5-2-5 matrix encoder and decoder system.

Background of the Invention

Apparatus for decoding a stereophonic pair of left and right input audio signals into a multiplicity of output signals is commonly referred to as a surround sound decoder or processor. Surround sound decoders work by combining the left and right input audio signals in different proportions to produce the multiplicity N

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of output signals. The various combinations of the input audio signals may be mathematically described in terms of a N row by 2 column matrix, in which there are 2N coefficients each relating the proportion of either left or right input audio signals contained in a particular output signal.

The matrix coefficients may be fixed, in which case the matrix is called passive, or they may vary in time in a manner defined by one or more control signals, in which case the matrix is described as active. The coefficients in a decoding matrix may be real or complex. Complex coefficients in practice involve the use of precise phase quadrature networks, which are expensive, and therefore most recent surround sound decoders do not include them, so that all of the matrix coefficients are real. In the bulk of the work described in this patent application, the matrix elements are also real. Real coefficients are inexpensive and will optimally decode a five channel film encoded with the active encoder described in this patent.

However, real coefficients are not optimal when decoding a film encoded from a five channel original using a passive encoder such as the one described in this application, and are also not optimal when decoding a film made with the standard four channel encoder of the prior art. A modification to the decoder design which will optimally decode such films is also described. Although the description is of a phase corrector to the inputs of the decoder, the correction could also be accomplished by making the matrix elements complex.

In a passive matrix, which is defined as a matrix in which the coefficients are constant, such as the Dolby Surround matrix, several ideal properties are achieved by suitable choice of the coefficients. These properties include the following:

Signals encoded with a standard encoder will be reproduced by a passive matrix decoder with equal loudness regardless of their encoded direction.

Signals where there is no specific encoded direction, such as music that has been recorded so that the two inputs to the decoder have no correlation, that is, decorrelated signals, will be reproduced with equal loudness in all output

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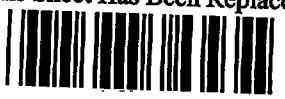

channels.

When the input signals are a combination of a directionally encoded component and a decorrelated component there is no change in either the loudness or the apparent separation of the decorrelated component as the encoded direction of the directionally encoded component changes.

A disadvantage of passive decoders is that the separation of both directional and decorrelated components of the input signals is not optimal. For example, a signal intended to come from front center is also reproduced in the left and right front output channels usually with a level difference of only 3dB. Therefore, most modern decoders employ some variation of the matrix coefficients with the apparent direction of the predominant sound source, that is, they are active rather than passive.

In the original Dolby Surround decoder format, only one rear channel output is provided, which typically is reproduced on more than one loudspeaker, all such loudspeakers being driven in parallel, so that there is no left-right separation in the rear channels. However, there is high separation between signals that are encoded in opposite directions.

Previous patents have described many aspects of active matrix surround sound decoders for conversion of a stereophonic audio signal pair into multiple output signals. The prior art describes how the apparent direction of a directionally encoded signal component can be determined from the logarithm of the ratio of the amplitudes of the component in the left and right channels of the stereophonic pair, along with the logarithm of the ratio between the sum of these amplitudes and the difference therebetween. This art will be assumed in this patent application, along with a great deal of art which pertains to smoothing the directional control signals thus or otherwise derived. We assume that these two directional control signals exist in a usable form. For the purposes of this invention, these directional control signals can be possibly derived from directional information recorded on a subchannel of a digital audio signal.

This invention concerns the use to which these directional control signals

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are put in controlling an active matrix which takes the signals on the two inputs and distributes them to a number of output channels in appropriately varying proportions dependent upon the directional control signals.

A simple example of such a matrix is given by Scheiber in U. S. Patent No. 3,959,590. Another matrix in common use is that of Mandell, described in U. S. Patent No. 5,046,098. A matrix with four outputs is described in detail in Griesinger, U.S. Patent No. 4,862,502, and a complete mathematical description of this matrix, along with a mathematical description of a six output matrix, is given in Griesinger, U. S. Patent No. 5,136,650. A different six output matrix is described in Fosgate, U. S. Patent No. 5,307,415. All of these prior matrices distribute the input audio signals among the various outputs under control of the directional control signals as described above.

Each of these matrices is constructed somewhat differently, but in each case each output is formed by a sum of the two input signals, each input signal having been first multiplied by a coefficient. Thus each matrix in the prior art can be completely specified by knowing the value of two coefficients for each output and how these coefficients vary as a function of the directional control signals which provide directional information as described above. These two coefficients are the matrix elements of a N by 2 matrix, where N is the number of output channels, which completely specifies the character of the decoder. In most prior art these matrix elements are not explicitly stated, but can be inferred from the descriptions given. In a particular embodiment they can also be easily measured.

Griesinger, U. S. Patent No. 5,136,650, issued August 4, 1992, gives the complete functional dependence of each matrix element on the directional control signals.

Since the above-referenced Griesinger patent issued, the film industry has developed a "five plus one" discrete sound standard. Many theater movie releases and some home releases are made with soundtracks comprising five separate full bandwidth audio channels, namely center, left front, right front, left rear, and right rear, with a reduced bandwidth sixth audio channel intended for very low

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frequency effects. Reproduction of such soundtracks requires special digital hardware to demultiplex and decompress the audio tracks into the 5+1 output channels. However, there is a very large selection of previously released film prints and videos which employ a two channel soundtrack matrix encoded format, both analog and digital. Such soundtracks are encoded during the mixing process using a standardized four channel to two channel encoder.

While earlier work by Griesinger and others has described the outputs of the decoder in terms of a complicated sum of various signals: the input signals, their sum and their difference, and the same four signals after passing through variable gain amplifiers controlled by the directional control signals, it is possible to collect the terms of each output that are related to a particular input and thereby to describe the matrix completely in closed form, so that the decoder can be realized either in digital or analog hardware components.

In a standard film decoder, a boost is applied to the front channels when a strongly steered signal such as dialog is present. This upsets the balance between such signals and background effects or music, relative to the balance between such signals in the discrete 5 channel movie theater system. An improved active encoder described herein is needed to correct the balance between the strongly steered front signals and music. There is also a need to improve both encoder and decoder performance in regard to left side and right side signals. A further improvement in the decoder is to limit the effects of abrupt changes in the directional control signals to provide better dynamic response to rapid changes therein.

In addition, the present invention constitutes further improvements to the decoder of the previous Griesinger U. S. Patent Applications referenced above.

Summary of the Invention

The present invention is concerned with realization of the active matrix having certain properties which optimize its psychoacoustic performance.

The invention is a surround sound decoder having variable matrix values so constructed as to reduce directionally encoded audio components in outputs which

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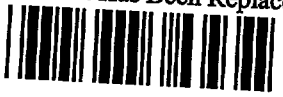

are not directly involved in reproducing them in the intended direction; enhance directionally encoded audio components in the outputs which are directly involved in reproducing them in the intended direction so as to maintain constant total power for such signals; while preserving high separation between 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.

In a preferred embodiment, a surround sound decoder is provided for redistributing a pair of left and right audio input signals including directionally encoded and non-directional components into a plurality of output channels for reproduction through loudspeakers surrounding a listening area, and incorporating circuitry for determining the directional content of the left and right audio signals and generating therefrom at least a left-right steering signal and center-surround steering signal.

The decoder includes delay circuitry for delaying each of the left and right audio input signals to provide delayed left and right audio signals; a plurality of multipliers equal to twice the number of output channels, organized in pairs, a first element of each pair receiving the delayed left audio signal and a second element receiving the delayed right audio signal, each of the multipliers multiplying its input audio signal by a variable matrix coefficient to provide an output signal; the variable matrix coefficient being controlled by one or both of the steering signals. A plurality of summing devices are provided, one for each of the plurality of output channels, with each of the summers receiving the output signals of a pair of the multipliers and producing at its output one of the plurality of output signals. The decoder has the 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; and so constructed to enhance directionally encoded audio components in the outputs which are directly involved in reproducing them in the intended direction so as to maintain constant

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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 so constructed to maintain 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.

This invention also includes improved active encoder embodiments which correct the balance between strongly steered front signals and decorrelated music signals due to the boost of front signals which occurs in a standard film decoder, and which also increase the separation between encoder outputs when uncorrelated left and right side inputs are presented to the encoder. It also encompasses modified performance in the film decoder specifications with regard to left or right side encoded signals. A further improvement in the decoder relates to the effects of abrupt changes in the directional control signals and limits the more slowly changing signal to provide better dynamic response to the rapidly changing signal.

Although the invention is primarily described in terms of analog embodiments, an advantage of the invention is that it can be implemented as a digital signal processor.

An advantage of the present invention is that the design of the decoding matrix provides high left to right separation in all output channels.

A further advantage of the invention is that it maintains this high separation regardless of the direction of the dominant encoded signal.

Another advantage of the invention is that the total output energy level of any non-encoded decorrelated signal remains constant regardless of the direction of the dominant encoded signal.

Another advantage of the invention is that it can reproduce conventionally encoded soundtracks in a way which closely matches the sound of a 5+1 channel discrete soundtrack release.

Yet another advantage of the invention is that it provides a simple passive

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matrix encoding into two channels of a five channel soundtrack that will decode into five or more channels with very little subjective difference from the five channel original.

Another advantage of the invention is that it provides an active encoder which has better performance in respect to the left and right surround inputs than that achievable with a passive five channel encoder.

While the decoder of the invention operates optimally when the active five channel encoder, another advantage of the invention is that with an added phase correction network it can also optimally reproduce movie soundtracks encoded with either the standard four channel passive encoder of the prior art or the five channel passive matrix encoder which is an aspect of the present invention.

An advantage of the active matrix encoder of the invention is that it provides dynamic control of the balance between strongly steered front signals and non-directional music to compensate for the boost applied to such steered signals in standard film decoders.

A further advantage of the encoder is that it provides improved separation of simultaneous left side and right side signals when decoded with a standard film decoder.

An advantage of the decoder of the invention is that it provides more of a level change in the front loudspeakers relative to the rear when a signal ispanned on either side of the listener, improving the apparent motion of such signal sources.

Another advantage of the decoder according to this invention is to limit the absolute value of one of the two steering signals when the other is rapidly changing, so that dynamic effects are better reproduced.

More particularly, the present invention is concerned with improvements to the derivation of suitable variable matrix coefficients as previously disclosed in Griesinger's U.S. Patent Nos. 4,862,502 (1989), 5,136,650 (1992), the July 1996 Griesinger U.S. Patent Application No. 08/684,948, and the November 1996 Griesinger U.S. Patent Application No. 08/742,460, as disclosed in the Provisional

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Patent Application filed September 1997 . The previously used coefficients were implemented in a decoder referred to here as version 1.11. The present invention includes two principal changes to the coefficients derived in the previous patent application No. 08/684,948 of July 19, 1996. The first is a change to the "TV matrix" correction in the rear channels. There is no change when the steering is in the rear direction, but it was found that the results were better if the 3dB reduction in rear output level was maintained as steering went forward. The output level rises back to the original level when the absolute value of the control signal $|lr|$ rises from zero to 22.5 degrees, but is independent of the value of the control signal $|cs|$. This change is described in the section on the TV matrix correction, and is illustrated in the revised figures for LRL and LRR. An advantage claimed for this revision is that there is less variation of the relative sound level of the rear channels when steering occurs in the forward direction, providing a more natural and smoother decoding.

The second change is in the treatment of the front channels and the center channel. Extensive listening to the decoder version 1.11 showed that it is necessary to consider the power in all channels in determining the treatment of the center channel. The mathematics in the section on the center channel in the previous (1996) patent application has been modified to reflect this change. The assumption is that the ratio of the power in the center input channel of the encoder to the power in the other channels should be preserved in the total acoustic power at the output of the decoder. The changes in the center channel require a different strategy for implementing the LRL matrix element. The strategy given here is mathematically elegant, but involves a divide. An implementation of this division uses a two dimensional look-up table instead.

Specifically, this invention comprises the following improvements to the coefficient values derived in the previous three patents and applications.

Firstly, adding interpolation to the LRL matrix element in the left rear quadrant near the cs=0 boundary.

Secondly, correcting a software error in the LRR matrix element in the left

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rear quadrant along the *lr*=0 boundary.

Thirdly, adding complexity to the LFL and LFR matrix elements in the left rear quadrant, such that an input steered to the left rear output is eliminated from the left front output.

Fourthly, repairing the LFL and LFR matrix elements so they follow the curve of sin and cos along the boundary from left to center, and from right to center, while retaining a boost along the *lr*=0 axis.

Fifthly, recalculating the mathematical analysis for the LFL and LFR elements in the front left quadrant. The object of the redesign (which goes beyond the 1991 patent) is to make the sum of the squares of the elements equal to 1 along the *cs*=0 axis. An advantage of this invention is to reduce unwanted variations of total power as a result of steering.

Sixthly, including a redesigned center channel boost function, which increases at a lower rate than the one in decoder version 1.11. In this aspect of the invention, an advantage is that the center boost function has been chosen carefully on the basis of listening tests to give a minimal sense of motion of vocals or dialog between the left and right main speakers and the center speaker, while maximizing the left – right separation of instruments which are present along with the vocals.

Seventhly, adding a special function CF which replaces the previous boost in LFR along the *lr*=0 axis with a cut, the cut designed to preserve in the sum of the powers from the outputs of the decoder the ratio of the power of the center component of signals to the encoder to the total power of signals to the encoder. An advantage of this aspect of the invention is that this procedure makes vocals in music, and dialog in films, have the identical balance in the decoded environment that they did in the material before encoding. This procedure also preserves the balance in recordings which were originally mixed for two channel playback. The new function CF remains close to zero – that is there is no subtraction of the right input to the decoder from the left input of the decoder when forming the left front output, and this low value is maintained until *cs*

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reaches about 30 degrees toward the front. As the control signal *cs* increases over this range the center channel level rises rapidly at first to a value about 3dB lower than the value for Dolby Pro-Logic, and then holds constant. As *cs* rises beyond 30 degrees, the center level rises rapidly to the same maximum used for Dolby Pro-Logic. The CF function also decreases rapidly over this range, increasing the subtraction, and removing the center component from the left and right front outputs. The value of CF also drops rapidly to the previous value when the absolute value of control signal *lr* approaches the boundary.

Eighthly, incorporating a panning correction to the new center matrix elements which corrects the level along the boundaries. This aspect of the invention conveys an advantage in reducing level fluctuations during steering in these directions.

The principal advantage of this invention is a reduction in the variations of various directional signals in the presence of strong steering, especially in the rear signals when steering is to the front, and in center signals when steering is in other directions. This is seen particularly in the corrections to TV matrix decoding.

An additional advantage of this invention is that it provides a smoother and more transparent reproduction of the surround sound effects without unwanted variations of the total acoustic output of center front signals due to steering activity.

Another advantage of this invention is to more accurately balance the levels of vocals in music and dialog in films with respect to the non-directional sounds so that the balance is identical in the decoded environment to that in the material before encoding.

Another advantage of the invention is to preserve the balance in recordings that were originally mixed for two channel playback.

Brief Description of the Drawings

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

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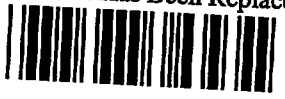

advantages thereof, will best be understood by reference to the following detailed description of an illustrative embodiment when read in conjunction with the accompanying drawing figures, wherein:

FIG. 1 is a block schematic of a passive matrix Dolby surround decoder according to the prior art;

FIG. 2 is a block schematic of a standard Dolby matrix encoder according to the prior art;

FIG. 3 is a block schematic of a five channel encoder for producing Dolby matrix compatible encoding of discrete five channel soundtracks according to the present invention;

FIG. 4 is a block schematic of a five channel embodiment of the decoder according to the invention;

FIGs. 5a and 5b show detailed schematics for a typical phase shifter that may be used in the circuit of FIG. 4;

FIGs. 6a-6e show the relationships between various signals in the decoder of FIG. 4;

FIG. 7 shows a block schematic of an active encoder according to the invention;

FIG. 8 shows a phase sensitive detection circuit for generation of an ls/rs signal for use with the phase correction circuit of FIG. 9;

FIG. 9 shows an input phase correction circuit to be applied ahead of the decoder of FIG. 4 for optimal decoding of passively encoded movie soundtracksincluding a graph showing the relationship between the control signal ls/rs and the steering angle Θ_{LS};

FIG. 10 shows a block schematic of a simplified active encoder according to the invention, also including a graph of the steering angle Θ_{RS} against the control signal rs/ls;

FIG. 11 shows a block schematic of an active matrix encoder having amplitude compensation for strongly steered front signals and better separation

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for simultaneous side inputs, according to the invention;

FIGs. 12a-12c show graphically the variation of the GL, GC and GR signals for front quadrant steering and of the left-left (LL) and left-right (LR) matrix elements as steering goes from left to left side in the encoder of FIG. 11; and

FIG. 13 shows graphically the maximum permissible values of each of control signals l/r and c/s as the other changes, for signals steered between left and center, as applied to the decoder of FIG. 4 or the seven channel variant thereof.

FIG. 14 is a perspective graphical view showing the value of the left rear left (LRL) matrix element in decoder version 1.11 of the general type shown in FIG. 4, illustrating a discontinuity near the left vertex;

FIG. 15 similarly shows the left rear right (LRR) matrix element of decoder version 1.11, illustrating a discontinuity in the rear along the lr=0 axis;

FIG. 16 shows in perspective graphical view the LRL matrix element as it was intended to be in decoder version 1.11, as contrasted with the flawed matrix element in FIG. 14 which was actually implemented;

FIG. 17 similarly shows the left front left (LFL) matrix element from U.S.Patent No. 4,862,502 and Dolby Pro-Logic, scaled so the maximum value is one;

FIG. 18 similarly shows the left front right (LFR) matrix element from U.S.Patent No. 4,862,502 and Dolby Pro-Logic, scaled by .71 so the minimum and maximum values are ±0.5;

FIG. 19 graphically represents in perspective the square root of the sum of the squares of LFL and LFR from U.S. Patent No. 4,862,502, scaled so the maximum value is one, showing that the value is constant at 0.71 along the axis from unsteered to right, while the unsteered to left rises 3dB to the value 1, and the unsteered to center or to rear falls by 3dB to the value 0.5, in which graph the rear direction profile is identical to that of the center direction;

FIG. 20 similarly represents the square root of the sum of the LFL and LFR matrix elements from the previous U.S. Patent Application No. 08/742,460, scaled

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so the maximum value is 1, illustrating the constant value of .71 in the entire right half of the plane, and the gentle rise to one toward the left vertex;

FIG. 21 shows in perspective graphically the left front left (LFL) matrix element in the decoder version 1.11, illustrating that the boost as the steering moves toward center is applied both along the *lr*=0 axis, and along the left to center boundary, and also the reduction in level as the steering moves to the rear;

FIG. 22 similarly shows the left front left matrix (LFL) element having the correct amplitude along the left to center boundary, as well as along the center to right boundary;

FIG. 23 is a graph showing the behavior of LFL and LFR along the rear boundary between left and full rear, where the slight glitch is due to the absence of a point at 22.5 degrees.

FIG. 24 shows in graphical perspective the left front left (LFL) matrix element as viewed from the left rear, illustrating the large correction along the left-rear boundary, which causes the front left output to go to zero when steering goes from left to left rear, while the output remains zero as the steering progresses to full rear, but along the *lr*=0 axis and in the right rear quadrant the function is identical to the Dolby matrix;

FIG. 25 shows similarly the left front right (LFR) matrix element, where the large peak in the left to rear boundary works in conjunction with the LFL matrix element to reduce the front output to zero along this boundary as steering goes from left rear to full rear, and 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;

FIG. 26 illustrates the root mean squared sum of LFL and LFR, according to the present invention;

FIG. 27 shows the square root of the sum of the squares of LFL and LFR including the correction to the rear level, viewed from the left rear, showing that 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, and showing the peak along the middle to center axis;

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FIG. 28 is a graph showing as a solid curve the center matrix value as a function of CS in dB, assuming sound power ratios identical to stereo, and using Dolby matrix elements with 3dB less power in the rear than typically used, and as a dotted curve the actual value of the center matrix elements in Pro-Logic, illustrating 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;

FIG. 29 similarly shows as a solid curve the value of the center matrix elements assuming equal power ratios to stereo, given the matrix elements and calibration actually used in Dolby Pro-Logic, and as a dotted curve the actual values of the center matrix elements in Pro-Logic, illustrating that the actual values are more than 3dB too low for all steerings.

FIG. 30 shows the square root of the sum of the squares of LRL and LRR, using the elements implemented in decoder version 1.11 illustrating 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, also showing the "mountain range" in the rear quadrant and including the "TV matrix" dip of 3dB at the center of the plane, which is hard to see in this projection;

FIG. 31 illustrates graphically the numerical solution for GS and GR for constant power level along the *cs*=0 axis, and zero output along the boundary between left and center;

FIG. 32 shows in perspective graphically 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, illustrating that except for the valley created by the "TV matrix" correction, the sum of the squares is close to one and continuous;

FIG. 33 shows similarly the center left (CL) matrix element of the four channel decoder (and the Pro-Logic decoder), which is also the graph of the center right (CR) matrix element if left and right are interchanged, showing that the middle of the graph and the right and rear vertices have the value 1, the center vertex has the value 1.41, but in practice this element is scaled so the maximum

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value is one;

FIG. 34 shows for comparison the center left matrix element in the decoder version 1.11, in which the middle value and the right and rear vertices have been reduced by 4.5dB, so that as *cs* increases, the center rises to the value of 1.41 in two slopes;

FIG. 35 shows graphically as a solid curve the center attenuation needed for the LFL and LFR values according to the present invention 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, and also shows as a dotted curve the center values for a standard decoder;

FIG. 36 shows graphically as a solid curve the value of GF needed for constant energy ratios with center attenuation GC, accordsing to the invention, and as a dashed curve the value of sin(*cs*)*corr1 (the previous LFR element), while the dotted curve shows sin(*cs*), illustrating that GF is close to zero until cs reaches 30 degrees, and then increases sharply;

FIG. 37 shows the left front right (LFR) matrix element with the correction for center level along the *lr*=0 axis, indicating that the value is zero in the middle of the plane (no steering) and remains zero as *cs* increases to 22.5 degrees along the *lr*=0 axis, falling off to match the previous value along the boundary from left to center and from right to center;

FIG. 38 shows in perspective the center left (CL) matrix element with the added center boost function according to the invention, also showing the correction for panning along the boundary between left and center;

FIG. 39 illustrates graphically the levels of the center output and the left output as a signal pans from center to left showing that with the correction the panning of the center, while not perfect, is reasonably close to the inverse of the left output (the values on the *cs* axis are inverted); and

FIG. 40 shows a block schematic of an active encoder according to the present invention.

Detailed Description of the Invention

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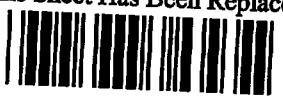

Preferred embodiments of the invention include a five channel and a seven channel decoder with maximum lateral separation, although reference will be made to general design principles that may be applied to decoders with other numbers of channels as well.

In designing a passive matrix, the encoding will be assumed to follow the standard Dolby Surround matrix, and the decoder has four outputs such that the left output signal from the decoder comprises the left input times one; the center is the left input times 0.7 (strictly $\sqrt{0.5}$ or 0.7071) plus the right input times 0.7; the right output signal is the right input signal times one; and the rear output is the sum of the left input times 0.7 and the right input times –0.7.

Referring to FIG. 1, there is a simplified schematic of a passive Dolby surround matrix decoder **1** according to the prior art, in which these signal relationships are maintained. The A (LEFT) and B (RIGHT) audio signals are applied respectively to the input terminals **2**, **4**, and are buffered by unity gain buffer amplifiers **6** and **8** respectively. They are also combined in the above-specified ratios by signal combiners **10** and **12**. The outputs of buffers **6**, **8** appear at the LEFT (L) and RIGHT (R) output terminals **14**, **16**, respectively, and the outputs of signal combiners **10**, **12**, appear at the CENTER (C) and SURROUND (S) output terminals **18**, **20**.

As stated previously, this matrix has constant gain in all directions, and all outputs are equal in amplitude when inputs are decorrelated.

It is possible to extend the passive matrix design to more than four channels. If we wish to have a left rear speaker, the appropriate signal can be made by using suitable matrix elements, but additional conditions are required to form a unique solution; the loudness of the decorrelated component of the signal should be equal in all outputs, and the separation should be high in opposite directions.

The matrix elements are given by sines and cosines of the direction angle of the output. For example if the angle α is defined such that $\alpha = 0$ for a full left

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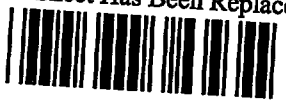

output and is 90° for an output at front center, then the front center matrix elements are:

Left matrix element = $\cos(\alpha/2)$ ...(1)

Right matrix element = $\sin(\alpha/2)$ ...(2)

Thus for $\alpha = 90°$, both matrix elements are 0.71, as specified by the standard Dolby Surround matrix.

The matrix elements as defined by equations (1) and (2) are valid for $\alpha = 0$ (full left) to $\alpha = 180°$ (full right), where the sign of the matrix element for left changes. For the left rear quadrant, α goes from 0° to −90°, so that the sign of the right component is negative. For the right rear quadrant, however, the left matrix element sign is negative. At center rear, $\alpha = 270°$ or −90°, and the two components are equal and opposite in sign; conventionally the right signal coefficient is negative in this case. This could be specified by stating the range of α in equations (1) and (2) as [−90°, 270°), where a square bracket implies inclusion of the adjacent limit value and a parenthesis implies that the limit is not included in the range.

The separation between two outputs is defined as the difference between the levels of a signal in one output and the signal in the other, expressed in decibels (dB). Thus if there is a full left signal, the right input component is zero, and the components in the left and center outputs are 1 and 0.71 respectively times the left input signal. The separation is a level ratio of 0.71 or −3dB (the minus sign is normally dropped.)

The separation between any two directions which have an angle difference of 90° is always 3dB for this matrix. For directions separated by less than 90°, the separation will be less than 3dB. For example, outputs at full rear ($\alpha = -90°$) and left rear ($\alpha = -45°$) will have a separation given by:

Separation = $\cos(45°) * L / (\cos(22.5°) * L) = 0.77$

= 2.3dB ...(3)

This situation can be improved with an active matrix. The object of an active matrix is to increase separation between adjacent outputs when there is a

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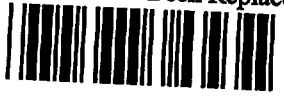

directionally encoded signal at the decoder inputs. We can also raise the question of how such a decoder behaves when the inputs consist entirely of decorrelated "music", and how the decoder behaves when there is a mixture of a directional signal and music. In this context, we shall use the word "music" to denote any decorrelated signal of such complexity that both the directional control signals referred to previously and assumed to be derived from the stereophonic audio input signals are effectively zero.

The following design criteria may be applied to any active matrix, noting that they are fulfilled with various degrees of success by decoders in the current art.

A. When there is no decorrelated signal, there should be a minimum output from those channels not related to the ones involved in reproducing the directional signal. For example, a signal which is intended to be reproduced at a location halfway between right and center should produce no output in the left and rear channels. Likewise a signal intended for center should have no output in either left or right outputs. (This is the principle of pairwise mixing, as extended to surround sound reproduction.)

B. The output from the decoder for directional signals should have equal loudness regardless of the encoded direction. That is, the sum of the squares of the various outputs should be constant if a constant level directional component is moved through all directions. Most current art decoders do not achieve this criterion perfectly. There are loudness errors in all, but these errors are not significant in practice. This is the constant loudness criterion.

C. The loudness of a music (i.e. decorrelated) component of an input signal should be constant in all output channels regardless of how the directional component of the input is moved, and regardless of the relative levels of the directional component and the music. This requirement means that the sum of the squares of the matrix elements for each output should be constant as the matrix elements change with direction. Decoders in the current art disobey this criterion in ways which are often noticeable. This may be called the constant

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power criterion.

D. The transition between the reproduction of a decorrelated music component only, and the reproduction of a directional signal only, as their relative levels change, should occur smoothly and involve no shifts in the apparent direction of the sound. This criterion is also violated in various significant ways by decoders in the current art. It may be called the constant direction criterion.

In a film decoder which must obey the specification for Dolby Pro-Logic, a surround sound reproduction system in common use, criterion D above does not apply, and instead the following criterion E must be satisfied:

E. The signal intended to come from any direction in the front of the room, from left through center to right, should be boosted in level by 3dB relative to the level such a signal would have in a passive Dolby Surround matrix when there is little or no decorrelated component of the input signals (i.e. no music is present.) When music is the dominant input signal (no correlated components present,) the level is not boosted. Thus as the decoder makes the transition from a music only signal to a pure directionally encoded signal, the level of the directional signal in the front hemisphere should be raised.

The optimal design of a decoder which matches the Dolby Pro-Logic specification should have decorrelated music constant in all channels except in outputs where there is a strong directionally encoded signal, and the music in these channels can rise in level a maximum of 3dB proportional to the strength of the directional signal relative to the music. Music level should never decrease in any output where there is no directionally encoded signal. This may be called the minimal gain-riding criterion.

In all current active matrix decoders an implied principle of operation is that in the absence of a directionally encoded signal the matrix should revert to the passive matrix described above, as implemented for the desired number of output channels. This assumption appears at first glance reasonable; however, it is neither necessary nor desirable from the point of view of psychoacoustic perception. Decoders according to this invention replace the above assumption

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with a requirement:

F. An active decoder matrix should have maximum lateral separation at all times, both during reproduction of decorrelated music signals and for music signals in the presence of a directionally encoded signal. For example if the music signal has violins only on the left and cellos only on the right, these locations should be maintained regardless of the strength or direction of a concurrently present directional signal. This requirement can only be relaxed when a strong directionally encoded signal is being removed from an output which should not reproduce it. Under these conditions, the music will drop in level unless the matrix elements are altered to add more energy to the affected channel from the direction opposite to the steered direction. This will reduce separation, but this separation reduction is difficult to hear in the presence of a strong directionally encoded signal.

The need for high separation (especially when there is no directionally encoded signal) comes from psychoacoustics. Prior art has conceived of the matrix as inherently symmetric, with all directions being treated as equally important. However, this is not the case in practice. Humans have two ears, and in watching film or listening to music they generally face forward. Thus frontal and lateral sounds are perceived differently.

There is a dramatic difference between a sound field having up to 4dB of separation and one which has more. (This fact was recognized in the CBS SQ matrix, which had lateral separation exceeding 8dB in the passive decoder, while sacrificing front to rear separation.) In the inventor's opinion, the difference between a discrete five channel film reproduction and a conventional matrix reproduction is due to the low lateral separation between the surround channels.

Griesinger, U. S. Patent No. 5,136,650, recognizes the value of this requirement (F) and describes a six channel decoder where the two additional channels are designed to be placed at the sides of the listener. These outputs have the desired properties for a left rear and a right rear output channel, as long as the directional component of the output is steered to the front hemisphere.

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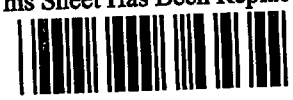

That is, they reduce the level of the steered component, regardless of its direction, and they have full left-right separation when there is no directionally encoded signal. The outputs described in the above-referenced patent do not have constant level for non-directionally encoded music in the presence of a steered signal, and that defect is corrected in the present invention.

The encoder design in the above-referenced patent was used with some modification to make a number of commercially available decoders. The matrix design in the rear hemisphere for these decoders was developed heuristically, but generally meets the requirements stated above fairly well. There is, however, more "pumping" with music than would be optimal, and the leakage of steered signals between the left and right rear outputs is more than the desired level. In this context, "pumping" is audible variation of the music signal due to variation of the directional control signals responding to the direction of the directionally encoded signal.

For both reasons, it was necessary to improve the decoder design, and this invention resulted from this design effort. It turns out that the requirements A through F above uniquely specify a matrix, which will be mathematically described below.

For mathematical simplification, the encoder assumed in the design of the decoder is a simple left-right pan pot. When steering from left to center to right a standard sine-cosine curve is used, as described by equations (1) and (2) above. These may be restated in the form:

$$L = \cos t \quad ...(4)$$

$$R = \sin t \quad ...(5)$$

where

$$t = a / 2 \quad ...(6)$$

In the frontal steering mode above, the angle t varies from 0 to 90°. For steering in the rear half of the room, from left to rear (surround) to right, the right channel pan pot output polarity is inverted. This can be described by the pair of equations:

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$$L = \cos t \quad \ldots(7)$$

$$R = -\sin t \quad \ldots(8)$$

Full rear steering occurs when $t = 45°$, and steering to left surround, a position intermediate between left and rear, occurs when $t = 22.5°$.

Note the similarity of this encoding to the matrix elements of the passive matrix described above. Here, however, the steering angle is divided by two and the sign change for rear steering is included explicitly.

In designing the decoder, it must first be decided what outputs will be provided, and how the amplitude of the steered component of the input will vary in each output as the input encoding steering angle varies. In the mathematical description below, this function can be arbitrary. However, in order to satisfy requirement B, the constant loudness criterion, so that loudness is preserved as a signal pans between two outputs, there are some obvious choices for these amplitude functions.

Assuming that there will be front left, right and center outputs, the amplitude function for each of these outputs is assumed to be the sine or cosine of twice the angle t. For example, as t varies from left, $t = 0°$, to center, $t = 45°$, the output amplitudes should be:

$$\text{Left output} = \cos 2t \quad \ldots(9)$$

$$\text{Center output} = \sin 2t \quad \ldots(10)$$

$$\text{Right output} = 0 \quad \ldots(11)$$

As t goes from center to right, $t = 45°$ to $90°$,

$$\text{Left output} = 0 \quad \ldots(12)$$

$$\text{Center output} = \sin(2t - 90°) = -\cos 2t \quad \ldots(13)$$

$$\text{Right output} = \cos(2t - 90°) = \sin 2t \quad \ldots(14)$$

These functions result in optimum placement of sources between left and center, and between right and center. These functions also result in very simple solutions to the matrix problem. In either of the above cases, any output signals intended for reproduction in the rear of the room should be identically zero.

In designing the five channel version of the improved decoder, a signal

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steered in the rear hemisphere between left and left surround, $t = 0$ to $t = 22.5°$, should have:

$$\text{Left rear output} = \sin 4t \quad \ldots(15)$$

$$\text{Right rear output} = 0 \quad \ldots(16)$$

and when steered between left surround and full rear the total rear output should stay the same. The matrix coefficients used to achieve this are not constant, but vary such that at full rear steering the matrix element for the right input into the left rear output goes to zero.

In the seven channel embodiment, as t goes from 0 to 22.5°, the output in both the left side and left rear outputs should be equal and smoothly rising, proportional to $\sin 4t$. As t goes from 22.5° to 45°, the output in the left side goes down 6dB and the output in the left rear goes up 2dB, keeping the total loudness, the sum of the squares of each output, constant.

As mentioned above, in the improved decoder even when the steered signal is fully to the rear, the left rear and right rear outputs have maximum separation for decorrelated music, since the matrix elements for the right input to the left rear output (and for the left input into the right rear output) are zero resulting in complete separation. Although the right rear has zero output to a steered signal as the steering angle t goes from 0 to 22.5°, the matrix elements used to achieve this signal cancellation are adjusted so that the music output is constant and has minimum correlation with the music signal in the left rear.

To additionally decrease the correlation in the surround field, the seven channel embodiment includes a time delay of about 15ms in the side channels, and in both versions the rear channels are delayed by about 25ms.

Once the loudness functions are chosen for the various outputs under steered conditions, these functions having left to right symmetry, the functional dependence of the matrix elements on the steering angle can be computed.

A standard Dolby surround installation has all the surround loudspeakers wired in phase, and Dolby screening theaters are similarly equipped. However, the standard passive matrix, described above with reference to FIG. 1, has a

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problem with the left rear and right rear outputs. A pan from left to surround results in a transition between L and L–R, and a pan from right to surround goes from R to R–L. Thus the two rear outputs are out of phase when they are fully steered rear. The Fosgate 6-axis decoder described in U.S. Patent No. 5,307,415, among others, has this phase anomaly. In listening to such decoders, this phase inversion was felt to be unacceptable, as a rear-steered sound, such as a plane fly-by, became both thin and phasey in the rear. The decoder of the present invention includes a phase shifter to flip the sign of the right rear output under full rear steering. The phase shift is made a function of the log ratio of center over surround, and is inactive when there is forward steering. Typical phase shifters for this purpose are described below with reference to FIGs. 5a and 5b.

Real world encoders are not as simple as the pan pot mentioned above. However, by careful choice of the method of detecting the steering angle of the inputs, the problems with a standard four-channel encoder can be largely avoided. Thus even a standard film made with a four channel encoder will decode with a substantial amount of directional steering in the rear hemisphere.

Referring to FIG. 2, which represents a standard encoder **21** according to the prior art, as shown in FIG. 1 of the prior Griesinger U. S. Patent No. 5,136,650, there are four input signals L, R, C and S (for left, right, center and surround, respectively,) which are applied to corresponding terminals **22, 24, 26** and **28** and signal combiners and phase shifting elements as shown. The left (L) signal **23** from terminal **22** and center (C) signal **25** from terminal **24** are applied to a signal combiner **30** in ratios 1 and 0.707 respectively; the right (R) signal **27** from terminal **26** and the center (C) signal **25** are similarly applied with the same ratios to signal combiner **32**. The output **31** of signal combiner **30** is applied to a phase shifter **34**, and the output **33** of signal combiner 32 is applied to a second identical phase shifter **38**. The surround (S) signal **29** from terminal **28** is applied to a third phase shifter **36**, which has a 90° phase lag relative to the phase shifters **34, 38**. The output **35** of phase shifter **34** is applied to signal combiner

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40, along with 0.707 times the output 37 of phase shifter 36. Similarly, the output 39 of phase shifter 38 is combined with –0.707 times the output 37 of phase shifter 36 in the signal combiner 42. The outputs A and B of the encoder are the output signals 41 and 43 of the signal combiners 40 and 42 respectively.

Mathematically, these encoder outputs can be described by the equations:

$$\text{Left output (A)} = L + 0.707C - 0.707jS \quad \ldots(17)$$

$$\text{Right output (B)} = R + 0.707C + 0.707jS \quad \ldots(18)$$

Although a standard four channel encoder will not work with five channel discrete film, it is possible to design a five channel encoder which will work very well with the improved decoder according to the present invention. Such an encoder is described with reference to FIG. 3.

The additional elements of the new encoder 48 are applied ahead of the standard encoder 21 of FIG. 2, described above.

The left, center and right signals 51, 53 and 55 are applied to terminals 50, 52 and 54, respectively, of FIG. 3. In each of the left, center, and right channels, an all-pass phase shifter, 56, 58 and 60 respectively, having a phase shift function $\phi(f)$ (shown as ϕ) is inserted in the signal path. The left surround signal 63 is applied to input terminal 62 and then through an all-pass phase shifter 66 with phase shift function ϕ–90°. The right surround signal 65 from input terminal 64 is applied to a ϕ–90° phase shifter 68.

The signal combiner 70 combines the left phase-shifter output signal 57 from phase shifter 56 with 0.83 times the left surround phase-shifted output signal 67 from phase shifter 66 to produce the output signal 71 labeled L, which is applied via terminal 76 to the left input terminal 22 of standard encoder 21.

Similarly, the signal combiner 72 combines the right phase-shifter output signal 61 from phase shifter 60 with –0.83 times the right surround phase-shifted output signal 69 from phase shifter 68 to produce the output signal 73 labeled R, which is applied via terminal 82 to the right input terminal 26 of standard encoder 21.

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Similarly, the signal combiner **74** combines –0.53 times the left surround phase-shifter output signal **67** from phase shifter **66** with 0.53 times the right surround phase-shifted output signal **69** from phase shifter **68** to produce the output signal **75** labeled S, which is applied via terminal **80** to the surround input terminal **28** of standard encoder **21**.

The output signal **59** of the center phase shifter **58**, labeled C, is applied via terminal **78** to the center input terminal **24** of standard encoder **21**.

The encoder of FIG. 3 has the property that a signal on any of the discrete inputs LS, L, C, R and RS will produce an encoded signal which will be reproduced correctly by the decoder of the present invention. A signal which is in phase in the two surround inputs LS, RS, will produce a fully rear steered input, and a signal which is out of phase in the two surround inputs will produce an unsteered signal, since the outputs A and B of the standard encoder will be in quadrature.

The mathematical description of the encoder of FIG. 3 used in conjunction with the standard encoder of FIG. 2 may be given in the form:

$$A = (L - j0.83LS) + 0.71C + 0.38(RS - LS) \quad ...(19)$$

$$B = (R + j0.83RS) + 0.71C - 0.38(RS - LS) \quad ...(20)$$

All current surround decoders which use active matrices control the matrix coefficients based on information supplied from the input signals. All current decoders, including that of the present invention, derive this information by finding the logarithms of the rectified and smoothed left and right input signals A and B, their sum A+B and their difference A–B. These four logarithms are then subtracted to get the log of the ratio of the left and right signals, l/r, and the log of the ratio of the sum and difference signals, which will be identified as c/s, for center over surround. In this description, l/r and c/s are assumed to be expressed in decibels, such that l/r is positive if the left channel is louder than the right, and c/s is positive if the signal is steered forward, i.e. the sum signal is larger than the difference signal. The attenuation values in the five channel passive encoder

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above are chosen to produce the same value of l/r when the LS input only is driven, it being understood that the simplified encoder is used to design the decoder when the angle t has been set to 22.5° (rear). In this case, l/r is 2.41, or approximately 8dB.

For a monaural signal which has been distributed with the simplified encoder between the two input channels such that A = cos t and B = ± sin t, l/r and c/s are not independent. To find the steering angle t, we need only find the arctangent of the left level divided by the right level, or if we define full left as t = 0, then:

$$t = 90° - \arctan(10 \wedge ((l/r) / 20)) \quad ...(21)$$

degrees if l/r is in dB as stated above.

However, since the two levels are compared in magnitude only, to determine whether the steering is front or back we need to know the sign of c/s, which is positive for forward steering and negative for rear steering.

In the real world, the input signals to the decoder are not derived from a pan pot but from an encoder as shown in FIG. 2, which utilizes quadrature phase shifters. In addition, there is almost always decorrelated "music" present along with steered signals.

In the following description, the problem of specifying the matrix elements is divided into four sections, depending on what quadrant of the encoded space is being used, i.e. left front, left rear, right front or right rear.

We will assume a seven channel decoder with left front, center, right front, left side, right side, left rear and right rear outputs. Two matrix elements must be specified for each output, and these will be different depending on the quadrant for the steering. The right front and right rear quadrant coefficients can be found by reflection about the front-back axis, as the matrix has left-right symmetry, so only the left front and left rear steering effects will be derived here.

For the front quadrant, we will assume that requirement D above, rather than requirement E for Dolby surround, is used, and add the correction later.

Front steering is similar to Griesinger (U. S. Patent No. 5,136,650) but the

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functions which describe the steering in the present invention are different, and unique. To find them we must consider each output separately.

The left output should decrease to zero as the angle t varies from 0 to 45°, since we do not want any center steered signals to appear in the left front channel. If $t = 0$ is full left, we define an angle

$$ts = \arctan(10 \text{ ^ } ((c/s) / 20)) - 45^\circ \quad ...(22)$$

The left output is the matrix element LL times the left input plus the matrix element LR times the right input. A fully steered signal from thesimplified encoder results in the left input A = cos ts and the right input B = sin ts over this range. We want the level in the left output to smoothly decrease as t increases, following the function FL(ts), which in our example decoder is assumed to be equal to cos(2ts). Thus the left output is described by:

$$\text{Left output} = LL \cos ts + LR \sin ts$$
$$= FL(ts) = \cos(2ts) \quad ...(23)$$

If the output to decorrelated music is to be constant, the sum of the squares of the matrix coefficients must be one, i.e.

$$LL^2 + LR^2 = 1 \quad ...(24)$$

These equations, which are basically in the same form for all outputs, result in a quadratic equation for LFR, which has two solutions. In every case, one of these solutions is greatly preferred over the other. For the left output,

$$LR = \sin ts \cos(2ts) +/- \cos ts \sin(2ts) \quad ...(25)$$

$$LL = \cos ts \cos(2ts) -/+ \sin ts \sin(2ts) \quad ...(26)$$

Choosing the preferred sign, which is minus in equation (25) and plus in equation (26), and applying mathematical identities, these simplify further to:

$$LL = \cos ts \quad ...(27)$$

$$LR = -\sin ts \quad ...(28)$$

The right output should be zero over the same range of the angle ts, i.e.

$$\text{Right output} = RL \cos ts + RR \sin ts = 0 \quad ...(29)$$

Once again, the decorrelated music should be constant, so

$$RL^2 + RR^2 = 1 \quad ...(30)$$

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and these lead by similar reasoning to the result

$$RL = -\sin ts \quad ...(31)$$

$$RR = \cos ts \quad ...(32)$$

The center output should smoothly decrease as steering moves either left or right, and this decrease should be controlled by the magnitude of l/r, not the magnitude of c/s. Strong steering in the left or right directions should cause the decrease. This will result in quite different values for the center left matrix element CL and the center right element CR, which will swap when the steering switches from right to left. The l/r based steering angle will be called *tl* here. It is assumed to go from 0 at full left to 45° when steering is full center or when there is no steered signal.

$$tl = 90° - \arctan(10 \wedge ((l/r) / 20)) \quad ...(33)$$

where l/r is expressed in dB.

The center output should smoothly increase as *tl* varies from 0 (full left) to 45° (center). The function for this increase will be called FC(*tl*), which is equal to sin(2*tl*) in this embodiment. By the above method,

$$\text{Center output} = CL \cos tl + CR \sin tl = FC(tl)$$

$$= \sin(2tl) \quad ...(34)$$

Once again, for constant loudness of the music,

$$CL^2 + CR^2 = 1 \quad ...(35)$$

which yields the solutions

$$CR = \sin tl \sin(2tl) -/+ \cos tl \cos(2tl) \quad ...(36)$$

$$CL = \cos tl \sin(2tl) +/- \sin tl \cos(2tl) \quad ...(37)$$

The preferred sign is plus in equation (36) and minus in equation (37). The matrix elements for the rear outputs during front steering are not as simple to derive as those for the front outputs. To derive them, we use the argument and formulae presented in Griesinger (U. S. Patent No. 5,136,650.)

The problem is that we want the left rear LRL matrix element to be 1 when there is no steering, and yet we want no directional output from this channel during either left or center steering. If we follow the method used above, we get

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matrix elements which give no output when the signal is steered to the left or center, but when there is no steering, the output will be the sum of the two input signals. This is a conventional solution, where there is poor separation when steering stops. We want full separation, which means LRL must be one and LRR must be zero with no steering.

To solve this problem, the matrix must be made dependent both on the value of l/r and that of c/s. A solution is given in Griesinger (U. S. Patent No. 5,136,650) in which side left and right outputs are the "supplemental outputs". The solution derived there solves the problem of canceling the directional component at all angles in the left side output, but the music component of the output decreases by 3dB as the steering goes to full center.

We can correct the coefficients to avoid this defect by multiplying them by the factor (cos *ts* + sin *ts*), where *ts* is an angle which is zero when c/s is one, and which increases to 45° when c/s is large and positive. In the following equations, the angles *ts* and *tl* are derived from c/s and l/r respectively:

$$ts = \arctan(c/s) - 45^\circ \quad \text{...(38)}$$

$$tl = \arctan(l/r) - 45^\circ \quad \text{...(39)}$$

Note that *tl* here is different from the angle defined previously for the center output.

In the terminology of the previous patent, the control signals developed at the inputs to several variable gain amplifiers (VGAs) are called GL, GC, GR and GS for left, center, right and surround respectively, and two supplemental signals GSL and GSR are derived from these for the left and right surround VGA's. The coefficients here described use a linear combination of the G values to provide the left and right coefficients as a function of the two angles *ts*, derived from c/s, and *tl*, derived from l/r, respectively.

By the definitions therein,

$$GL = ((\cos tl - \sin tl)/\cos tl) = 1 - \tan tl \quad \text{...(40)}$$

$$GC = 2\,(\sin ts/(\cos ts + \sin ts)) \quad \text{...(41)}$$

(there is a factor of two that was omitted in the printing of the earlier patent),

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$$GS = (\cos ts + \sin ts) \cdot \left[\frac{(1-\sin tl)}{\cos tl} \cdot \frac{(\cos tl - \sin tl)}{\cos tl} - 0.5 \cdot \frac{2\sin ts}{(\cos ts + \sin ts)} \right]$$

$= 0$

...(42)

(since this is a front quadrant), and

$$GSL = GL\,((1 - \sin tl)/\cos tl)$$

$$= GL\,(\sec tl - \tan tl)$$

$= (1 - \tan tl)(\sec tl - \tan tl)$

$$= (\cos ts + \sin ts) \cdot \left[\frac{(\cos tl - \sin tl)}{\cos tl} - 0.5 \cdot \frac{2\sin ts}{(\cos ts + \sin ts)} \right]$$

...(43)

and the left and right supplemental signals are given by:

$$LS = A\,(1 - GSL) - 0.5\,(A + B)\,GC - 0.5\,(A - B)\,GS - B \times GL \quad ...(44)$$

$$RS = B\,(1 - GSR) - 0.5\,(A + B)\,GC + 0.5\,(A - B)\,GS - A \times GR \quad ...(45)$$

Thus, the coefficients LSL and LRL are given by:

$$LSL = LRL = (\cos ts + \sin ts)\,(1 - GSL - 0.5GC) \quad ...(46)$$

which becomes, after some manipulation,

$$LSL = LRL = (\cos ts + \sin ts)\,(\sec tl - 1) \times (\sec tl - \tan tl) - \sin ts \quad ...(47)$$

The coefficients LSR and LRR are also equal, given by:

$$LSR = LRR = (\cos ts + \sin ts)\,(-0.5\,GC - GL)$$

...(48)

which becomes, after some manipulation,

$$LSR = LRR = (\cos ts + \sin ts)\,(\tan tl - 1) - \sin ts \quad ...(49)$$

The right side and rear outputs when the input is steered between left and center can be found with the previous method, but the steering angle used must

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be *ts*, derived from c/s, so that it will revert to the right input when there is no steering. We need only remove signals which are steered to center. The equations to solve are:

Right rear output = RRL cos *ts* – RRR sin *ts* = 0 ...(50)

and

$RRL^2 + RRR^2 = 1$ ...(51)

which yield the solution:

RRR = RSR = cos *ts*

RRL = RSL = sin *ts* ...(52)

The above equations completely specify the matrix elements for front steering. For rear steering, when c/s is negative the following are true:

The left and right main elements are the same as for front steering, except that the angle *ts* is determined from the absolute value of log(c/s) which yields:

ts = arctan(10 ^ (s/c) / 20)) – 45° ...(53)

and the sign of the cross matrix element is reversed, yielding:

LL = cos *ts* ...(54)

LR = sin *ts* ...(55)

and

RL = sin *ts* ...(56)

RR = cos *ts* ...(57)

The center matrix elements are identical in rear steering as they depend only on angles derived from l/r, and are not dependent on the sign of c/s.

The side left and side right outputs should have full separation when steering is low or zero. However, the signal on the left side and rear outputs must be removed when there is strong left steering.

We use the previous definition for *tl* for center steering,

tl = 90° – arctan(10 ^ ((l/r) / 20)) ...(58)

as *tl* varies from 0 to 22.5°. Under strong steering, the left side and left rear outputs are zero when *tl* = 0°, but increase with *tl* according to the value sin 4*tl*. In the presence of uncorrelated music, represented by the signals A = cos *t*, B =

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$-\sin t$, the coefficients LSL, LRL, LSR and RSR must satisfy:

$$LSL = LRL \quad ...(59)$$

$$LSR = LRR \quad ...(60)$$

to have equal outputs at the sides and rear, and the amplitude during steering follows $FS(tl) = \sin 4tl$, so that

$$LSL \cos tl - LSR \sin tl = FS(tl) \quad ...(61)$$

For the music to have constant level,

$$LSL^2 + LSR^2 = 1 \quad ...(62)$$

Solving as before,

$$-LSR = \sin tl \; FS(tl) +/- \cos tl \sqrt{(1 - FS(tl)^2)} \quad ...(63)$$

$$LSL = \cos tl \; FS(tl) -/+ \sin tl \sqrt{(1 - FS(tl)^2)} \quad ...(64)$$

Simplifying and using the preferred sign, as before,

$$-LSR = \sin tl \sin 4tl + \cos tl \cos 4tl \quad ...(65)$$

$$LSL = \cos tl \sin 4tl - \sin tl \cos 4tl \quad ...(66)$$

which may be further reduced to:

$$-LSR = \cos 3tl \quad ...(67)$$

$$LSL = \sin 3tl \quad ...(68)$$

The right side and right rear outputs are inherently free of the left input when there is steering in the left rear quadrant, but we must remove signals steered center or rear, so terms must be included that are sensitive to c/s. Right side and right rear outputs are equal, except for different delays, and we have to solve:

$$\text{Right rear/side output} = RSL \cos ts + RSR \sin ts = 0 \quad ...(69)$$

$$RSL^2 + RSR^2 = 1 \quad ...(70)$$

which yield the solution:

$$RSL = \sin ts \quad ...(71)$$

$$RSR = \cos ts \quad ...(72)$$

So far, the decoder design meets all of the requirements set out at the start. Signals are removed from outputs where they do not belong, full separation is

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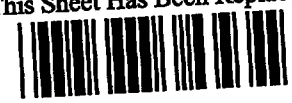

maintained when there is no steering, and the music has constant level in all outputs regardless of steering. Unfortunately, we cannot meet all of these requirements for the rear output in the rear quadrant. One of the assumptions must be broken, and the least problematic one to break is the assumption of constant music level as the steering goes to full rear. The standard film decoder does not boost the level to the rear speaker, and thus a standard film decoder does not increase the music level as a sound effect moves to the rear. The standard film decoder has no separation in the rear channels. We can get the rear separation we want only by allowing the music level to increase by 3dB during strong rear steering. This is in practice more than acceptable. Some increase in music level under these conditions is not audible — it may even be desirable.

We have been finding the matrix elements to the rear based on a steering angle *tl* derived from the l/r level ratio. As we move from $tl = 22.5°$ to $tl = 45°$, this ratio expressed in dB decreases to zero, while the log of the center to surround ratio (c/s) becomes a large negative value.

Consider what happens when a directional signal at $tl = 22.5°$ is faded down into non-directional music. In this case, again, the log of l/r decreases to zero as the non-directional music becomes predominant. We need to distinguish this case from that above, where the steering goes strongly to the rear. The best solution is to make the matrix elements relax to high separation when l/r goes to zero, while keeping the music level constant. The result is easy to derive:

$$tl = 90° - \arctan(l/r) \quad ...(73)$$

$$LRL = \cos(45° - tl) \quad ...(74)$$

$$LRR = -\sin(45° - tl) \quad ...(75)$$

where *tl* goes from 22.5° to 45°. These matrix elements keep the music level constant, but they cause the output of a steered signal to decrease by 3dB when the signal goes to the rear. We can fix this by adding a dependency on c/s, by boosting the LRL value by an amount proportional to the increase in the log of the c/s ratio. Solving for the value of boost needed to keep the rear output level constant, we can express the results in a table:

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c/s in dB	RBOOST
−32	0.41
−23	0.29
−18	0.19
−15	0.12
−13	0.06
−11	0.03
−9	0.01
−8	0.00

TABLE 1: Variation of RBOOST with c/s

In terms of these results, the left rear output matrix coefficients in the five channel version are:

$$LSL = \cos(45^\circ - tl) + \text{RBOOST}(\log c/s) \quad ...(76)$$

$$LSR = -\sin(45^\circ - tl) \quad ...(77)$$

and similarly for the right channel,

$$RSL = \sin(45^\circ - tl) \quad ...(78)$$

$$RSR = \cos(45^\circ - tl) + \text{RBOOST}(\log c/s) \quad ...(79)$$

For the seven channel embodiment of the invention, we add an additional dependency on c/s to take into account the desired reduction of output in the left side and right side channels as the steering goes to full rear, remembering that left side and left rear coefficients were equal in the case of steering from full left to left rear. The reduction of side output is accompanied by a boost in the corresponding rear output to maintain constant power in the steered signal. It may also be desirable to increase the cross term, which reduces the separation a little, but apparently this is not audible.

We define a rear side boost function RSBOOST(*ts*) using an angle *ts* derived from the value of c/s:

$$ts = 90^\circ - \arctan(s/c)$$

where *ts* varies from 22.5° to 45°, so that the RSBOOST function rises from zero at *ts* = 22.5° to 0.5 at *ts* = 45°.

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Then

$$RSBOOST = 0.5 \sin(2(ts - 22.5°)) \quad ...(80)$$

and for the side outputs,

$$LSL = \cos(45° - tl) + RBOOST(\log c/s) - RSBOOST(ts) \quad ...(81)$$

$$LSR = -\sin(45° - tl) \quad ...(82)$$

$$RSL = \sin(45° - tl) \quad ...(83)$$

$$RSR = \cos(45° - tl) + RBOOST(\log c/s) - RSBOOST(ts) \quad ...(84)$$

and for the rear outputs,

$$LRL = \cos(45° - tl) + RBOOST(\log c/s) + 0.5\ RSBOOST(ts) \quad ...(85)$$

$$LRR = -\sin(45° - tl) \quad ...(86)$$

$$RRL = \sin(45° - tl) \quad ...(87)$$

$$RRR = \cos(45° - tl) + RBOOST(\log c/s) + 0.5\ RSBOOST(ts) \quad ...(88)$$

For the film decoder mode, we have to replace criterion D above by criterion E, which entails boosting the levels in front channels by 3dB in all front directions. The matrix can be made to perform this way by adding similarly derived boost terms to the front elements during front steering. For example, during left steering the LL matrix element, here called LFL, should be increased by a boost function depending on l/r, where we define two angles:

$$tlr = 90° - \arctan(l/r) \quad ...(89)$$

$$trl = 90° - \arctan(r/l) \quad ...(90)$$

Then (cf. eq. (27) above),

$$LFL = \cos ts + LFBOOST(tlr) \quad ...(91)$$

and for steering to the right,

$$RFR = \cos ts + LFBOOST(trl) \quad ...(92)$$

Both center matrix elements are also boosted during center steering:

$$CL = \sin tl + 0.71\ LFBOOST(ts) \quad ...(93)$$

$$CR = \cos tl + 0.71\ LFBOOST(ts) \quad ...(94)$$

These equations completely specify the additional requirements for a film

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decoder.

When there is no center channel loudspeaker, the Dolby specification suggests that the center channel output should be added to the left front and right front outputs with a gain of −3dB or 0.707. While this reproduces the center channel dialog at the proper level, it reduces the separation between left and right. For example, when there is no steering, the center output is 0.71L + 0.71R. Adding this to left and right yields a left output of 1.5L + 0.5R and a right output of 1.5R + 0.5L, so that the separation is reduced to 0.5/1.5 = 9.5dB.

To avoid this effect, it would be better to modify the left and right matrix elements when there is center steering, using the angle *ts* derived from c/s, so that:

$$LFL = 1 + LFBOOST(ts) \quad ...(95)$$

$$RFR = 1 + LFBOOST(ts) \quad ...(96)$$

$$LFR = RFL = 0 \quad ...(97)$$

Unlike the previously derived matrix coefficients, these do not remove the dialog from the left and right channels, and also keep it at the proper loudness in the room, while maintaining full left-right separation for music as long as the steering is in the front hemisphere.

In a preferred five channel embodiment shown in FIG. 4, five of the seven channels described above are implemented, and the decoder provides the left, center, right, left rear and right rear outputs, the left side and right side outputs being omitted. It is understood from the above mathematical description that the circuitry for the left rear and right rear outputs of the seven channel decoder can be obtained by similar circuitry to that for the left and right surround outputs shown, with an additional 10ms delay similar to the blocks **96** and **118** which implement 15ms delays.

The addition of the RBOOST, RSBOOST and LFBOOST functions as described for the seven channel decoder, the film decoder mode and the missing center channel mode in the last section will be simple modifications apparent to those skilled in the art. In the digital implementation, they consist merely of

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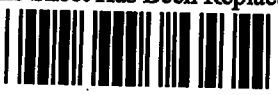

adding the appropriate boost expressions derived from the angles *ts* and *tl* with appropriate definitions based on the steered direction to the corresponding matrix coefficients before performing the multiplications and additions required to generate the matrixed output signals.

In the decoder **90** of FIG. 4, the input terminals **92** and **94** respectively receive the left and right stereophonic audio input signals labeled A and B, which may typically be outputs from the encoders of FIGs 2, 3, or 7, directly or after transmission/recording and reception/playback through typical audio reproduction media.

The A signal at terminal **92** passes through a short (typically 15ms) delay before application to other circuit elements to be described below, so as to permit the signal processing which results in the l/r and c/s signals to be completed in a similar time period, thereby causing the control signals to act on the delayed audio signals at precisely the right time for steering them to the appropriate loudspeakers.

The A signal from terminal **92** is buffered by a unity gain buffer **98** and passed to a rectifier circuit **100** and a logarithmic amplifier **102**.

Similarly, the B signal from terminal **94** is passed through a buffer **104**, a rectifier **106** and a logarithmic amplifier **108**.

The outputs of the logarithmic amplifiers **102** and **108**, labeled A" and B" respectively, are combined by subtractor **110** to produce the l/r directional control signal, which is passed through switch **112** to the matrix circuitry described below. In the alternate position of switch **112**, a time constant comprising resistor **114** and capacitor **116** is interposed in this path to slow down the output transitions of the l/r signal.

The B signal from terminal **94** is also passed through a 15ms delay for the reason stated above.

The A and B signals from terminals **92** and **94** are combined in an analog adder **120**, rectified by rectifier **122** and passed through logarithmic amplifier **124**.

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Similarly, the A and B signals are subtracted in subtractor **126**, then passed through rectifier **128** and logarithmic amplifier **130**. The signals from the logarithmic amplifiers **124** and **130** are combined in subtractor **132** to produce the signal c/s, which is passed through switch **134**. In the alternative position of switch **134**, the signal passes through the time constant formed by resistor **136** and capacitor **138**, which have identical values to the corresponding components **114** and **116**. Thus far, the control voltage generation circuit has been described. As is typical of such circuits, the l/r and c/s signals vary in proportion to the logarithms of the ratios between the amplitudes of left A and right B, and of center (sum) and surround (difference) of these signals.

The matrix elements are represented by the circuit blocks **140** - **158**, which are each labeled according to the coefficient they model, according to the preceding equations. Thus, for example, the block **140** labeled LL performs the function described by equation (27), (54), (91) or (95) as appropriate. In each case, this function depends on the c/s output, which is shown as an input to this block with an arrow, to designate it as a controlling input rather than an audio signal input. The audio input is the delayed version of left input signal A after passing through the delay block **96**, and it is multiplied by the coefficient LL in block **140** to produce the output signal from this block.

The outputs of the several matrix elements are summed in summers **160** - **168** thus providing the five outputs L, C, R, LS and RS at terminals **172**, **174**, **176**, **178**, and **180** respectively. As mentioned above, the RS signal is passed through a variable phase shifter **170** before being applied to the output terminal **180**. Phase shifter **170** is controlled by the c/s signal to provide a phase shift which changes from 0° to 180° as the signal c/s steers from front to rear.

In the seven channel version of the decoder, circuit elements **152** - **158**, **166**, **168** and **170** are duplicated, being fed from the same points as their corresponding elements shown in FIG 4, but with the coefficients LRL, LRR, RRL and RRR in blocks corresponding to **152** - **158** respectively, and with additional 10ms delays

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similar to blocks **96** and **118**, which may be inserted either ahead of these blocks or after the corresponding summer elements to blocks **166** and **168**.

Although an analog implementation is shown in FIG. 4, it is equally possible, and may be physically much simpler, to implement the decoder functions entirely in the digital domain, using a digital signal processor (DSP) chip. Such chips will be familiar to those skilled in the art, and the block schematic of FIG. 4 will be readily implemented as a program operating in such a DSP to perform the various signal delays, multiplications and additions, as well as to derive the signals l/r and c/s and the angles *tl* and *ts* from these signals, to be used in the equations previously disclosed, so as to provide the full functionality of the decoder according to the present invention.

Turning to FIG. 5a, an analog version of the phase shifter **170** is shown. In this phase shifter circuit, the input signal RS' is buffered by an operational amplifier **182** and then inverted by a second operational amplifier **184** with the input resistor **186** and equal feedback resistor **188** defining unity gain. The outputs of amplifiers **182** and **184** are respectively applied through variable resistor **190** and capacitor **192** to a third operational amplifier **196**, which buffers the voltage at the junction of the variable resistor **190** and capacitor **192** to provide the output signal RS to terminal **180** of FIG. 4. This circuit is a conventional single pole phase shifter having an all-pass characteristic.

The variable resistor **190** is controlled by the c/s signal in such manner that the turnover frequency of the phase shifter is high when the signal is steered to the front, so that the rear output signals are out of phase (due to the matrix coefficients) but reduces as the signal steers to the rear, so that the rear output signals become in phase due to inversion of the right rear output RS. Although the phase shift is not the same at all frequencies, the psychoacoustic effect of this phase shifter is acceptable and reduces the phasiness of the rear signals substantially. As will be apparent to those skilled in the art, more complex multi-pole phase shifters could be used, but would require additional circuitry in all of

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the output channels, so it does not provide a cost-effective way of smoothly reversing the phase of the one rear channel where this is desired.

In FIG. 5b is shown a conventional variable digital delay element that may be used in implementing a digital embodiment of the delay block **170** of the circuit of FIG. 4. In this circuit, the gain value g is controlled by the value of control signal c/s so as to perform the same function as for the analog phase shifter of FIG. 5a. In this circuit, the signals applied to adder **200** are summed and delayed by delay block **202**, the output of which is fed back through a multiplier **204** of gain g to one of the inputs of adder **200**. The RS' signal is applied to the other input of adder **204** and also to multiplier **206**, where it is multiplied by a coefficient $-g$. The output signal from delay block **202** is multiplied by $(1 - g^2)$ in multiplier **208**, and added to the signal from multiplier **206** in adder **210** to provide the RS signal at the output of adder **210**.

While the performance of this phase shifter is not quite identical to that of its analog counterpart in FIG. 5a, it is sufficiently similar to provide the desired effect.

FIGs. 6a through 6e show graphically the variations of the various matrix coefficients of the decoder of FIG. 4 and its enhancements that are described by equations in the preceding section to the description of FIG. 4, for further clarification of the operation of this decoder.

In FIG. 6a, the curves A and B represent the variation of coefficients LL (LFL) and –LR (–LFR) respectively as the value of c/s ranges from 0dB to about 33dB. These curves follow the sine – cosine law as derived in equations (27) and (28). The variation of RR (RFR) and RL (RFL) is similar in form for steering in the right front quadrant.

The curves C and D respectively show the corresponding values of LFL and LFR for the decoder according to the previous Griesinger Patent No. 5,136,650 for comparison. In these curves, which approach the value 0.5 under strong center steering, the music component is 3dB too low, hence the new decoder curves A and

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B which meet at 0.71 provide constant music level, while the old curves do not.

In FIG. 6b are shown the curves E and F representing the center coefficients CL and CR under l/r steering from center (0dB) to left (33dB). The left coefficient CL increases by 3dB while the right coefficient CR decreases to zero as the steering moves to the left. Similar considerations apply but in the opposite sense when the steering is to the right.

The curves G and H represent CL and CR respectively in the decoder of Griesinger's previous patent referenced above, showing that again the music level is not maintained constant, as the curve G does not increase by the required 3dB.

Turning to FIG. 6c, Curves J and K represent the values of the coefficients LSL and LSR respectively as the ratio l/r goes from 0dB (no steering or center steering) to 33dB, representing full left steering. The LSL curve J reduces to zero, as it is removing left signal from the left surround channel, while the LSR signal increases so that the level of the music remains constant in the room. It is clear from the curves that there is a break point at 8dB, corresponding to a steering angle of 22.5° to the rear. Here the matrix elements must total (in r.m.s. fashion) to 1 when the input has only a directional signal. This is achieved if they have values of cos 22.5° or 0.92 and sin 22.5° or 0.38, as can be seen from the curves.

In this context, note that l/r can be zero dB either when the signal is steered fully rear, or when there is no steered component of the signal. In either case, the matrix relaxes to the full left-right separation that is desired.

In FIG. 6d, the curve L represents the RBOOST value tabulated above in TABLE 1 and used in equations (76) and (79), and subsequently. The value of LSL is too small when steering to full rear, so the value of RBOOST is added to it to keep the music level constant. Only LSL is boosted, so complete separation is maintained. The value of RBOOST depends only on c/s, as c/s varies from -8dB to −33dB (full rear) i.e. the x-axis of the graph is −c/s, with c/s in dB.

Also shown in FIG 6d is the curve M which represents the value of RSBOOST. In the seven-channel version of the decoder, this value is subtracted from the left side coefficient and half of it is added to the left rear component,

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when steering between left rear (–8dB) to full rear (–33dB). Again, the axis is –(c/s in dB), and this curve goes from zero to 0.5, as expressed in equation (80) above.

Finally, in FIG. 6e is shown the curve N which represents the variation of the correction factor (sin *ts* + cos *ts*) with the control signal c/s applied to the rear and side surround channels to keep the level of music constant, as described above subsequent to equation (39).

Turning to FIG. 7, there is shown an active encoder suitable for use in movie soundtrack encoding generally, and particularly with reference to the decoder embodiments presented above.

In FIG. 7, the same five signals LS, L, C, R and RS are applied to the correspondingly numbered terminals **62**, **50**, **52**, **54**, **64** respectively as in the encoder of FIG. 3. For each of these signals there is a corresponding level detector and logarithmic amplifier to provide signals proportional to the logarithms of the amplitudes of each of these signals. These elements are numbered **212-230**. The logarithmic signals are respectively labeled lsl, ll, cl, rl and rsl, corresponding to the inputs LS, L, C, R and RS. These signal levels are then compared in a comparator block (not shown), whose action is described below.

Attenuators **254** and **256** attenuate the LS signal by factors of 0.53 and 0.83 respectively, and attenuators **258** and **260** attenuate the RS signal by factors of 0.83 and 0.53 respectively.

Each of the five input signals passes through an all-pass phase shift network, the blocks labeled **232**, **234**, providing phase shift functions ϕ and ϕ–90° respectively for the attenuated LS signal from attenuators **254** and **256** respectively, blocks **236**, **238**, and **240** providing the phase shift function ϕ to each of L, C and R signals respectively. A signal combiner **242** sums 0.38LS with –0.38RS to provide a center surround signal to phase shifter block **244**, which has a phase shift function ϕ. The phase shifter blocks **246** and **248** provide phase shift functions ϕ–90° and ϕ respectively in the RS channel from attenuators **258**

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and **260** respectively.

A signal combining matrix **250** sums the LS(ϕ) signal attenuated by the attenuator **254**, with gain sin Θ_{LS}, the LS(ϕ–90°) signal attenuated by the attenuator **256**, with gain cos Θ_{LS}, the L(ϕ) signal, the C(ϕ) signal with gain 0.707, and the surround signal S = (0.38LS–0.38RS) with phase ϕ, which is labeled S(ϕ), to produce the left output signal A at terminal **44**.

A similar matrix **252** sums the RS(ϕ) signal with gain sin Θ_{RS}, the RS(ϕ–90°) signal with gain cos Θ_{RS}, the R(ϕ) signal, the C(ϕ) signal with gain 0.707, and the S(ϕ) signal, to produce the right output B at terminal **46**.

The steering angles Θ_{LS} and Θ_{RS} are made dependent upon the log amplitude signals lsl, ll, cl, rl and rsl in the following manner in this embodiment of the invention:

Whenever lsl is larger than any of the remaining signals, then Θ_{LS} approaches 90°, otherwise, Θ_{LS} approaches 0. These values may be extremes of a smooth curve. Similarly, if rsl is larger than any of the other signals, Θ_{RS} approaches 90°, otherwise Θ_{RS} approaches 0.

The particular advantage of this mode of operation is that when a signal is applied to the LS or RS input only, the output of the encoder is real, and produces an l/r ratio in the decoder of 2.41:1 (8dB), which is the same value produced by the simplified encoder and the passive encoder.

Turning to FIG. 8, which shows a part of a decoder according to the invention having complex rather than real coefficients in the matrix, the figure illustrates a method for generating a third control signal ls/rs (in addition to the signals l/r and c/s generated by the decoder in FIG. 4), which is used for varying the additional phase shift network of FIG. 9 that is placed ahead of the decoder of FIG. 4 in order to effect the generation of complex coefficients in the matrix.

It will be seen that the A and B signals are now applied to terminals **300** and **302** respectively, instead of to terminals **92** and **94** of FIG. 4. An all-pass

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phase shift network **304** having the phase function $\phi(f)$ of frequency f, and a second all-pass phase shift network **306** having the phase function $\phi(f)-90°$ receive the A signal from terminal **300**. The phase shifted signal from **304** is attenuated by a factor −0.42 in attenuator **308** and the lagging quadrature phase shifted signal from **306** is attenuated by the factor 0.91 in attenuator **310**. The outputs of attenuators **308** and **310** are summed in summer **312**.

The B signal at terminal **302** is passed through an all-pass phase shift network **314** so that the output of summer **312** is signal A shifted by 65° relative to signal B at the output of phase shifter **314**.

The output of summer **312** is passed through attenuator **316** with an attenuation factor 0.46, and to one input of a summer **318**, where it is added to the phase-shifted signal B from shifter **314**. Similarly, the output of phase shifter **314** is attenuated by attenuator **320** with the same factor 0.46 and passed to summer **322** where it is added to the output of summer **312**, the phase-shifted A signal. The particular choices of coefficients in attenuators **308**, **310**, **316** and **320** are made so that signals applied to the LS input only of the passive encoder will produce no output at the summer **308** and a signal applied to the RS input only will produce no output at the summer **322**. The object thus is to design a circuit that will recognize as input of the decoder the case when the signal is only being applied to the left side or right side of the encoder. It does this by a cancellation technique, such that one or the other of the two signals goes to zero when the condition exists.

The output of summer **318** is passed into level detection circuit **324** and log amplifier **326**, while the output of summer **322** is passed through level detector **328** and logarithmic amplifier **330**. The outputs of log amplifiers **326** and **330** are passed to subtractor **332** which produces an output proportional to their log ratio. This output may be selected by switch **334**, or the output from the R-C time constant formed by resistor **336** and capacitor **338**, which have values identical to the corresponding components shown in FIG. 4, may alternatively be selected by

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switch **334** and passed to terminal **340** as the steering signal ls/rs.

Thus the signal ls/rs will either be a maximum positive value when a signal is applied to the LS input of the passive encoder, or a maximum negative value when a signal is applied to the RS input.

The purpose of the signal ls/rs is to control the input phases applied to the decoder of FIG. 4. For this reason, the network of FIG. 9 is interposed between the A and B signals applied to terminals **92** and **94** of FIG. 4.

The circuit shown in FIG. 9 includes a phase shifter **342** of phase function ϕ, which may be the same shifter as **304** in FIG. 8, followed by an attenuator **344** having the attenuation value cos Θ_{RS}, while the phase shifter **346**, which may be the same shifter as **306** in FIG. 8, of phase function ϕ–90°, is passed through attenuator **348** with attenuation factor sin Θ_{RS}. The outputs of attenuators **344** and **348** are summed by summer **350** to provide a modified A signal at terminal **352**, which is to be directly connected to terminal **92** of FIG. 4.

In the lower part of FIG. 9. the B signal is applied to terminal **302** as in FIG. 8, and in one branch passes through phase shifter **354** of phase function ϕ and attenuator **356** of attenuation factor cos Θ_{LS}, while in the other branch it passes through phase shifter **358** of phase function ϕ–90° and attenuator **360** of attenuation factor sin Θ_{LS}. The signals from attenuators **356** and **360** are combined in subtractor **362** to provide a modified B signal at terminal **364**, which is to be directly connected to the terminal **94** in FIG. 4. The result in the change of phase is to produce better separation between the LS and RS outputs of the decoder (as well as the LR and RR outputs in a 7-channel version) when only the LS or RS inputs of the passive encoder are being driven with signals.

The relationship between the control signal ls/rs and the steering angle Θ_{LS} is shown in the inset graph of FIG. 9. As ls/rs reaches 3dB, the angle Θ_{LS} begins to change from 0° rising towards 65° at high values of ls/rs. An exactly complementary relationship applies to the other steering angle Θ_{RS} which is

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controlled by the inverse of ls/rs, which we call rs/ls, so that when rs/ls exceeds 3dB, the value of Θ_{RS} begins to increase from 0°, moving towards an asymptote at –65° when rs/ls is at its maximum value. As Θ_{LS} and Θ_{RS} vary, the matrix coefficients effectively become complex due to the phase changes at the inputs to the main part of the decoder shown in FIG. 4.

FIG. 10 illustrates an alternative embodiment of an encoder that differs from that of FIG. 7 by simplifying the phase shift networks. The number of phase shift networks can by reduced by combining the real signals before sending them through the ϕ phase shifter, thus resulting in only two ϕ and two ϕ–90° phase shift networks. The description of Θ_{LS} and Θ_{RS} is also simplified. Θ_{LS} approaches 90° when lsl/rsl is greater than 3dB, and otherwise is zero (just as in the decoder design). Likewise, Θ_{RS} approaches 90° when rsl/lsl is greater than 3dB, and otherwise is zero.

In FIG. 10, elements corresponding to those in the right half of FIG. 7, namely the attenuators **254-260** and the ϕ–90° phase shifters **234** and **246** have been correspondingly numbered. In order to provide a more detailed discussion of the difference between this encoder and that of FIG. 7, the elements of FIG. 10 not so corresponding have also been numbered. It will be seen that the coefficients shown in signal combiner elements **242**, **250** and **252** of FIG. 7 have been extracted from the signal combiners and applied separately to each of the relevant signals in attenuator elements **262-274**, and that these signals thus modified are combined in simple summing devices **276-284**, while the five ϕ phase shifters **232**, **236-240** and **248** have been replaced by two phase shifters **286-288**.

In FIG. 10, the signal path for the LS signal from terminal **62** of FIG. 7 passes as before through attenuator element **256** and ϕ–90° phase shifter **234**, then passing through the actively controlled attenuator **270** having attenuation factor cos Θ_{LS}, this being the coefficient formerly shown in signal combiner **250** of FIG. 7. This signal is summed in summer **276** as one component of the signal output labeled A at terminal **44** of FIG. 7.

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The signal path for the RS signal at terminal **64** in FIG. 7 similarly passes through attenuator **258** and phase shifter **246**, then through active attenuator **274** having attenuation coefficient cos Θ_{RS}, formerly part of signal combiner **252** of FIG. 7, to summer **280** where it is one component of the signal labeled B at terminal **46** of FIG. 7.

The signal path for the center signal C from terminal **52** of FIG. 7 passes first through attenuator **266** with attenuation coefficient 0.71, after which it is applied to summers **278** and **282**. The L signal from terminal **50** of FIG. 7 is applied directly to summer **278**. The R signal from terminal **54** of FIG. 7 is applied directly to summer **282**. The LS signal is also applied through attenuator **254**, and through active attenuator **268** with attenuation coefficient sin Θ_{LS} to the summer **278**. The RS signal is also passed through attenuator **260** and active attenuator **272** with attenuation coefficient Θ_{RS} to the summer **282**. Finally, the LS signal passes through attenuator **262** of coefficient 0.38 and the RS signal passes through attenuator **264** of coefficient −0.38, both attenuated signals being summed in summer **284**, before the result is applied to summer **278** with positive sign and summer **282** with negative sign.

The output of summer **278** is passed through ϕ phase shifter **286** to summer **276**, and the output of summer **282** is passed through ϕ phase shifter **288** to summer **280**, summers **276** and **280** respectively providing the signals A and B to terminals 44 and **46** of FIG. 7.

Examination of the attenuation and summation of each of the signals LS, L, C, R and RS into each of the outputs A and B will show that these output signals are identical in content to those of FIG. 7, but with three fewer of the expensive phase shifters relative to FIG. 7.

In FIG. 10 is also shown graphically the relationship between the angle Θ_{RS} and the value of rs/ls (or −ls/rs) for signals steered in the right side quadrant. This angle affects the circuit elements **272** and **274**, as indicated by the arrows. An exactly similar relationship exists between the steering angle Θ_{LS} and the

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value of ls/rs, this angle affecting circuit elements **268** and **270**.

Turning to FIG. 11, an encoder is shown, that is very similar in construction to the encoder of FIG. 10. Those elements that are comparable in function are therefore numbered correspondingly. There are several new elements, the four gain control elements, variable attenuators **290-293**, and two control signal generator elements **294, 295**. The input and output terminals have been numbered in correspondence with FIG. 7.

The purpose of the added gain control elements is to correct both the balance between strongly steered front signals and music, and the reduction of separation in response to simultaneous left side and right side signals. When strongly steered signals occur in the left, center, or right channels, the Dolby Pro-Logic compatible type of decoder, i.e. in this case one that meets criterion E, rather than criterion D, applies a boost of 3dB in the front channels. This boost is quite audible as a shift in the balance between dialog and music, for example. Typically, in mixing a soundtrack for playback with a Dolby-compatible decoder, the recording levels of dialog and other strongly steered front channels is compensated by the sound mixer who listens to the sound track through a decoder which applies this boost. However, five-channel films encoded through either a passive encoder or the type of active encoder discussed previously with reference to FIGs. 7 and 10, will not be so compensated.

In the new encoder, the three front signals L, C and R, are passed through three variable attenuators **290-292** respectively having gain coefficients GL, GC and GR. These coefficients are controlled by steering control signals derived from the outputs of the encoder. To do this, the output signals A and B are fed into the inputs of a steering signal voltage generator **294** which comprises identical circuitry to that of the decoder of FIG. 4. The two steering voltages l/r and c/s are thus derived, and will be identical to those generated in an active decoder. These two steering voltages affect the gain coefficients in the manner shown in FIGs. 12a and 12b. The signal l/r and inverse r/l control gains GL and GR respectively

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of elements **290** and **292**, while gain GC of element **291** is controlled by c/s.

When l/r is positive (steering is to the left), the value of GL is reduced from 1 according to the curve shown in FIG. 12a, while the value of GR remains at 1. Similarly, when l/r is negative, the value of GR is reduced according to the same curve (in relation to $|l/r|$) while the value of GL is constant at 1. Likewise, when the front/rear steering c/s is positive (steered to the front), the gain GC varies with c/s according to the curve of FIG. 12a, but GC remains at 1 when the c/s signal is negative. The curve in FIG. 12a is the inverse of the curve N shown in FIG. 6e.

Since the l/r and c/s signals are generated within a feedback loop, because changing a gain also affects the steering voltage, the correction applied to each of the front signals will exactly match the boost applied to them in the film decoder. The result of this is that dialog, music and strong left or right sound effects maintain the balance of the original discrete mix when the original five channels are encoded to two and then decoded back to five or seven channels. There is in fact very little loss of subjective quality when the two channel version is compared to the five channel original. Most of the time, there is no apparent difference at all.

A further improvement in the encoder of FIG. 11 is the addition of the gain coefficient GS of variable attenuator **293**, which is controlled by the control voltage generation circuit **295**. The gain coefficient GS acts upon the signal from summer **284**, which is the difference signal between the left side and right side input signals (multiplied by 0.38) The purpose of this side difference signal is to provide the proper negative value of the c/s signal when there is a strongly steered left side or right side input to the encoder. However, this side difference signal reduces the separation between left side and right side inputs when both are present at the same time. This reduction in separation is particularly important in the case when the LS and RS inputs are nearly equal but uncorrelated, such as during music, applause, or surround effects like rain.

During these unsteered effects, we would like to disable the difference signal, and this can be accomplished by reducing the value of GS whenever there

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is no strong correlation between the left side and right side signals applied to the encoder.

The presence of correlation can be determined from the steering voltages derived from the left side and right side inputs to the encoder, using a control voltage generation circuit **295** similar to that in element **294**, which thus produces the control signals ls/rs and cs/ss. The ls/rs steering voltage was also derived on the original version of the active encoder shown in FIG. 7, to control the values of Θ_{LS} and Θ_{RS}. While this feature is retained in the encoder of FIG. 11, additional circuitry determines the front-back components of the side signals. Both the ls/rs and cs/ss signals control the gain GS of attenuator element **293**. The ls/rs signal also controls the steering angle *tls* in attenuators **270** and **272**, and its inverse, rs/ls, controls the steering angle *trs* in attenuators **272** and **274**.

The value of GS is then determined by taking the larger of the absolute values of signals ls/rs and cs/ss, limiting this value to 7dB, dividing by 7, then subtracting the result from 1. Thus any signal with correlation of 7dB or more will result in GS = 1 so that the encoder works as before, but when uncorrelated signals are applied to LS and RS, the value of GS will diminish accordingly and the decoder will revert to the high separation between these inputs.

In the process of comparisons of encoded/decoded signals versus unencoded multichannel sound, it became apparent that the output from the left front or right front channels during side steering was not reduced enough. According to the Dolby Pro-Logic specification, which does not include left and right side channels, the left front output of the decoder reduces in amplitude by only 2.5dB. This behavior of the front channels is intentional, in order to follow the Dolby specification, but in the standard Dolby specification there are no side channels to decode, and only a single rear output. There is therefore a need to modify the Dolby specification for the left and right front outputs during rear steering when there are side speakers.

In the modified specification, the left front and right front outputs are

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reduced by an additional 3dB when there is rear steering on the same side. Thus the front left signal is reduced by this amount as a signal pans from left to left side, and the right front signal is similarly reduced as a signal pans from right to right side. With the side speakers installed, this clearly improves the apparent motion of a signal moving from the front to either side, and then to the rear; however, it is not so large a departure that it makes much audible difference with a standard Pro-Logic encoded film. The variation in gain for the LL and LR matrix elements for left to left side steering is shown in FIGs 12b and 12c respectively. Similar curves apply to the right side steering.

Another aspect of the decoder improvements is a special limiting correction that may be applied digitally to the l/r and c/s directional control signals. This has the advantage of improving the speed and the accuracy of the steering. During a pan from left to center of a strongly steered signal, the l/r and c/s signals are not independent, but follow a complementary path, shown in FIG. 13. If the logarithmic detectors act rapidly, this curve will be followed dynamically, but when a time constant is included, the value of the rising signal can increase rapidly, but the falling signal is usually changing at a slower speed. The result is that the falling signal is higher than it should be, reducing the dynamic separation. To correct this problem, the signal that is changing more rapidly is used to limit the other signal to follow the curve of FIG. 13. Although some prior art decoders included circuits for limiting control voltage excursions during rapid changes, these circuits were not based on the rate of change of the control signals but rather their absolute values. A particular advantage of the rate of change method is that the increasing signal is enabled to rise rapidly while the falling signal, which represents steering the matrix in a previous direction, is forced to yield to the more rapidly changing signal.

Remembering the definitions of these signals, it is quite easy to work out the relationships between the control signals, that can only occur as a maximum limit, which will not be reached in the presence of decorrelated music.

If we consider a pan from left to center, L = cos t and R = sin t, then the

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control signals are

$$l/r = 20 * \log_{10}(\cos t/\sin t) \quad ...(98)$$

$$c/s = 20 * \log_{10}((\cos t + \sin t)/(\cos t - \sin t)) \quad ...(99)$$

These relationships are plotted against each other in FIG.13.

FIG. 14 shows for reference the form of the left rear left (LRL) matrix element coefficient used in the matrix of FIG. 4, as implemented in the decoder according to the prior patent application No. 08/684,948. The value of this coefficient is plotted in three-dimensional format as the height of the ordinates with respect to the *cs* and *lr* control signals, which are derived from the usual log-ratio detectors in the decoder of FIG. 4. The *cs* signal represents the ratio of center front to rear surround signal amplitudes and the *lr* signal represents the ratio of left to right signal amplitudes. Each of these signals is encoded as an angle ranging from zero to 90 degrees. As can be seen in the illustration, there is a discontinuity in the value of this element near the left vertex, forming a small "cliff" as the signal moves towards the rear. There is also a central "valley" in this representation. FIG. 15 similarly shows the value of the left rear right matrix element, which has a similar discontinuity along the *lr*=0 axis towards the rear.

These discontinuities are due to an error in the theory described in the previous patent application. The problem is that there is a correction applied to the matrix element in the left rear quadrant which was done by a table indexed through a variable called *lr bounded*. It turns out this type of correction only works when the error being corrected is symmetric. The correction along the *cs* axis is symmetric, but the correction along the *lr*=0 boundary is not. A better way to do this correction is through an interpolation, as was done in the theory in the previous paper for the LRR matrix element along the *lr* axis. In practice the interpolation can be performed through two additional lookup tables at little computational cost. The equations for this interpolation will be shown later. The LRR matrix element is correctly interpolated along the *lr* axis. However as can be seen in FIG. 15, there is a discontinuity along the *cs* axis. This discontinuity was due to a programming error in the decoder version 1.11. The

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theory presented in the previous patent application produces the matrix element shown in FIG. 16.

FIG. 16 shows the left rear right matrix element as it should have been implemented in the decoder of version 1.11 according to the previous patent application, in which the discontinuity has been removed. With the correct interpolation and implementation, there are no discontinuities at the lr=0 boundary.

Turning now to the TV sound matrix, FIG. 17 shows the LFL matrix element as implemented in U.S. Patent No. 4,862,502, and in Dolby Pro-Logic, scaled so that the maximum value is 1.

Assuming that cs and lr are the steering directions in degrees in the center/surround and left/right axis respectively, in the 1989 patent the equations for the front matrix elements were given as:

In the left front quadrant,

$$LFL = 1 - 0.5*G(cs) + 0.41*G(lr) \quad ...(100) \qquad LFR = -0.5*G(cs) \quad ...(101)$$

In the right front quadrant,

$$LFL = 1 - 0.5*G(cs) \quad ...(102)$$

$$LFR = -0.5*G(cs) \quad ...(103)$$

In the left rear quadrant,

$$LFL = 1 - 0.5*G(-cs) + .41*G(lr) \quad ...(104)$$

$$LFR = 0.5*G(-cs) \quad ...(105)$$

In the right rear quadrant,

$$LFL = 1 - 0.5*G(-cs) \quad ...(106)$$

$$LFR = 0.5*G(-cs) \quad ...(107)$$

The function $G(x)$ is described in U.S. Patent No. 4,862,502, and specified in U. S. Patent No. 5,307,415. It varies from 0 to one as x varies from 0 to 45 degrees. In the previous patent application it is shown to be equal to $1-\tan(|r|/|l|)$ where $|r|$ and $|l|$ are the right and left input amplitudes.

In FIG. 18, the LFR matrix element from U.S. Patent No. 4,862,502 and Dolby

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Pro-Logic is shown, scaled by 0.71 so the minimum value and maximum values are ±0.5.

In the previous patent application No. 08/684,948 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 of the LFL and LFR matrix elements should be a constant. It was pointed out in the application that this goal should be relaxed in the direction of the steering — that is, when the steering is full left, the sum of the squares of the matrix elements should rise by 3dB. We can see that the above matrix elements do not meet this requirement.

In FIG. 19 is shown the square root of the sum of the squares of LFL and LFR from U.S. Patent No. 4,862,502, scaled so that the maximum value is one. Notice that the value is constant at 0.71 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. The rear direction is identical to the center direction, but is not easily seen due to the perspective in this view.

The previous patent application No. 08/684,948 corrected this amplitude error by replacing the function G(x) in the matrix equations with sines and cosines:

For the left front quadrant

$$LFL = \cos(cs) + 0.41*G(lr) \quad ...(108)$$

$$LFR = -\sin(cs) \quad ...(109)$$

For the right front quadrant

$$LFL = \cos(cs) \quad ...(110)$$

$$LFR = -\sin(cs) \quad ...(111)$$

For the left rear quadrant

$$LFL = \cos(-cs) + 0.41*G(lr) \quad ...(112)$$

$$LFR = \sin(-cs) \quad ...(113)$$

For the right rear quadrant

$$LFL = \cos(-cs) \quad ...(114)$$

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$$LFR = \sin(-cs) \quad \ldots(115)$$

In FIG. 20 is shown the square root of the sum of the LFL and LFR matrix elements from the previous patent application, No. 08/684,948, scaled so the maximum value is 1. Note the constant value of 0.71 in the entire right half of the plane, and the gentle rise to one toward the left vertex.

The decoder version 1.11 made several changes to these matrix elements. Keeping the basic functional dependence, an additional boost was added along the cs axis in the front, and a cut was added along the cs axis in the rear. The reason for the boost was to improve the performance with stereo music which 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,

$$LFL = (\cos(cs) + 0.41*G(lr))*\text{boost1}(cs) \quad \ldots(116)$$

$$LFR = (-\sin(cs))*\text{boost1}(cs) \quad \ldots(117)$$

For the right front quadrant,

$$LFL = (\cos(cs))*\text{boost1}(cs) \quad \ldots(118)$$

$$LFR = (-\sin(cs))*\text{boost1}(cs) \quad \ldots(119)$$

For the left rear quadrant,

$$LFL = (\cos(-cs) + 0.41*G(lr))/\text{boost}(cs) \quad \ldots(120)$$

$$LFR = (\sin(cs))/\text{boost}(cs) \quad \ldots(121)$$

For the right rear quadrant,

$$LFL = (\cos(cs))/\text{boost}(cs) \quad \ldots(122)$$

$$LFR = (\sin(cs))/\text{boost}(cs) \quad \ldots(123)$$

The function $G(x)$ was defined in the previous patent application. It is equal to $G(x) = 1-\tan(45-x)$, and is the identical function used in the Dolby matrix.

The function boost1(cs) as used in the decoder version 1.11 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. (See corr1 in the pseudocode below.) Boost(cs) is

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given by corr(x) in the code below, in which comments are preceded by the % symbol:

```
% calculate a boost function of +3dB at 22.5 degrees
% corr(x) goes up 3dB and stays up.  corr1(x) goes up then down again
for x = 1:24;  % x has values of 1 to 24
  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
  corr(x) = 1.41;
  corr1(x) = corr(48-x);
end
```

These equations produce the surface shown in FIG. 21 for the LFL matrix element in decoder version 1.11. 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.

The boost along the boundary creates a panning error. Also the cut in the rear direction is not optimal. There are two areas where the performancecan be improved. The first 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. 21 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.

As explained in a previous disclosure, 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

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no net left or right steering. However the method chosen to implement this increase in the decoder of version 1.11 was independent of the value of *lr*, and resulted in an increase in level when a strong signal panned across the boundary.

The problem is that 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 code:

We assume both *lr* and *cs* > 0 for a signal in the left front quadrant, also assuming that *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
```

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.

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. (This code is written in a modified Matlab, where variables are multivalued vectors. Comments are preceded by the % symbol..)

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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_tbl = sin($a1$);

cos_tbl = cos($a1$);

cos_tbl_plus = cos($a1$).*corr1(a+1);

cos_tbl_plus = cos_tbl_plus-cos_tbl; % this is the one we use

cos_tbl_minus = cos($a1$)./corr(a+1);

sin_tbl_plus = sin($a1$).*corr1(a+1);

sin_tbl_plus = sin_tbl_plus-sin_tbl; % this is the one we use

sin_tbl_minus = sin($a1$)./corr(a+1);

sin_tbl_plus and cos_tbl_plus are the difference between a plain sine and cosine, and the boosted sine and cosine. We now define

LFL = cos(cs) + 0.41*G(lr) + cos_tbl_plus(bcs) ...(124)

LFR = - sin_tbl_plus(bcs) + sin(cs) ...(125)

LFL and LFR in the front right quadrant are similar, but without the +0.41*G term. These new definitions lead to the following matrix elements.

The steering in the rear quadrant is not optimal either. In the curve above when the steering is toward the rear the matrix elements are given by

LFL = cos_tbl_minus(−cs) + 0.41*G(−cs) ...(126)

LFR = sin_tbl_minus(−cs) ...(127)

These matrix elements are very nearly identical to the Dolby matrix elements. Consider the case when a strong signal pans from left to rear. The Dolby 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. However in a decoder according to the present invention it is 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

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the signal pans further to full rear. The matrix elements used in the version 1.11 decoder — 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.

This performance can be improved by altering the LFL and LFR matrix elements in the left rear quadrant. Notice here we are concerned with how the matrix elements vary along the boundary between left and rear. The mathematical method given in the AES paper ("Multichannel Matrix Surround Decoders for Two-Eared Listeners," David Griesinger, AES preprint No. 4402, October, 1996) 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:

$$LFL = \cos(t)*F(t) -/+ \sin(t)*(\mathrm{sqrt}(1-F(t)\text{^}2)) \quad ...(128)$$

$$LFR = -(\sin(t)*F(t) +/- \cos(t)*(\mathrm{sqrt}(1-F(t)\text{^}2))) \quad ...(129)$$

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

$$LFL = \cos(t)*\cos(4*t)+\sin(t)*\sin(4*t) \quad ...(130)$$

$$LFR = -(\sin(t)*\cos(4*t)-\cos(t)*\sin(4*t) \quad ...(131)$$

These elements work well — the front left output is reduced smoothly to zero as t varies from 0 to −22.5 degrees. FIG. 22 shows that the left front left matrix element has the correct amplitude along the left to center boundary, as well as along the center to right boundary.

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,

$$LFL = -\sin(t) \quad ...(132)$$

$$LFR = \cos(t) \quad ...(133)$$

Note that these matrix elements are a far cry from the matrix elements along the lr=0 boundary, where in the previous patent application the values were

$$LFL = \cos(cs) \quad ...(134)$$

$$LFR = \sin(cs) \quad ...(135)$$

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FIG. 23 shows the behavior of LFL and LFR along the rear boundary between left and full rear. The slight glitch is due to the absence of a point at 22.5 degrees.

We need a method of smoothly transforming the above equations into the equations along the boundary as *lr* and *cs* approach the boundary. A linear interpolation could be used. In the processor typically used in these decoderss, where multiplies are expensive, a better strategy is to define a new variable – the minimum of *lr* and *cs*:

```
% 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_tbl(x) = (cos(ax)–sin(ax))/(cos(ax)+sin(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
```

We then define LFL and LFR in this quadrant as:

LFL = cos(*cs*)/(cos(*cs*)+sin(*cs*)) – front_boundary_table(*bp*) + 0.41*G(*lr*) ...(136)

LFR = sin(*cs*)/(cos(*cs*)+sin(*cs*)) + front_boundary_table(*bp*) ...(137)

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.

In the right rear quadrant,

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$$LFL = \cos(cs)/(\cos(cs)+\sin(cs)) \quad ...(138)$$

$$LFR = \sin(cs)/(\cos(cs)+\sin(cs)) \quad ...(139)$$

FIG. 24 shows the left front left matrix element as viewed from the 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. However, along the lr=0 axis and in the right rear quadrant the function is identical to the Dolby matrix.

FIG. 25 shows the left front right matrix element. Note the large peak in the left to rear boundary. This works in conjunction with the LFL matrix element to reduce the front output to 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.

One of the major design goals of the design of the improved matrix of the present invention 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 relaxed 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.

We can test the success of our design by plotting the square root of the sum of the squares of the matrix elements.

FIG. 26 shows the root mean squared sum of LFL and LFR, using the new design. (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.) 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. This peak is a result of the deliberate boost of the left and right outputs during half-front steering. Note that

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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.

FIG. 27 shows the square root of the sum of the squares of LFL and LFR including the correction to the rear level, viewed from the left rear. 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.

The next concern addressed in the present invention is correcting the values of the rear matrix elements during front steering.

The rear matrix elements in Griesinger's 1989 U.S. Patent No. 4,862,502 are given by:

For the front left quadrant:

$$LRL = .71 - .71*G(lr) + .41*.71*G(cs) \quad ...(140)$$

$$LRR = -.71*G(lr) + .41*.71*G(cs) \quad ...(141)$$

For the rear left:

$$LRL = .71 - .71*G(lr) + .41*.71*G(-cs) \quad ...(142)$$

$$LRR = .71*G(lr) + .41*.71*G(-cs) \quad ...(143)$$

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

The rear matrix elements in the Dolby Pro-Logic are

For the front left quadrant:

$$LRL = .71 - .71*G(lr) + .41*.71*G(cs) \quad ...(144)$$

$$LRR = -.71*G(lr) + .41*.71*G(cs) \quad ...(145)$$

For the rear left:

$$LRL = .71 - .71*G(lr) \quad ...(146)$$

$$LRR = .71*G(lr) \quad ...(147)$$

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

A brief digression on the surround level in Dolby Pro-Logic — note that the Dolby elements are identical in the front but do not include the boost dependent on *cs* in the rear. This difference is in fact quite important. The equations above

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somewhat disguise the way these decoders are actually used. 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 one. In addition, when the decoder is finally put to use, the gain of each output to the loudspeaker is adjusted so the sound power is the same at the listening position when a signal is presented to the decoder which has been encoded from the four major directions – left, center, right, and rear. 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.

The lack of a level boost in the rear direction in the Dolby decoder means that during the calibration procedure the gain of the rear outputs will be raised by 3dB relative to the other outputs. In fact, for the Dolby decoder in practice:

$$LRL = 1 - G(lr) \quad ...(148)$$

$$LRR = -1 \quad ...(149)$$

The difference is not trivial. When the front elements are scaled so they have a maximum value of one when there is full steering in one direction, we find that during unsteered conditions the elements from the 1989 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 1989 outputs. Note that the calibration procedure results in a matrix which 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*(Ain + Bin), and the Pro-Logic matrix does not meet this specification.If there are two speakers sharing this output each will be 3dB softer, which will make all five speakers have approximately equal sound power when the decoder inputs are uncorrelated. When the matrix elements from the 1989 patent are used, the same calibration procedure results in 3dB less sound power from the rear when the decoder inputs are uncorrelated. The elements for the rear outputs in the new design include a form of level boost when the outputs are

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fully steered – either to the left or right sides – or completely to the rear. Thus they follow the 1989 patent in terms of their surround level when they are unsteered.

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

Ain = Lin + .71*Cin ...(150)

Bin = Rin + .71*Cin ...(151)

When Ain and Bin are played in stereo, the sound power in the room will be proportional to Lin^2 + Rin^2 + Cin^2. If all three components have roughly equal amplitudes, the ratio of the center component to the left plus right component will be 1:2.

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 Cin to Lin and Rin. 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. that the unsteered condition of the matrix is identical to Dolby Surround, then the center matrix elements should have the shapes described by FIGs. 28 and 29.

In FIG. 28, the solid curve shows the center matrix value as a function of *cs* +1 in dB, assuming sound power ratios identical to stereo, and using Dolby matrix elements with 3dB less power in the rear than typically used. The dotted curve is the actual value of the center matrix elements in Pro-Logic. 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.

Similarly, in FIG. 29, the solid curve shows the value of the center matrix elements assuming equal power ratios to stereo given the matrix elements and calibration actually used in Dolby Pro-Logic. The dotted curve shows the actual values of the center matrix elements in Pro-Logic. Notice that the actual values

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are more than 3dB too low for all steerings.

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 often 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, which is identical to the unsteered condition of the previous figure.

The present invention also includes the creation of two independent rear outputs, as described below.

The major problem with both the 1989 patent elements and the Dolby elements is that there is only a single rear output. The disclosure given in Griesinger's 1991 U.S. Patent No. 5,136,650 created two independent outputs at the sides, and the math in that patent was incorporated in the front left quadrant of the U.S. Patent Application No. 08/684,948 of July 1996. 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:

$$LRL = 1 - GS(lr) - 0.5*G(cs) \quad ...(152)$$

$$LRR = - 0.5*G(cs) - G(lr) \quad ...(153)$$

As can be seen, these matrix elements are very similar to those of the 1989 Griesinger U.S. Patent 4,862,502, but with the addition of a $G(lr)$ term in LRR and a $GS(lr)$ 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 $GS(lr)$, using 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 $G\text{^}2(lr)$, although a more complicated formula is given in the 1991 patent (5,136,650). The two formulae can be shown to be identical.

In the July 1996 application these elements are corrected by being given a

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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.

For the right front quadrant the matrix elements are identical to the rear elements given in the 1989 patent (4,862,502), and were implemented in the version 1.11 decoder.

The left front quadrant problems can be seen in FIG. 30 which shows the square root of the sum of the squares of LRL and LRR, using the matrix elements implemented in decoder version 1.11 as given above. Note that in the front left quadrant, there is a 3dB dip along the line from the middle to the left vertex and a nearly 3dB boost in the level along the boundary between left and center. The mountain range in the rear quadrant will be discussed later. This drawing includes the "TV matrix" dip of 3dB in the middle of the plane, which is hard to see in this projection.

First we consider the dip in the sum of the squares along the *cs*=0 axis. This dip exists because of the use of G(*lr*) in LRR. This choice was entirely arbitrary — although it makes implementation in analog circuitry easy. 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 that:

$$LRL = \cos(cs) - GS(lr) \quad \text{...(154)}$$

$$LRR = -\sin(cs) - GR(lr) \quad \text{...(155)}$$

We want the sum of the squares to be one along the cs=0 axis, so

$$(1-GS(lr))^2 + (GR(lr))^2 = 1 \quad \text{...(156)}$$

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

$$LRL*\cos(t) + LRR*\sin(t) = 0 \quad \text{...(157)}$$

These two equations result in a messy quadratic equation for GR and GS,

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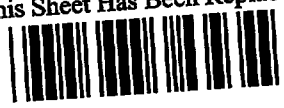

which can be solved numerically.

Figure 31 shows the numerical solution for GS and GR for constant power level along the *cs*=0 axis, and zero output along the boundary between left and center.

Use of GS and GR as shown results in a large improvement along the *lr*=0 axis, as intended. However, the peak in the sum of the squares along the boundary between left and center remains. In a practical design it is probably not very important to compensate for this error, but we can attempt to do so with the following strategy. We will divide both matrix elements by a factor which depends on a new combined variable based on *lr* and *cs*. Call the new variable xymin. (In practice the divide can be replaced by a multiply by the inverse of the factor described below.) A procedure for defining xymin (in Matlab notation) is:

```
% 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. If we multiply it by four, it will vary from zero to 90 degrees, and can be used below.

In the front left quadrant

$$LRL = (\cos(cs) - GS(lr))/(1+.29*\sin(4*xymin)) \quad ...(158)$$

$$LRR = (-\sin(cs) - GR(lr))/(1+.29*\sin(4*xymin)) \quad ...(159)$$

In the front right quadrant,

$$LRL = \cos(cs) \quad ...(160)$$

$$LRR = -\sin(cs) \quad ...(161)$$

As explained in the previous paper, these elements are additionally multiplied by the "TV matrix" correction, which reduces the amplitude when the steering is near the middle of the plane. We will call the correction for TV matrix

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tvcorr(|*lr*|+|*cs*|). Tvcorr(|*lr*|+|*cs*|) is –3dB at zero, and 1 when the argument is 22.5 degrees and higher.

FIG. 32 shows the square root of the sum of the squares of LRL and LRR using the new values for GR and GS. This factor shows up in FIG. 32 as a valley centered on zero steering. Note that except for the valley created by the "TV matrix" correction, the sum of the squares is close to one and continuous.

In the present invention, the "TV matrix" correction has been modified to depend only on the absolute value of *lr* when *cs* is frontal. This will cause the surface above to remain at .71 along the *lr*=0 axis in the front. In this case the correction for TV matrix becomes tvcorr(|*lr*|). Tvcorr(|*lr*|) is -3dB at zero, and 1 when the argument is 22.5 degrees and higher.

The rear matrix elements during rear steering were previously discussed with reference to FIGs. 14-16. The rear matrix elements given in the 1991 patent (5,136,650) were not appropriate to a 5 channel decoder, and were modified heuristically in Lexicon's CP-3 product. The July 1996 patent application (08/684,948) used a mathematical method to derive these elements along the boundary of the left rear quadrant. As described in the previous paper this procedure resulted in discontinuities along the *lr*=0 axis, and along the *cs*=0 axis. In the decoder version 1.11 these discontinuities were repaired (mostly) by additional corrections to the matrix elements, which preserved their behavior along the steering boundaries. The software error in the LRR element was shown earlier in this application, as was the small discontinuity along the *cs*=0 boundary for the LRL element (see FIGs. 14, 15).

For the new elements described here these errors have been corrected, first by using an interpolation along the *cs*=0 boundary for LRL, where the value is made to match the value of GS(*lr*) when *cs* is zero, and smoothly rises to the value given by the previous math as *cs* increases negatively toward the rear. In the new software, LRR interpolates along the *cs*=0 axis to GR(*lr*). In the decoder version 1.11 LRR interpolated to G(*lr*).

We will first consider the Left Rear Left and Left Rear Right matrix elements when the steering is neutral or anywhere between full right and right

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rear. That is, *lr* can vary from 0 to –45 degrees, and *cs* can vary from 0 to –22.5 degrees.

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.

The matrix elements given in the 1991 patent (5,136,650) 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. We will define two new functions which are simply equal to sin and cosine of *cs* over this range.

LRL = cos(-cs) = sri(-cs) ...(162)

LRR = sin(-cs) = sric(-cs) ...(163)

To complete LRL and LRR over the range of cs = 0 to -22.5 we must add a gain reduction for the "TV matrix" mode. Once again in the "TV matrix" mode we desire 3dB less output when steering is neutral, but rising to the value in the "Logic 7" decoder version when the steering is more than 22.5 degrees to the rear. Performance is somewhat improved by making this reduction sensitive to the sum of $|lr|$ and $|cs|$. This is achieved in the current design by reducing both the RRR and RRL elements by 3dB when the sum is zero, and raising them back to their original values as the sum goes to 22.5 degrees. Once again, the slope of this gain change is relatively arbitrary, as long as both RRR and RRL are altered in the same way. We will call the correction for TV Matrix tvcorr($|lr|+|cs|$). Tvcorr($|lr|+|cs|$) is –3dB at zero, and 1 when the argument is 22.5 degrees and higher.

LRL = cos(–*cs*)*tvcorr($|lr|+|cs|$) = sri(–*cs*)*tvcorr($|lr|+|cs|$) ...(164)

LRR = sin(–*cs*)*tvcorr($|lr|+|cs|$) = sric(–*cs*)*tvcorr($|lr|+|cs|$) ...(165)

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 cos(x). We will use these functions again in defining the Left Rear matrix elements during Left steering.

Now consider the same matrix elements as *cs* becomes greater than –22.5

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degrees. As mentioned in Griesinger's earlier AES paper and patents and applications previously cited, LRL should rise to one or more over this range, and LRR should decrease to zero. Simple functions fulfill this:
(remember that *cs* is negative)

LRL = (cos(45+cs) + rboost(-cs)) = (sri(-cs) + rboost(-cs)) ...(166)

LRR = sin(45+cs) = sric(-cs) ...(167)

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

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 45 to 22.5 or to zero. The matrix elements given in the amended application of November 1996 perform this, but as we showed earlier, there are problems with continuity at the *cs* = 0 boundary (see FIG. 15.)

For the decoder version 1.11 a solution was found which uses functions of one variable and several conditionals. In Griesinger's AES paper and the patent applications cited previously the problem at the *cs* = 0 boundary arises because on the forward side of the boundary the LRL matrix element is given by GS(*lr*). On the rear side the function given by the AES paper has the same end points, but is different between them.

The mathematical method in the AES paper provides the following equations for the Left Rear matrix elements over the range 22.5 < *lr* < 45:
(remember that *t* = 45–*lr*)

LRL = cos(45–*lr*)*sin(4*(45–*lr*))–sin(45–*lr*)*cos(4*(45–*lr*))

= sra(*lr*) ...(168)

LRR = –(sin(45–*lr*)*sin(4*(45–*lr*))+cos(45–*lr*).*cos(4*(45–*lr*)))

= –srac(*lr*) ...(169)

If *cs* <= 22.5, *lr* can still vary from 0 to 45. The AES paper defines LRL and LRR when *lr* has the range 0 < *lr* < 22.5, as shown in Figure 6 in the AES paper.

LRL = cos(*lr*) = sra(*lr*) ...(170)

LRR = -sin(*lr*) = -srac(*lr*) ...(171)

The two functions sra(*x*) and srac(*x*) are defined for 0 < *lr* < 45.

In the version 1.11 decoder the following technique is used to fix the

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discontinuity across the *cs*=0 boundary. In the AES paper near *cs*=0, LRL and LRR are both functions of a single variable. To fix the lack of continuity along the *cs*=0 boundary we add a function of a composite of *lr* and *cs*. The new variable is lr_bounded, the bounded difference between *lr* and *cs*. The definition of this variable is sufficiently complicated that I will present it in pseudo-c (MATLAB) form.

```
lr_bounded = lr - cs;  % find the difference
if (lr_bounded <0) % % only if lr > cs
  lr_bounded  = 0;
if (45-|cs| < lr_bounded) % use the smaller of the two values
  lr_bounded  = 45-cs;
```

We define a new function which is equal to the difference between the previous equations when *cs*=0. This is rear_active_correct(lr_bounded).

For $0 < x < 45$

$$\text{Rear_active_correct}(x) = \text{sra}(x) - (1 - \text{GSL}(x)) \quad \ldots(172)$$

$$\text{LRL} = (\text{sri}(cs) + \text{sra}(lr) - \text{rear_active_correct}(\text{lr_bounded}) - 1) * \text{tvcorr}(|lr| + |cs|) \quad \ldots(173)$$

The important point about this method is that it works when $lr < 22.5$, but it does not work when *lr* is larger. A better technique, which was not used in the version 1.11 decoder, is the interpolation technique which is used for LRR.

The version 1.11 decoder's LRR coefficient uses a better technique. 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.

The choice used in the version 1.11 of "logic- 7" decoder — 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 $\text{srac}(lr)$.

There is also the possibility of a discontinuity along the $lr=0$ axis. We can solve this by adding a term to LRR which is found by using cs_bounded. The term

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is simply sric(cs_bounded). This term will insure continuity across the *lr*=0 axis.

First we define cs_bounded

```
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))*tvcorr(|lr|+|cs|);
for cs = 15 to 22.5
    LRR = (-srac(lr) + sric(cs_bounded))*tvcorr(|lr|+|cs|);
```

In the decoder according to the present invention LRL is computed with interpolation, just as for LRR:

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

As the steering goes from left rear to full rear the elements follow the ones given in the AES paper, with the addition of the corrections for rear loudness.

For cs > 22.5, lr < 22.5

$$LRL = (sra(lr) + sri(cs) + rboost(cs)) \quad ...(174)$$

$$LRR = -srac(lr) + sric(cs_bounded) \quad ...(175)$$

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.

The next improvements to discuss is the implementation of the Center matrix elements in the present invention.

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The center matrix elements in the decoder version 1.11 have major differences from the center elements in the July 1996 patent application. The '89 patent and Dolby Pro-Logic have the following matrix elements.

For front steering

$$CL = 1 + .41*G(cs) - G(lr) \quad ...(176)$$

$$CR = 1 + .41*G(cs) \quad ...(177)$$

For rear steering

$$CL = 1 - G(lr) \quad ...(178)$$

$$CR = 1 \quad ...(179)$$

Since the matrix elements have a symmetry about the left/right axis, the values of CL and CR for right steering can be found by swapping CL and CR.

FIG. 33 shows the Center Left (CL) matrix element of the four channel and Dolby Pro-Logic decoder plotted in three dimensions. This is also the graph of the Center Right matrix element if we swap left and right. 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.

In the July 1996 patent application these elements are replaced by sines and cosines:

For front steering

$$CL = \cos(45-lr)*\sin(2*(45-lr))-\sin(45-lr)*\cos(2*(45-lr)) + .41*G(cs) \quad ...(180)$$

$$CR = \sin(45-lr)*\sin(2*(45-lr))+\cos(45-lr)*\cos(2*(45-lr))+ .41*G(cs) \quad ...(181)$$

These equations were never implemented. The version 1.11 implementation is based on the steering in the 1989 patent, but with a different scaling, and a different function of *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 (.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 version 1.11 decoder was chosen relatively arbitrarily.

In version 1.11 the new boost function of *cs* starts at zero as before, and

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rises with *cs* in such a way as CL and CR increase 4.5dB as *cs* goes from zero to 22.5 degrees. In version 1.11 the increase is a constant number of dB for each dB of increase in *cs*. The 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 newmatrix 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.

FIG. 33 shows the Center Left matrix element in the Logic 7 decoder version 1.11. Note that, relative to the plot of FIG. 33, 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.

The solution for the center used in version 1.11 is 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 matrix 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 value above gives too low an output from the center speaker when the pan is half-way between.

We now consider the center channel in the new design.

The center channel output must be derived from the A and B inputs to the decoder. While it is possible to remove either the A or the B input from the center channel output using matrix techniques, any time the steering is not biased either left or right, the center channel must reproduce the sum of the A and B inputs with some gain factor, either a boost or a cut. How loud should it be?

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The answer to 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 forward with some kind of cross mixer, such as a stereo width control, the matrix elements given above (the 1989 elements of U.S. Patent No. 4,862,502, the July 1996 patent application elements, the version 1.11 decoder elements, and the new ones according to the present invention) all completely restore the original separation. The version 1.11 elements (with the level boost when *cs* is approximately 22.5) also restore the original separation.

However, if the input to the decoder consists of uncorrelated left and right channels to which an unrelated center channel has been added:

$$A = Lin + .71*Cin \quad ...(182)$$

$$B = Rin + .71*Cin \quad ...(183)$$

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

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 signal is always derived from the sum of the A and B inputs, and as its amplitude is raised the amplitude of the Lin and Rin signals must rise along with the amplitude of the Cin signal.

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

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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.

If we want to design the matrix so the ratio of the amplitudes of Lin, Rin, and Cin are preserved when switching between stereo reproduction and matrix reproduction, the sound power of the Cin 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 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.

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.

In FIG. 35, the solid curve shows the center attenuation needed for the new LFL and LFR 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. The dotted curve shows the center values for a standard decoder.

As can be seen in the solid curve, 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. This steep change in amplitude is audible in practice. Also, although the relative balance of the center channel information in a popular recording is well preserved, if one is standing close to the center speaker the sudden changes in level can be annoying. Furthermore the loudness of the center channel is extreme. We tested this curve and found the center balance to be excellent, but the front sound stage was dominated by the center loudspeaker, and left-right separation was minimal.

There is a better solution. The center attenuation shown in figure 35 is derived assuming the matrix elements previously given for LFL and LFR. What if

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we used different elements? Specifically, do we need to be aggressive about removing the center component from the left and right front outputs?

Listening tests show that the version 1.11 decoder elements are needlessly aggressive about removing the center component. Acoustically there is no need that they should do so. The energy removed from them must be given to center loudspeaker. If we don't remove this energy it comes from the left and right front speakers, and the soundfield is similar. The trick is to achieve a balance between the center power in the left and right speakers, and the center power in the center speaker.

We create the optimal system by first choosing a gentler function for the increase of level in the center output as cs increases along the lr=0 axis. We can then solve for the decrease in level needed in the left and right front outputs to keep the power of the Cin component in the soundfield in the room constant.

Let us assume that the center channel is reduced in level by 4.5 dB below the level in Griesinger's 1989 decoder (4,862,502), or –7.5dB total attenuation. This is a factor of 0.42.

For front steering,

$$CL = .42 - .42*G(lr) + GC(cs) \quad ...(184)$$

$$CR = .42 + GC(cs) \quad ...(185)$$

For rear steering,

$$CL = .42 - .42*G(lr) \quad ...(186)$$

$$CR = .42 \quad ...(187)$$

Several functions were tried for GC(cs). The one given below is specified in terms of the angle cs in degrees, and was obtained by some trial and error.

In MATLAB notation:

```
center_max = .65;
center_rate = .75;
center_max2 = 1;
center_rate2 = .3;
center_rate3 = .1;
```


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```
if (cs < 12)
  gc(cs+1) = .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
```

This function is plotted in FIG. 36 as the solid curve.

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 Cin 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 Lin and Rin components, and should have the current shape at the left to center boundary, as well as the right to center boundary.

We assume that:

$$LFL = GP(cs) \quad ...(188)$$

$$LFR = GF(cs) \quad ...(189)$$

$$CL = .42 - .42*G(lr) + GC(cs) \quad ...(190)$$

$$CR = .42 + GC(cs) \quad ...(191)$$

The power from the front left and right speakers is given by:

$$PLR = (GP^2+GF^2)*(Lin^2+Rin^2) + (GP-GF)^2*Cin^2 \quad ...(192)$$

Power from the center speaker is:

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PC = GC^2*(Lin^2+Rin^2) + 2*GC^2*Cin^2 ...(193)

Power from the rear speakers depends on the matrix elements we use. We will assume that 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, then,

PREAR = (.71*(cos(*cs*)*(Lin+.71*Cin) - sin(*cs*)*(Rin+.71*Cin)))^2 ...(194)

If we assume that Lin^2 and Rin^2 are approximately equal, for two speakers,

PREAR = .5*Cin^2*((cos(*cs*)-sin(*cs*))^2 + Lin^2 ...(195)

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

PT = (GP^2 + GF^2 + GC^2)*(Lin^2 + Rin^2)
+ ((GP – GF)^2 + 2*GC^2)*Cin^2 + PREAR
...(196)

The ratio of Cin power to Lin and Rin power is therefore (assuming equal Lin^2 and Rin^2):

RATIO = ((GP^2 + GF^2 + GC^2)*(Lin^2 + Rin^2) + ((GP–GF)^2
+ 2*GC^2)*Cin^2 + PREAR)/PREAR
= (((gp(*cs*)–gf(*cs*))^2 + 2*gc(*cs*)^2 +.5*(cos(*cs*)–sin(*cs*))^2)*Cin^2/
2*(gp(*cs*)^2 + gc(*cs*)^2 + gf(*cs*)^2) + 1) * Lin^2
= (Cin^2/Lin^2)*((gp(*cs*)–gf(*cs*))^2 + 2*(gc(*cs*)^2) + .5*(cos(*cs*)
– sin(*cs*))^2) / 2*(gp(*cs*)^2 + gc(*cs*)^2 + gf(*cs*)^2) + 1) ...(197)

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

RATIO = (Cin^2/Lin^2)*0.5 ...(198)

If this ratio is to be constant regardless of the value of Cin^2/Lin^2 for the active matrix, which is desirable, then:

((gp(*cs*)-gf(*cs*))^2 – 2*gc(*cs*)^2) + .5*(cos(*cs*)–sin(*cs*))^2)
= (gp(*cs*)^2 + gc(*cs*)^2 + gf(*cs*)^2) - .5) ...(199)

Equation (199) can be solved numerically. If we assume the GC value above, and GP = LFL as before, then FIG. 36 shows the resulting values for GF in the solid curve, sin(*cs*)*corr1 (the previous LFR element) in the dashed curve, and

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sin(*cs*) in the dotted curve. Note that GF remains close to zero until *cs* reaches 30 degrees, and then increases sharply. In practice we arbitrarily increase GF beyond 30 degrees to reach the value of 0.71 like the dashed and dotted curves. This causes complete cancellation of the center channel in the left and right during strong steering. Also, GF must smoothly interpolate to the previous value along the boundaries. These curves all have a negative sign in practice.

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);
```

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```
    if (bcs <1)  %> 46)
      bcs = 1;  %46;
    end
  end
```

The LFR element can now be written:

```
% 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
```

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.

FIG. 37 shows the left front right (LFR) matrix element with the correction for center level along the *lr*=0 axis. Note that the value is zero in the middle of the plane (no steering) and remains zero as *cs* increases to 22.5 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.

We now consider the panning error in the center output.

FIG. 38 shows the center left (CL) matrix element with the new center boost function GC(*cs*). Note the correction for panning along the boundary between left and center.

As it turns out, the new center function (if we write it this way):

$$CL = .42 - .42*G(lr) + GC(cs) \quad \text{...(200)}$$

$$CR = .42 + GC(cs) \quad \text{...(201)}$$

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 the July

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1996 patent application give a smooth function of cos(2*cs) along the left boundary, which creates smooth panning between left and center. We would like our new center function to have similar behavior along this boundary.

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

center_fix_tbl = .8*(corr1-1);

CL = .42 - .42*G(lr) + GC(cs) + center_fix_table(xymin)

CR = .42 + GC(cs) + center_fix_table(xymin)

FIG. 39 shows the levels of the center output and the left output as a signal pans from center to left. Note that with the correction the panning of the center, while not perfect, is reasonably close to the inverse of the left output. (The values on the *cs* axis are inverted).

We now consider a new five channel encoder (termed "Logic 7") which is designed to operate correctly with the decoder specified by the equations and algorithms given above.

There are two major goals for this 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 7 decoder according to the present invention with minimal inaccuracy. 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 five-channel mix. The apparent position of the sound source in stereo should also be as close as possible to the apparent position in the five channel original.

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 single adjustment of the 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

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mixed into the output of the encoder in such a way that the energy is preserved. That is, the total energy at the output of the encoder should be the same, regardless of which input is being driven. This technique will include most film sources and 5-channel music sources where instruments have been assigned to all five loudspeakers. Although such music sources are not common at the present time, it is the inventor's opinion that they will become common in the future. But music recordings where the foreground instruments are placed in the front three channels, with primarily reverberation in the rear channels, require a different encoding technique.

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. As we will see later, 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 continually -6dB or less compared to the front channels, the surround gain is gradually lowered 3dB to correspond to the European standard.

During tests with the IRT in Munich it was found that a particular tape was encoded incorrectly by the encoder described in the AES paper (preprint No. 4402). A new architecture was developed to solve the problem with this tape. Although the encoding of this particular tape was only marginally improved, the new encoder is superior in its performance on a wide variety of difficult material. The original encoder was developed first as a passive encoder, and it performed reasonably well with a variety of input signals. The new encoder will also work in a passive mode, but is primarily intended to work as an active encoder (i.e. one in which the encoding depends on the types of signal presented to its inputs.) The active circuitry corrects several small errors inherent in the design. However, even without the active correction, the performance is better than that of the

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previously described encoder.

After extensive listening, several 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 inputs 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, there is a net rearward bias which can tend to reduce the signal power of dialog in the center channel. This can be important in a film, where dialog intelligibility is of paramount importance. The new encoder compensates for this effect by raising the center channel input to the encoder slightly under these conditions.

The new encoder, shown in block schematic form in FIG. 40, handles the left, center and right channels identically to the previous design, and identically to the Dolby encoder, providing that the center attenuation function *fcn* in attenuator **302** is equal to 0.71 or -3dB.

In accordance with the previous designs of encoder shown in FIGs 10 and 11, the left (L), center © and right (R) signals are presented to input terminals **50**, **52**, and **54** of the encoder circuitry, respectively. The left side (LS) and right side (RS) signals are presented to the input terminals **62** and **64** respectively. An additional signal LFE (for low frequency effects in a 5 + 1 mix) is applied to a new input terminal **370**. The C and LFE signals pass through attenuator/gain elements **372**, **374**, respectively, where C is amplified by the factor *fcn* and LFE by a factor of 2.0. These signals are each applied to both the summing circuits **278** and **282**. The L signal is applied directly to summing circuit **278** and the R signal is similarly applied to summing circuit **282**. The surround signals are also applied to these summing circuits, but only after some manipulation, which appears to be more complex than it really is. In the surround channel attenuators **376**, **378**, **380**, and **382**, the functions *fc()* and *fs()* direct the surround channels either to a path (through phase shift elements **234** and **246**) with a 90 degree phase shift

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relative to the front channels (which proceed through the phase shifters **286** and **288**) or to a path with no relative phase shift. In the basic encoder, *fc* is one and *fs* is zero, so that the active path is through the 90 degree phase shifters. Thus, the LS signal passes unchanged through block **376** to attenuator **396**, where it is multiplied by a 0.91 factor, then passes to the adder **406**, where it is mixed with the cross-coupled RS signal from the attenuator **404** which has gain of $-crx$. The value of *crx* is typically 0.38. It controls the amount of negative cross feed for each surround channel. The signal then passes through a 90 degree phase shifter **234** and an adder **276**, where it is mixed with the other signals from phase shifter **286** to this adder, and passes to the output terminal **44** as the "A" signal. As in the previous encoder, when there is only an input to one of the surround channels, the A and B outputs at terminals **44** and **46** respectively have an amplitude ratio of –.38/.91, which results in a steering angle of 22.5 degrees to the rear.

The RS signal applied to terminal **64** similarly passes through attenuator **382** with unity gain to the inverting element **400**, and then through an attenuator **402** with a gain of 0.91, as for the LS channel. This signal is then added to $-crx$ times the unmodified RS signal in adder **408**. As for the LS channel, the signal passes through a 90 degree phase shifter element **246** and thence to an adder element **280**. The R, C and LFE signals after combination in summing circuit **282** pass through a phase shift element **288** into the adder **280**, where they mix with the phase-shifted RS and cross-fed LS signals to provide the "B" output signal at terminal **46**.

As usual, for each of the output signals at terminals **44** and **46**, the output level is unity, as the sum of the squares of 0.38 and 0.91 is one.

While the output of the encoder is 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 RS inputs with the same signal, a common practice in film, all the signals at the summing nodes are in phase, so the total level in the output is .38 + .91, or 1.29. This output is too strong by the factor of 1.29, or 2.2dB. Active circuitry (not shown, but similar to the active circuitry in the decoder) is

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included in the encoder to reduce the gain by the 2.2dB factor when this situation occurs, i.e. when the two surround channels are similar in amplitude and are in phase.

Another error occurs when the two surround inputs are equal in level and out of phase. In this case, the two attenuation factors subtract, so the output level is .91 – .38, or .53. This signal will decode as a center direction signal, at a reduced level. This error is severe. The previous encoder 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 (not shown, but similar to that in the decoder) increases the value of *fs* when the two rear channels are similar in level but 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, which represents an unsteered signal, this being the desired effect.

In discussions at the IRT in Munich, it was noted 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 3dB and added into the left front channel. This encoder has many disadvantages when encoding multichannel film sound, or recordings which have specific instruments assigned to the surround channels. Both the loudness and 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. It was decided that the new encoder of the present invention should also incorporate this 3dB attenuation when classical music was being encoded, and that one could detect this condition through monitoring the relative levels of the front and surround channels in the encoder.

A major function, therefore, 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

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surround channels are much softer than the front channels. Circuitry similar to that in the decoder 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. This maximum 3dB 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. However, instruments which are intended to be strong in the rear channels are encoded with full level.

There is another function of the real coefficient mixing path *fs* for the surround channels. Note that this path, through attenuators **378** and **380** also passes through cross-feed elements **384** and **386** to adders **392** and **394** in the opposite channels, with 0.91 attenuator elements **388** and **390** in the main signal paths, before being applied to the summing circuits **278** and **282**. When a sound is moving from the left front input toward the left rear input, active circuitry (not shown) compares the levels in these inputs and detects that these signals are similar in amplitude and in phase, and under these conditions, *fc* is reduced to zero while *fs* is increased to one. This change to real coefficients in the encoding results in a more precise encoding of this type of pan. In practice, this function is probably not essential, but it seems to be an elegant refinement.

In summary, then, the active circuits comprise elements for comparing the level and phase between the front and rear channels on each side, and for comparing the relative energy in the front and rear channels. These circuits are easily implemented in the form of log-ratio detectors and are well known to those skilled in the art. Dependent upon the outputs from these detectors, these active circuits

1. Reduce the level of the surround channels by 2.2dB when the signals 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;

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3. Decrease the level of the surround channels by up to 3dB when the surround level is much less (-8dB) than the front channels;

4. Increase the level and negative phase of the rear channels when their level is similar to the front channels; and

5. Make the surround channel mix use real coefficients when a sound source is panning from a front input to the corresponding rear input.

Additional improvements to the encoder are likely to include a feature for the front channels such that when the two front channels are out of phase the encoder will not cause the decoder to place the sound in the rear, as at present, but will detect this condition and make the encoded output appear to be unsteered (i.e. a quadrature phase shift between the A and B channels will result.)

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, without departing from the spirit of the invention.

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1. A surround sound decoder for redistributing a pair of left and right audio input signals including directionally encoded and non-directional components into a plurality of output channels 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:

delay means for delaying each of said left and right audio input signals to provide delayed left and right audio signals;

a plurality of multiplier means 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 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 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 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.

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2. The decoder of claim 1 wherein said left and right audio input signals were originally encoded from five channels onto two channels and wherein said plurality of output signals is five such that said left and right audio input signals are decoded into five output signals which are reproduced by amplification and application of the said five amplified output signals to five loudspeakers arranged to surround the listener.

3. The decoder of claim 2 wherein said loudspeakers are positioned at locations to the left front, center front, right front, left rear and right rear of the listening position, said output signals being typically named according to their intended directional locations relative to the listener's position.

4. The decoder of claim 3 wherein a sixth audio output signal of reduced bandwidth, intended for low frequency sound effects, is provided in addition to the five output signals enumerated.

5. The decoder of claims 1 thru 4 wherein said left and right audio input signals are decoded into said plurality of output signals by means of analog circuitry.

6. The decoder of claims 1 thru 4 wherein said left and right audio input signals are decoded into said plurality of output signals by means of digital signal processing circuitry, after first converting said input signals into digital form, and finally converting said output signals back into analog form suitable for reproduction on a like plurality of loudspeakers surrounding the listener.

7. The decoder of claim 3 wherein the said variable matrix coefficients determining the level of the rear output signal sis maintained at a 3dB lower level when the said left-right steering signal is of small magnitude and rises to the full magnitude when this signal reaches a magnitude equivalent to a steering angle of

22.5 degrees or more, but is independent of the said center-surround steering signal, in a mode of operation intended for television sound reproduction, thereby providing less variation of the sound level of the rear outputs relative to the front outputs when steering occurs in the forward direction resulting in more natural and smoother surround sound effects.

8. The decoder of claim 3 wherein the absolute value of one of the said steering signals is limited when the other of said steering signals is rapidly changing, thereby providing improved responsiveness to dynamic effects.

9. The decoder of claim 3 wherein in each front quadrant the coefficients calculated for the left and right components of the input signal are made such that the sum of the squares of these elements is equal to one when the center-surround steering signal is close to zero, so as to reduce unwanted variations of the total power delivered to the loudspeakers as a result of steering.

10. The decoder of claim 3 wherein a center boost function is provided such as to result in minimal apparent motion of central sound sources in the front while retaining maximum left-right separation of unsteered sounds.

11. The decoder of claim 3 wherein a new center front matrix coefficient is made dependent on the said center-surround steering signal in such manner that up to an effective angle of 30 degrees toward the front the center channel output rises to a value of 3dB lower than a standard type of decoder, then rises more rapidly to reach the same maximum level used in the standard type of decoder at full front steering, additionally cutting the level of the center component of the signal in the left and right front channels so as to preserve in the sum of the powers from each output of the decoder the ratio of the power of the center input signal to the total power of all other input signals into the encoder from which the said left and right audio input signals to said decoder were derived: thereby preserving in the outputs of the decoder the same balance

between center and other signals as was present before the signals were encoded, and also preserving the balance between center and other signals in recordings originally mized for two channel playback.

12. An active encoder suitable for encoding five channels of full bandwidth audio onto two output channels, said five input channels being respectively left front, center front, right front, left rear or surround, and right rear or surround, and said output channels comprising left or A and right or B channels respectively, comprising:

first, second, third, fourth and fifth input terminals for said left front, center front, right front, left surround, and right surround channels respectively;

an attenuator circuit connected to said second input terminal for attenuating said center front signal by a factor *fen*;

first and second summing circuits, said first summing circuit receiving the attenuated center front signal from said attenuator circuit and the left front input from said first input terminal directly, and said second summing circuit also receiving said attenuated center front signal and the right front signal from said third input terminal directly;

first functional attenuator with attenuation function *fc(l, ls)* for attenuating said left surround signal received from said fourth input terminal;

second functional attenuator with attenuation function *fc(r, rs)* for attenuating said right surround signal received from said fifth input terminal;

third functional attenuator with attenuation function *fs(l,ls)* for attenuating said left surround signal received from said fourth input terminal;

fourth functional attenuator with attenuation function *fs(r,rs)* for attenuating said right surround signal received from said fifth input terminal;

first, second, third and fourth cross-feed attenuators having an attenuation factor *-crx* each receiving the output signal from said first, second, third and fourth functional attenuators, respectively;

first, second, third and fourth fixed attenuators having an attenuation factor of 0.91 each receiving the output signal from said first, second, third and

fourth functional attenuators, respectively;

first, second, third and fourth adders each receiving the output signals from said first, second, third and fourth fixed attenuators, respectively, and adding to them the output signals from said second, first, fourth and third cross-feed attenuators, respectively;

first and second phase shifter circuits having a phase shift function *-90°* receiving the outputs of said first and second adders respectively;

third and fourth phase shifter circuits having a phase shift function *-0°* receiving the outputs of said first and second summing circuits respectively, said first and second summing circuits also receiving the outputs of said third and fourth adders respectively and combining them with the left front and right front input signals respectively together with the attenuated center front signal from said attenuator circuit connected to said second input terminal;

third summing circuit for summing the output signals from said first and third phase shifter circuits respectively to provide said left or A output signal to a first output terminal;

fourth summing circuit for summing the output signals from said second and fourth phase shifter circuits respectively to provide said right or B output signal to a second output terminal;

first, second, third, fourth and fifth log amplitude detector means for detecting the amplitudes of the signals applied to said first, second, third, fourth and fifth input terminals;

first and second comparing means for comparing the log amplitudes of the front and rear signals on the left side and on the right side respectively;

means responsive to the output of said first comparing means for reducing the function *fc(l,ls)* of said first functional attenuator from one to zero and increasing the function *fs(l,ls)* of said third functional attenuator from zero to one in a complementary manner as the predominant signal direction changes from left front towards left surround;

means responsive to the output of said second comparing means for reducing the function *fc(r,rs)* of said second functional attenuator from one to zero

and increasing the function *fs(r,rs)* of said fourth functional attenuator from zero to one in a complementary manner as the predominant signal direction changes from right front towards right surround;

third comparing means for comparing the log amplitudes of the front signals with those of the rear signals;

means responsive to the output signal from said third comparing means for reducing the function *fcn* by up to 3dB as the front signals exceed the rear signals by up to 8dB or more;

fourth comparing means for comparing the phase and amplitude between the left surround and right surround input signals applied to said fourth and fifth input terminals; and

means responsive to the output signal from said fourth comparing means for reducing the gain of said first and second functional attenuators by up to 2.2dB when the two surround signals are similar in amplitude and in phase, and for increasing the gains of said third and fourth functional attenuators when the two surround signals are similar in amplitude but in antiphase, thereby forcing the A and B outputs to be in quadrature phase relationship representing an unsteered condition.

13. The encoder of claim 12 further comprising:

a sixth input terminal for receiving a low frequency effects signal;

a gain stage of gain 2.0 for amplifying said low frequency effects signal;

said amplified low frequency effects signal being applied equally to additional inputs of said first and second summing circuits so as to appear equally and in phase in both A and B outputs at said first and second output terminals respectively.

This Sheet Has Been Replaced

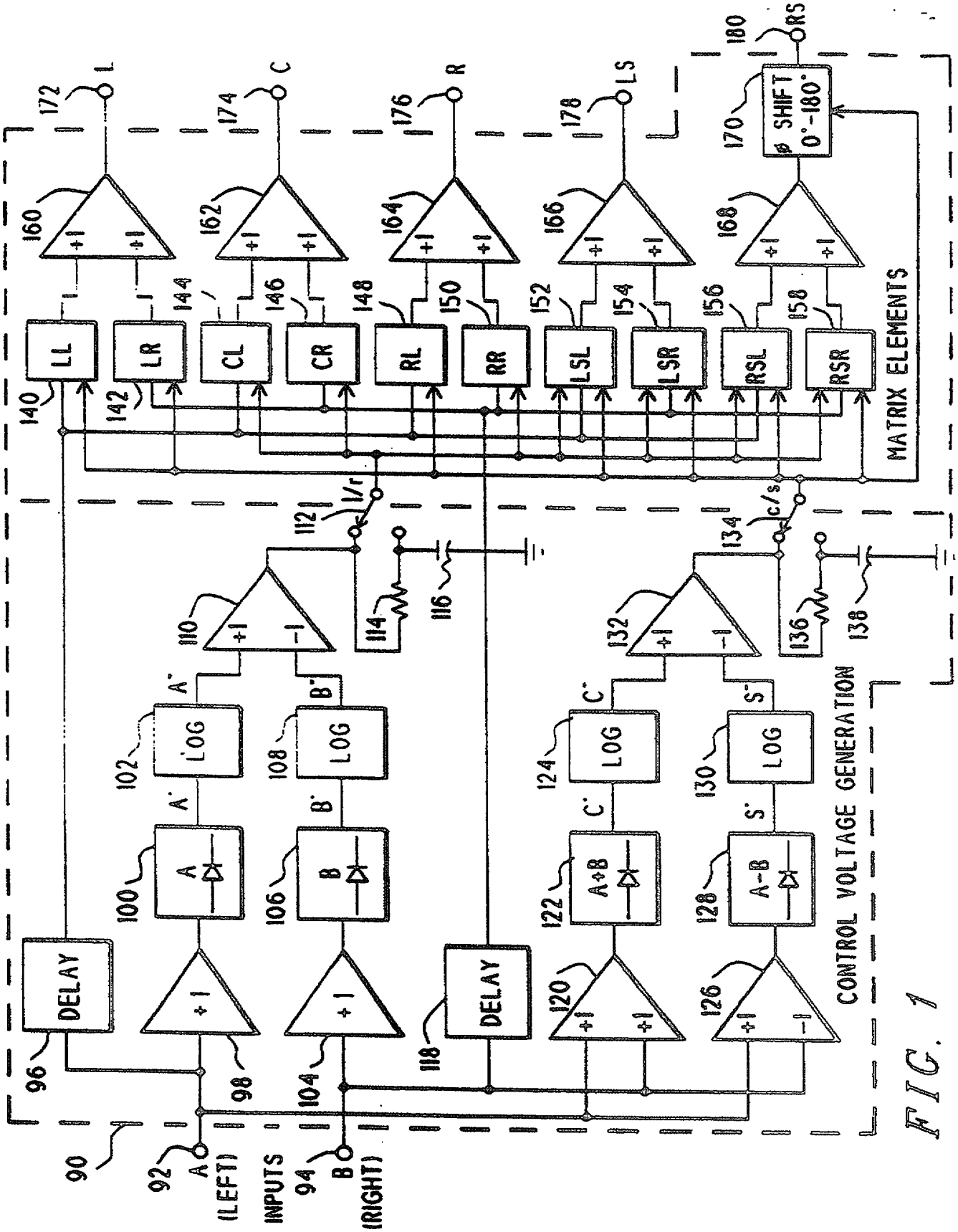
L
C
R
LS
RS
SHIFT 0°-180°
LL
LR
CL
CR
RL
RR
LSL
LSR
RSL
RSR
MATRIX ELEMENTS
l/r
c/s
LOG
A
B
A+B
A-B
DELAY
CONTROL VOLTAGE GENERATION
A (LEFT) INPUTS
B (RIGHT)

FIG. 1

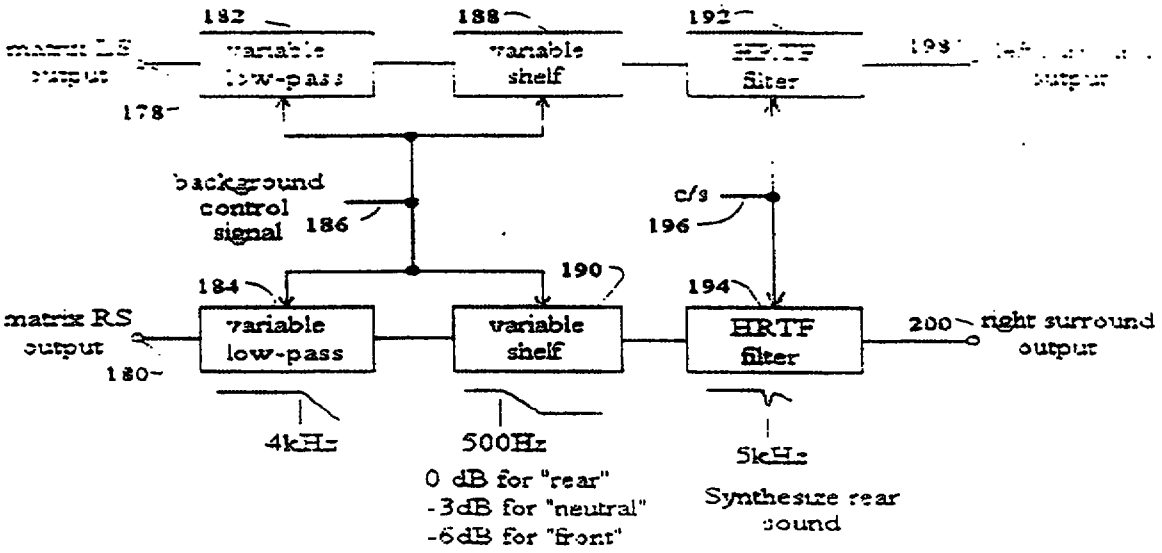
182
188
192
193
variable low-pass
variable shelf
HRTF filter
178
background control signal 186
c/s 196
190
184
194
matrix RS output 180
variable low-pass
variable shelf
HRTF filter
200 right surround output
4kHz
500Hz
5kHz
0 dB for "rear"
-3dB for "neutral"
-6dB for "front"
Synthesize rear sound

FIG. 2

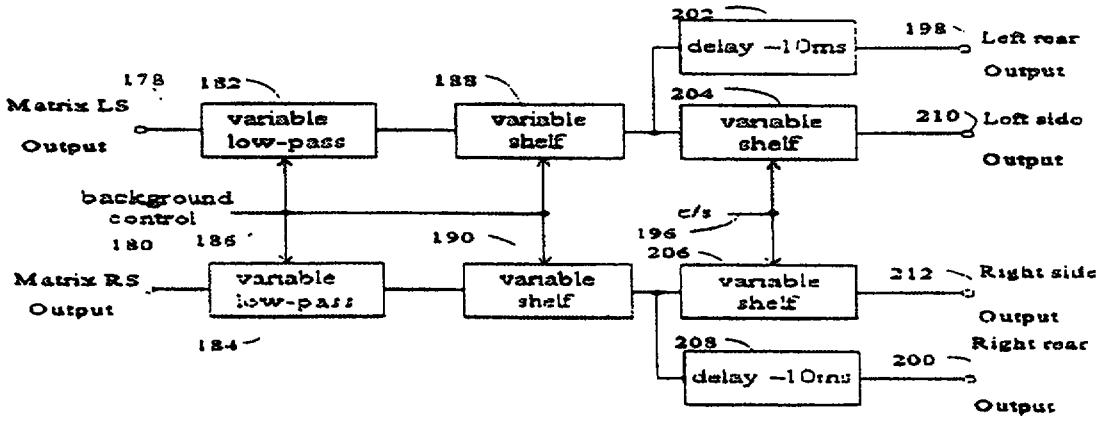
202
delay ~10ms
198 Left rear Output
178
182
188
204
Matrix LS Output
variable low-pass
variable shelf
variable shelf
210 Left side Output
background control
180
186
190
c/s
196
206
Matrix RS Output
variable low-pass
variable shelf
variable shelf
212 Right side Output
184
208
delay ~10ms
200 Right rear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

FIG. 3

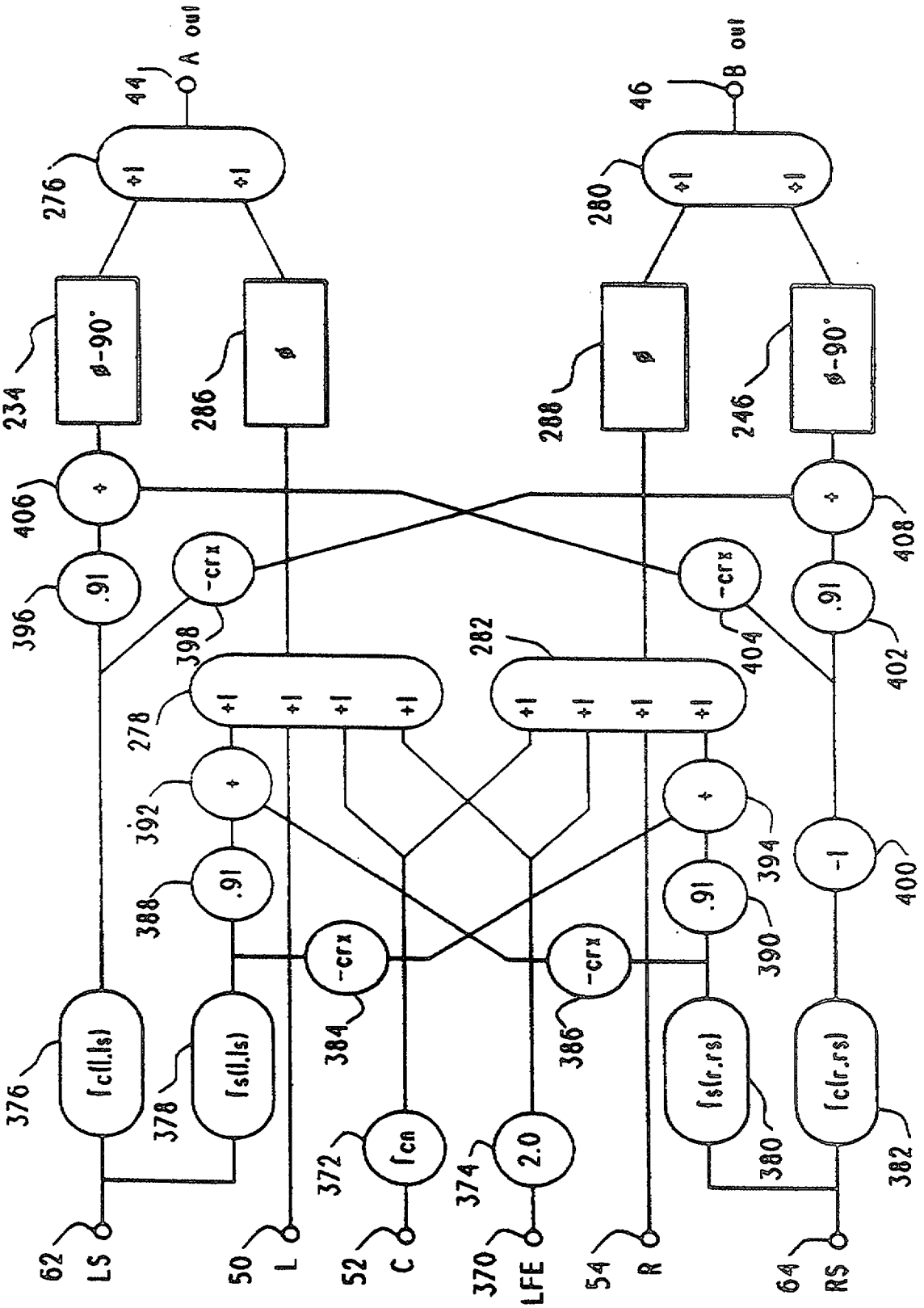
A out
B out
44
46
276
280
+1
234
ø-90°
286
ø
288
246
406
408
396
.91
-crx
398
404
402
282
278
392
388
394
390
400
376
fc(l,ls)
378
fs(l,ls)
384
386
fs(r,rs)
380
fc(r,rs)
382
372
fcn
374
2.0
62
LS
50
L
52
C
370
LFE
54
R
64
RS

FIG. 4

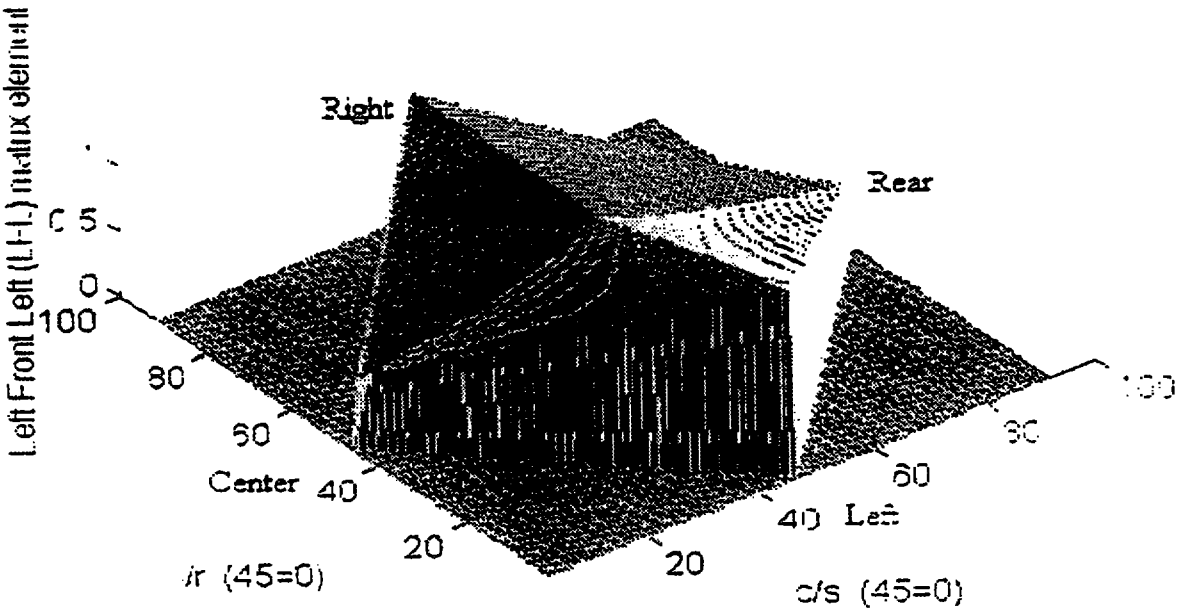
Left Front Left (LFL) matrix element
0.5
0
100
80
60
Center
40
20
l/r (45=0)
Right
Rear
20
40 Left
60
80
100
c/s (45=0)

FIG. 5

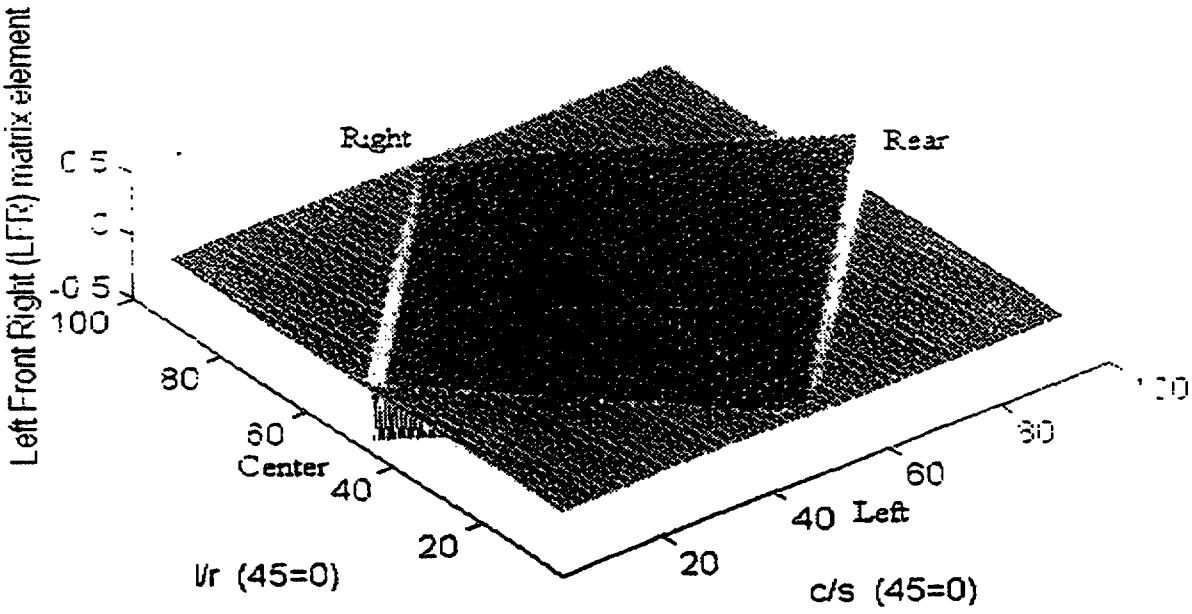
Left Front Right (LFR) matrix element
0.5
-0.5
100
80
60
Center
40
20
l/r (45=0)
Right
Rear
20
40 Left
60
80
c/s (45=0)

FIG. 6

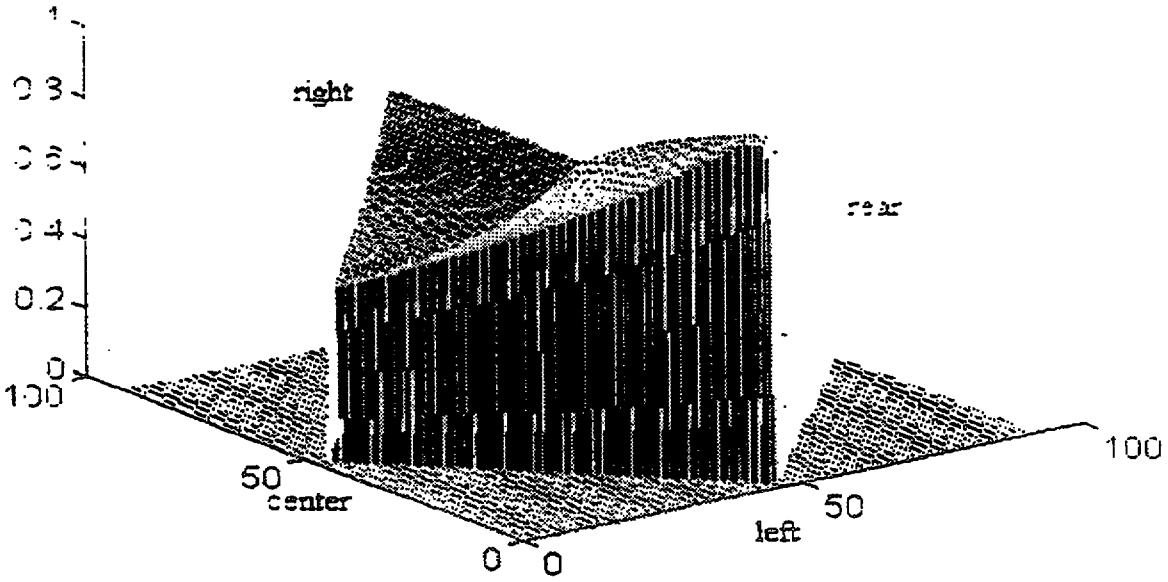
right
rear
0.2
0
100
50
center
0
0
50
left
100

FIG. 7

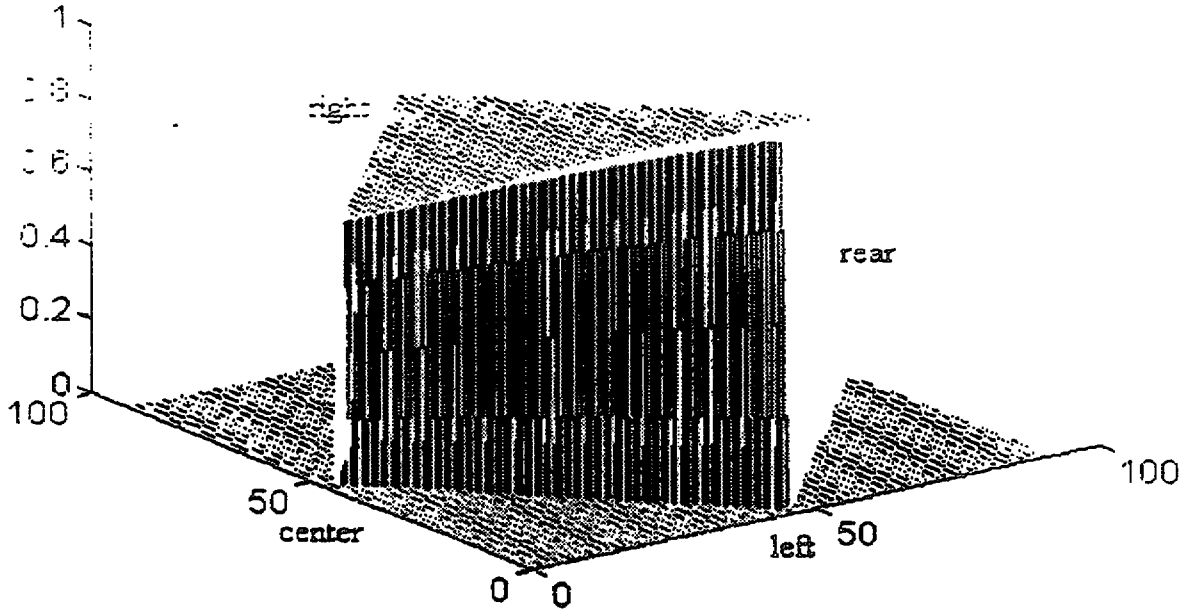
rear
0.4
0.2
0
100
50
center
0
0
50
left
100

FIG. 8

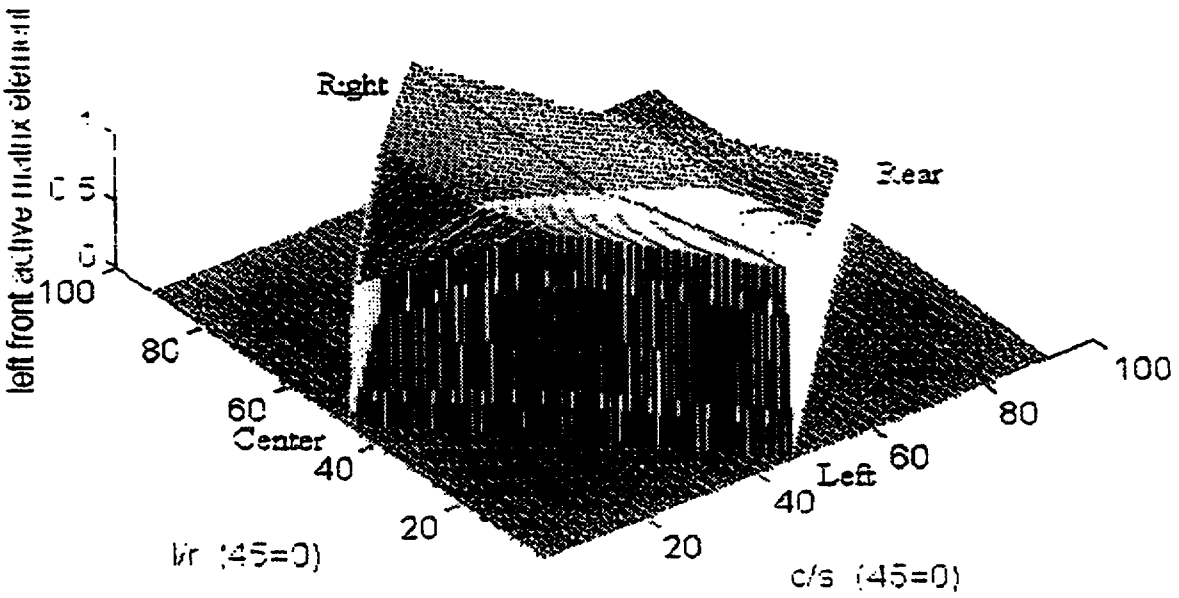
Right
Rear
100
80
60
Center
40
20
Left
lr (45=0)
c/s (45=0)

FIG. 9

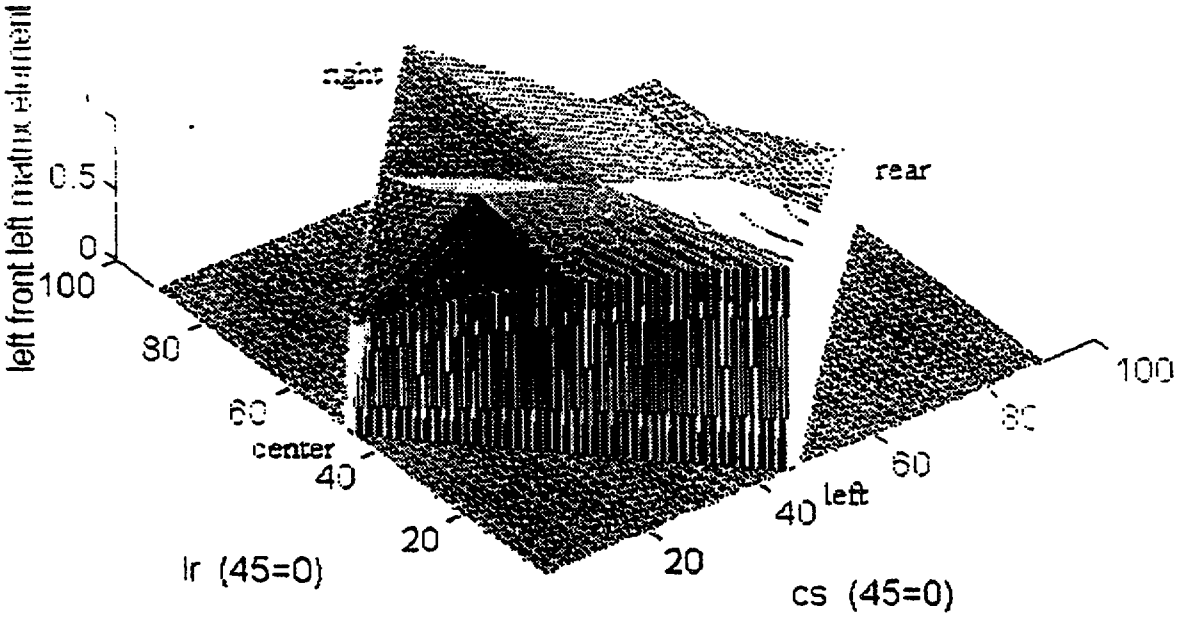
right
rear
0.5
0
100
80
60
center
40
20
left
lr (45=0)
cs (45=0)

FIG. 10

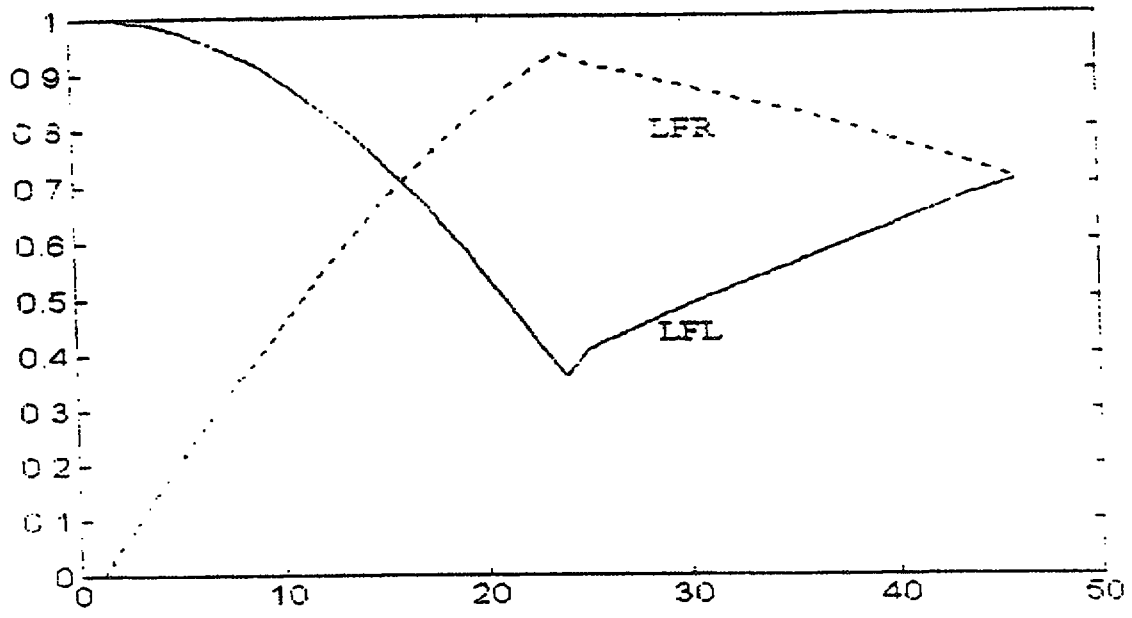
1
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0
10
20
30
40
50
LFR
LFL

FIG. 11

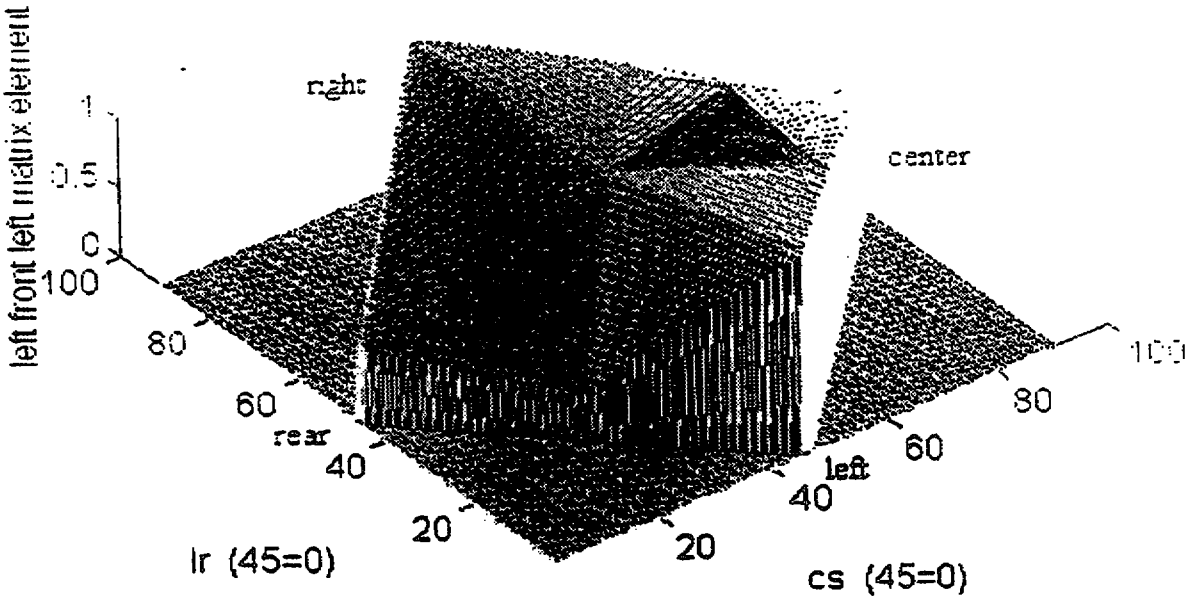
left front left matrix element
1
0.5
0
100
80
60
rear
40
20
lr (45=0)
right
center
left
0
20
40
60
80
100
cs (45=0)

FIG. 12

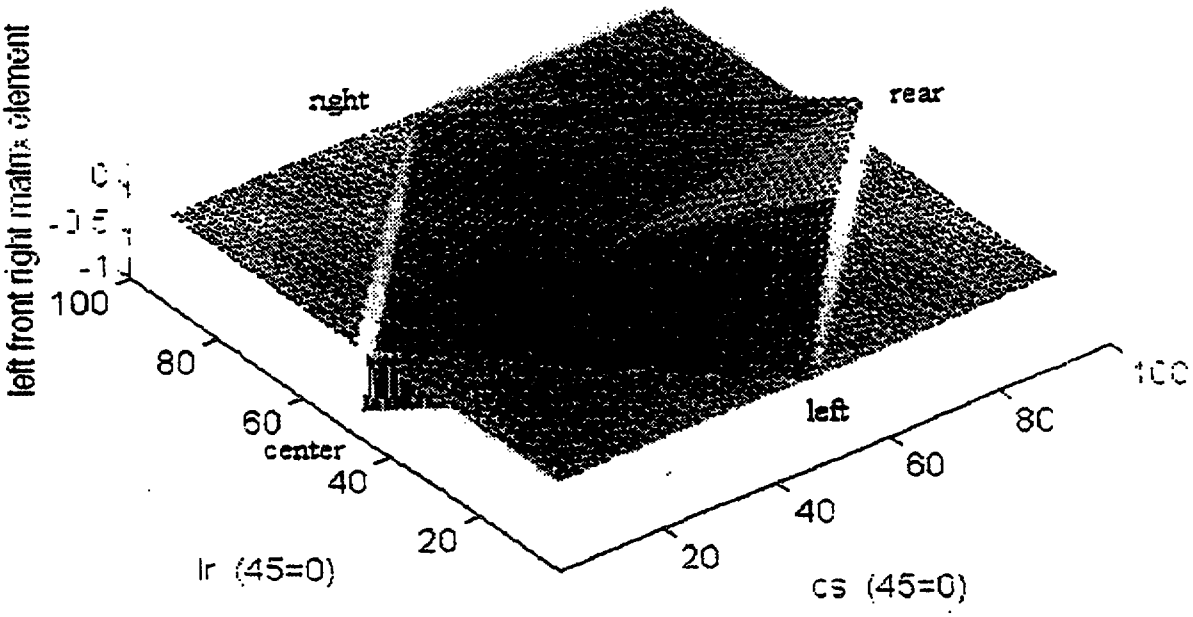
left front right matrix element
right
rear
left
center
lr (45=0)
cs (45=0)
100
80
60
40
20
0
-1

FIG. 13

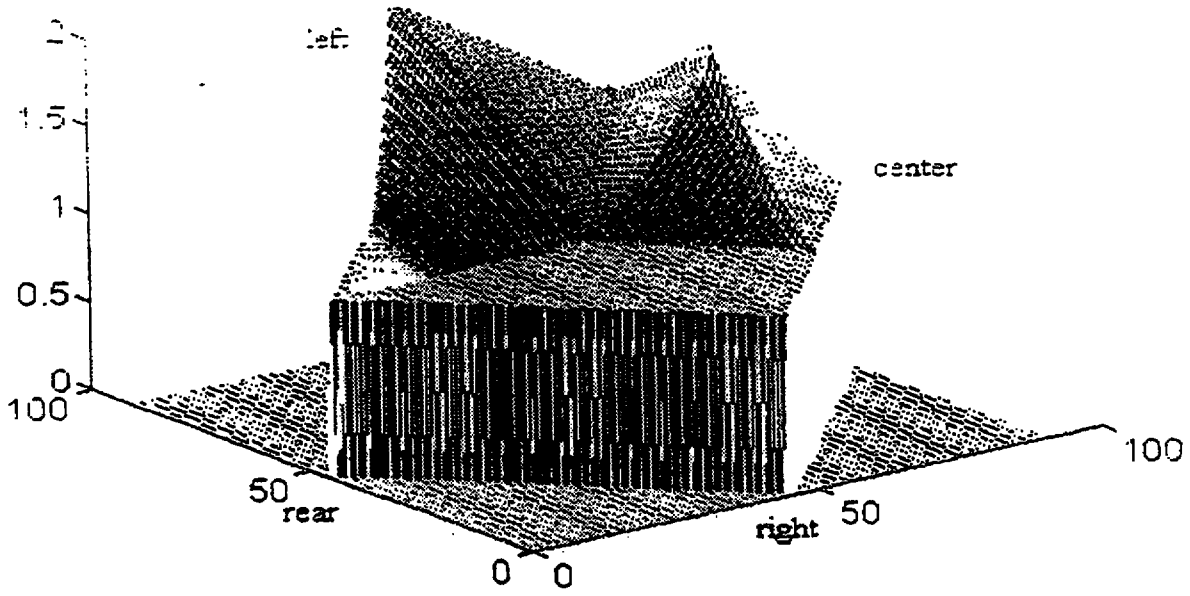
left
center
rear
right
100
50
0
1.5
1
0.5

FIG. 14

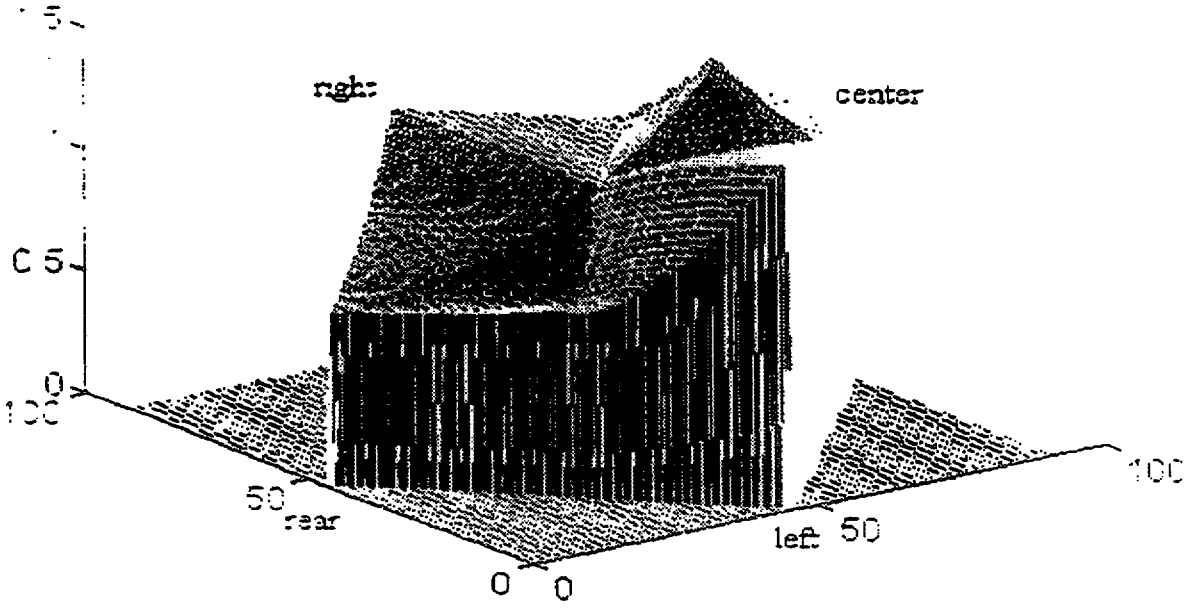
right
center
50
rear
0
0
left 50
100

FIG. 15

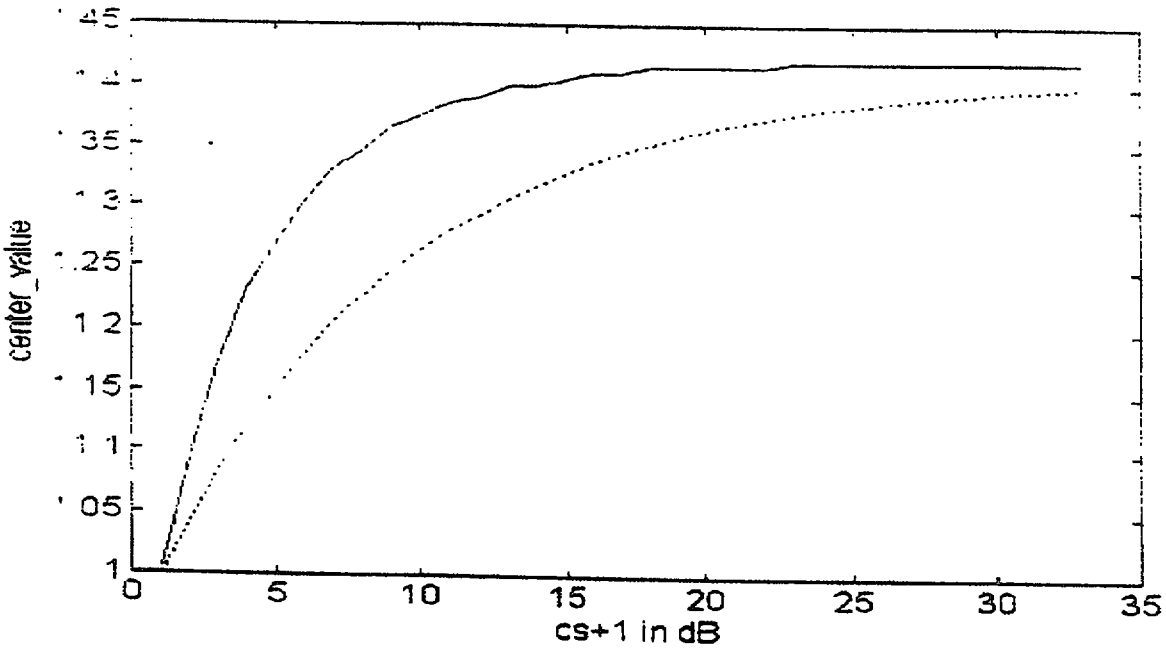
center_value
1
1.05
1.1
1.15
1.2
1.25
1.3
1.35
1.4
1.45
0
5
10
15
20
25
30
35
cs+1 in dB

FIG. 16

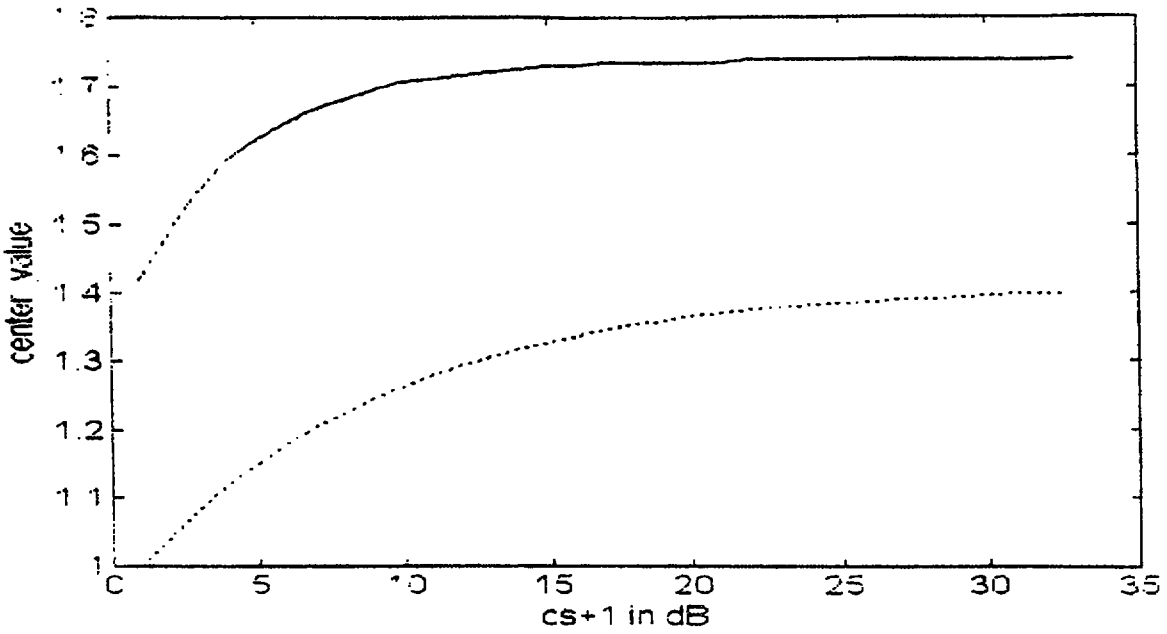
center value
1
1.1
1.2
1.3
1.4
1.5
1.6
1.7
1.8
0
5
10
15
20
25
30
35
cs+1 in dB

FIG. 17

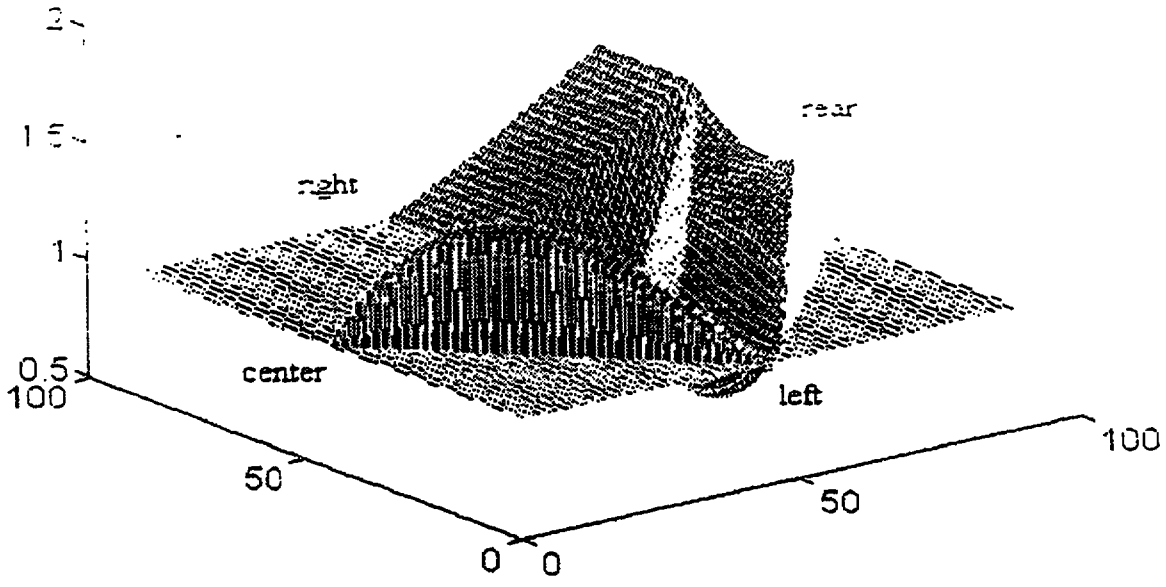
2
1.5
1
0.5
rear
right
center
left
100
50
0
0
50
100

FIG. 18

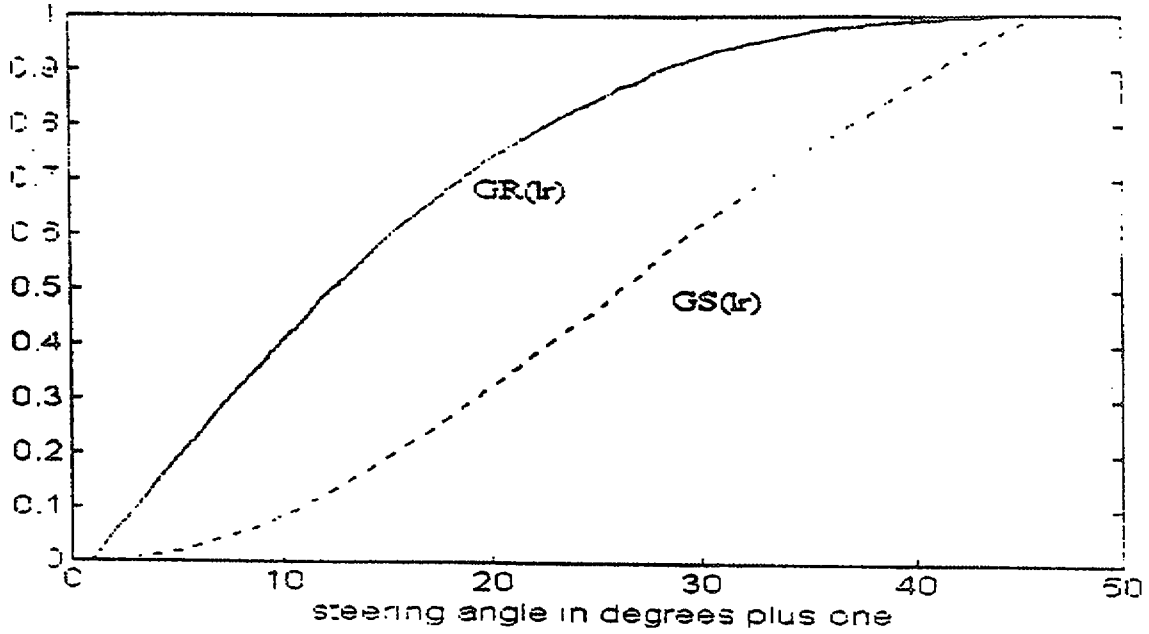
GR(lr)
GS(lr)
0.5
0.4
0.3
0.2
0.1
0
10
20
30
40
50
steering angle in degrees plus one

FIG. 19

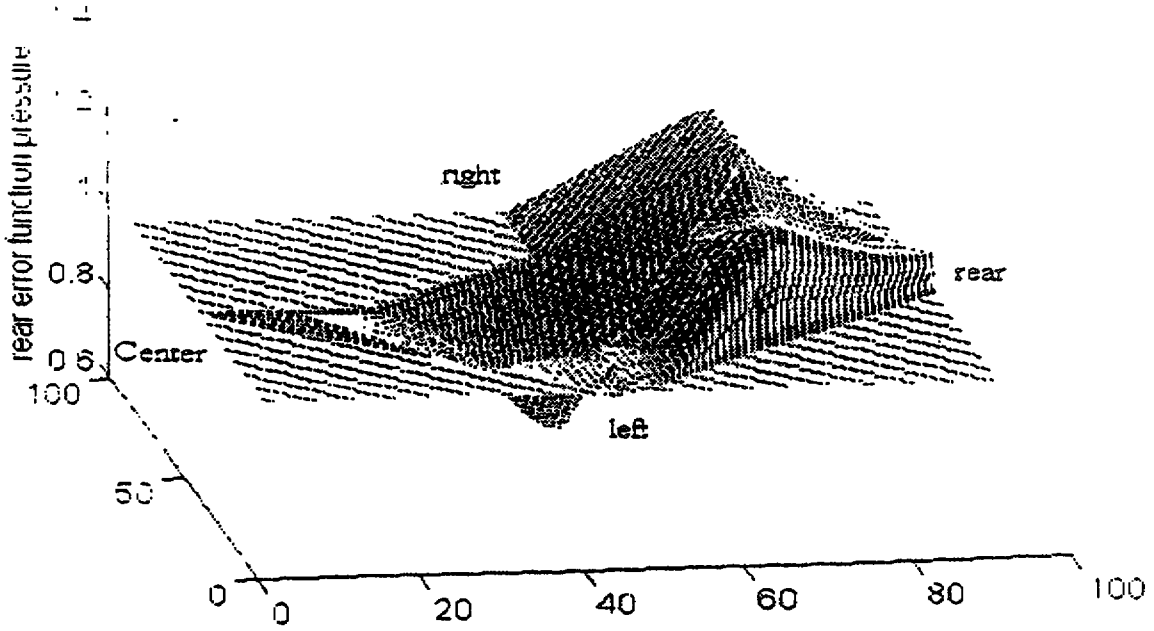
right
rear
left
Center
0.8
0.6
100
50
0
0
20
40
60
80
100

FIG. 20

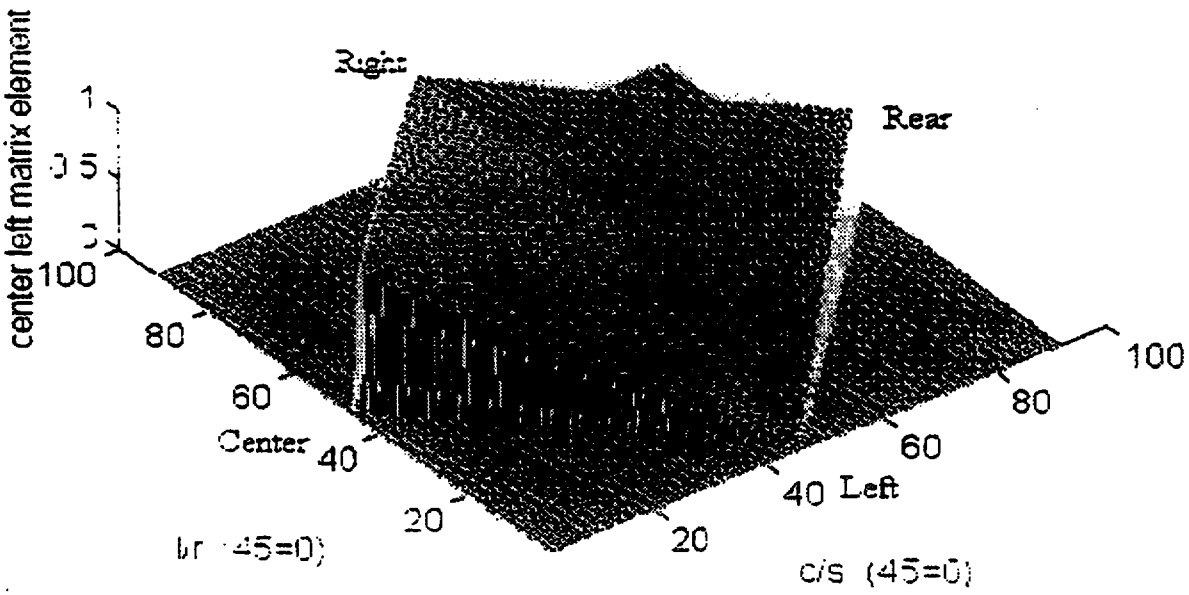
center left matrix element
1
0.5
0
100
80
60
Center
40
20
Right
Rear
100
80
60
40 Left
20
l/r (45=0)
c/s (45=0)

FIG. 21

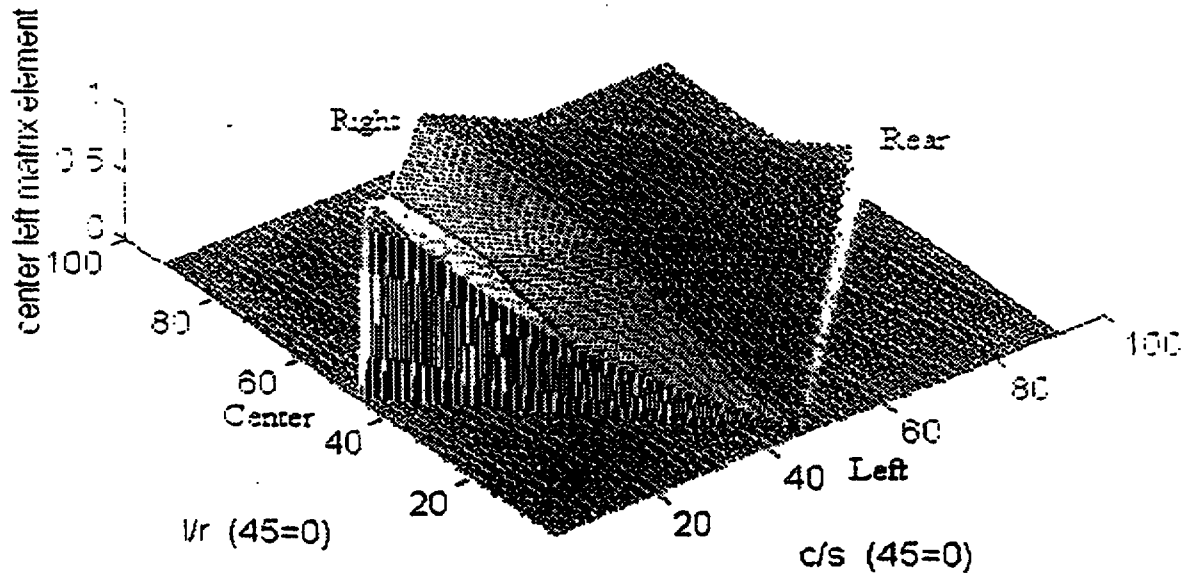
center left matrix element
100
80
60
Center
40
20
Right
Rear
100
80
60
40 Left
20
l/r (45=0)
c/s (45=0)

FIG. 22

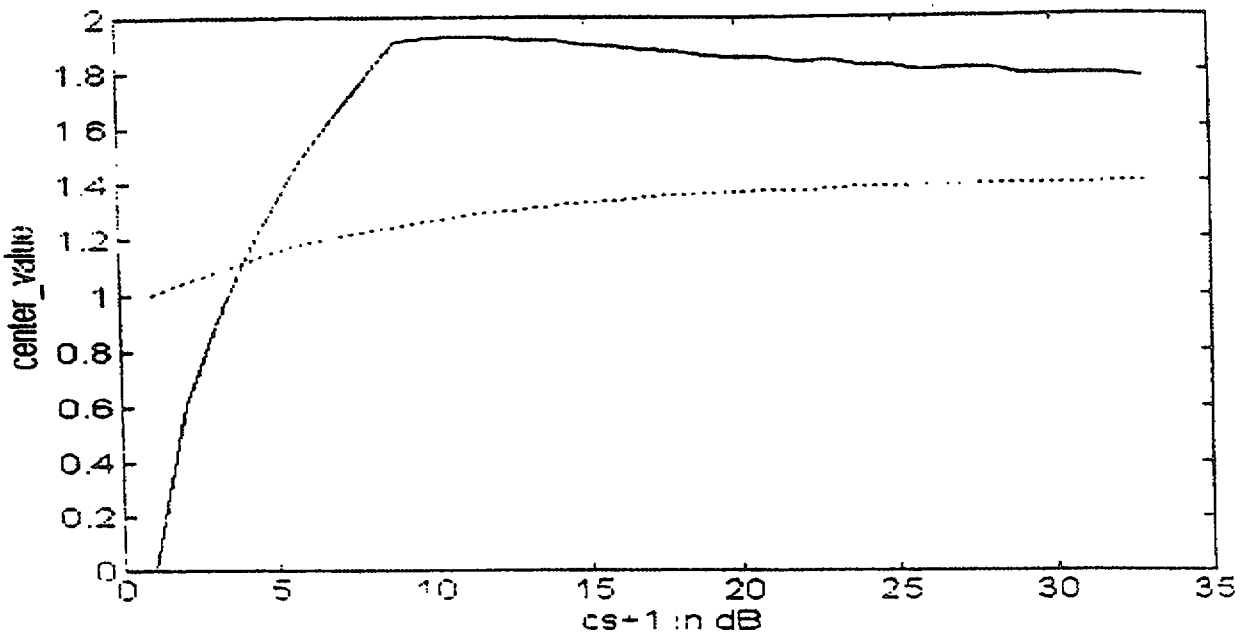
center_value
2
1.8
1.6
1.4
1.2
1
0.8
0.6
0.4
0.2
0
0
5
10
15
20
25
30
35
cs+1 in dB

FIG. 23

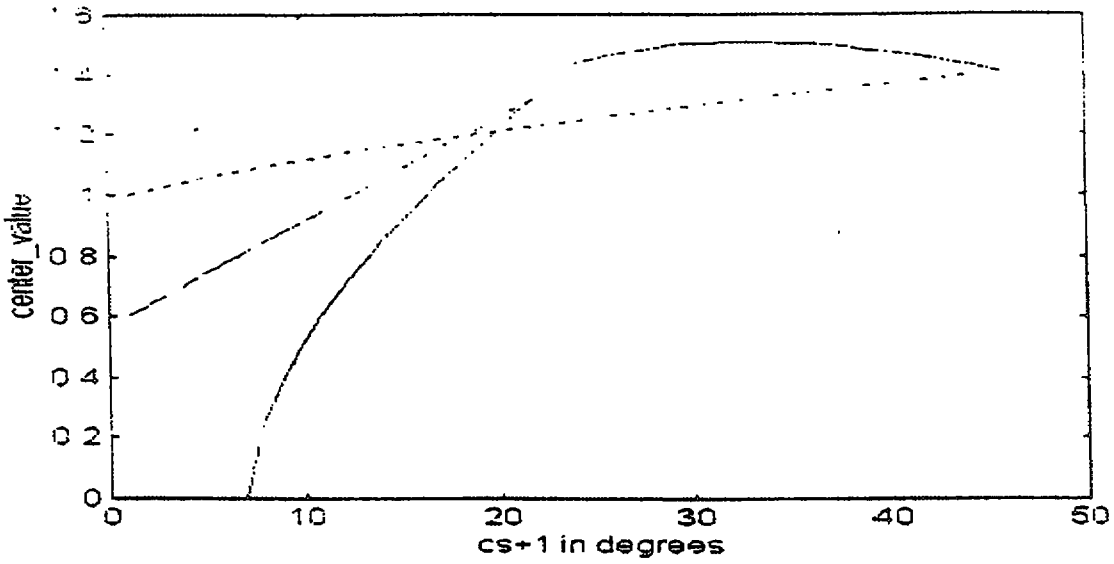
center_value
1.8
0.6
0.4
0.2
0
0
10
20
30
40
50
cs+1 in degrees

FIG. 24

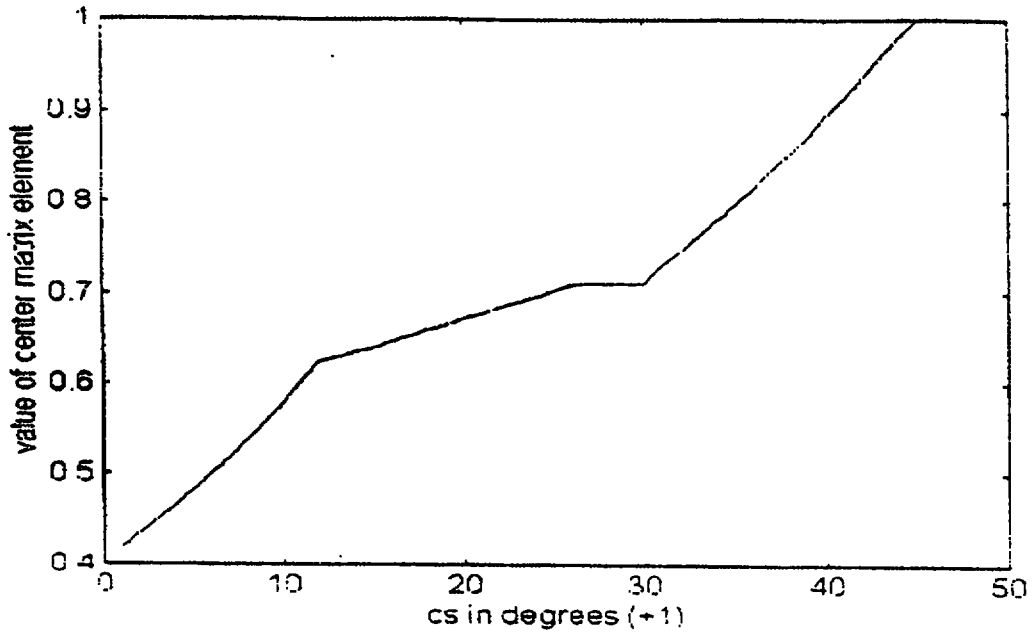
value of center matrix element
1
0.9
0.8
0.7
0.6
0.5
0.4
0
10
20
30
40
50
cs in degrees (+1)

FIG. 25

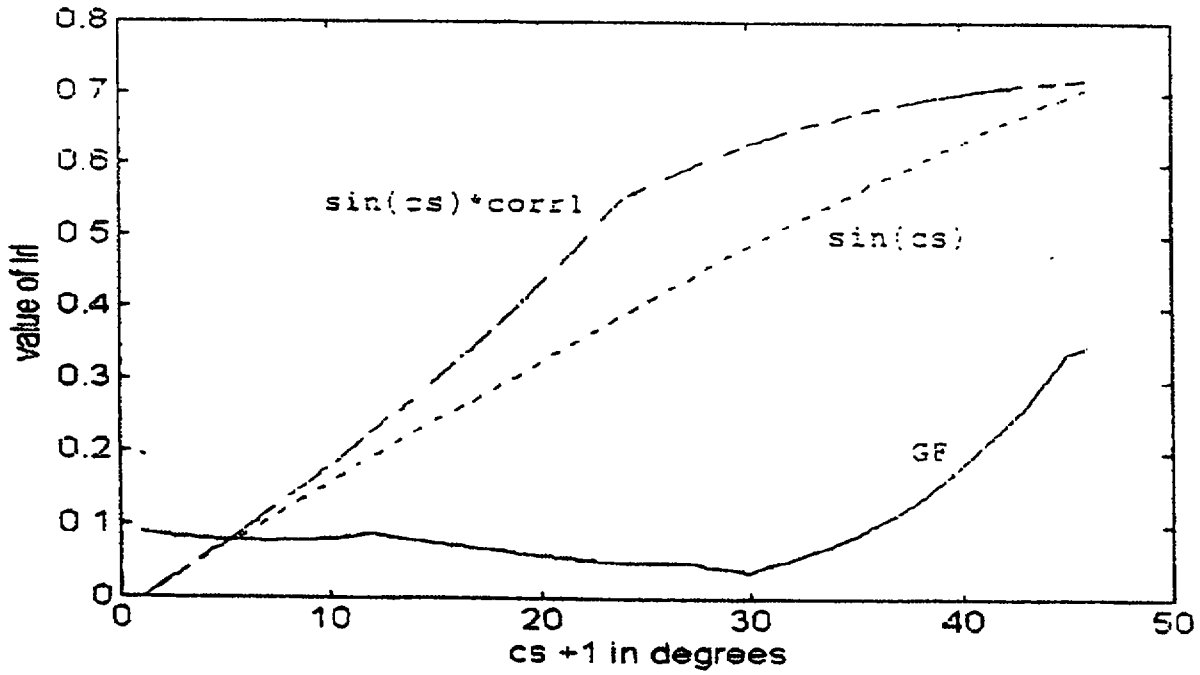
value of Irl
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
sin(cs)*corr1
sin(cs)
GF
0
10
20
30
40
50
cs +1 in degrees

FIG. 26

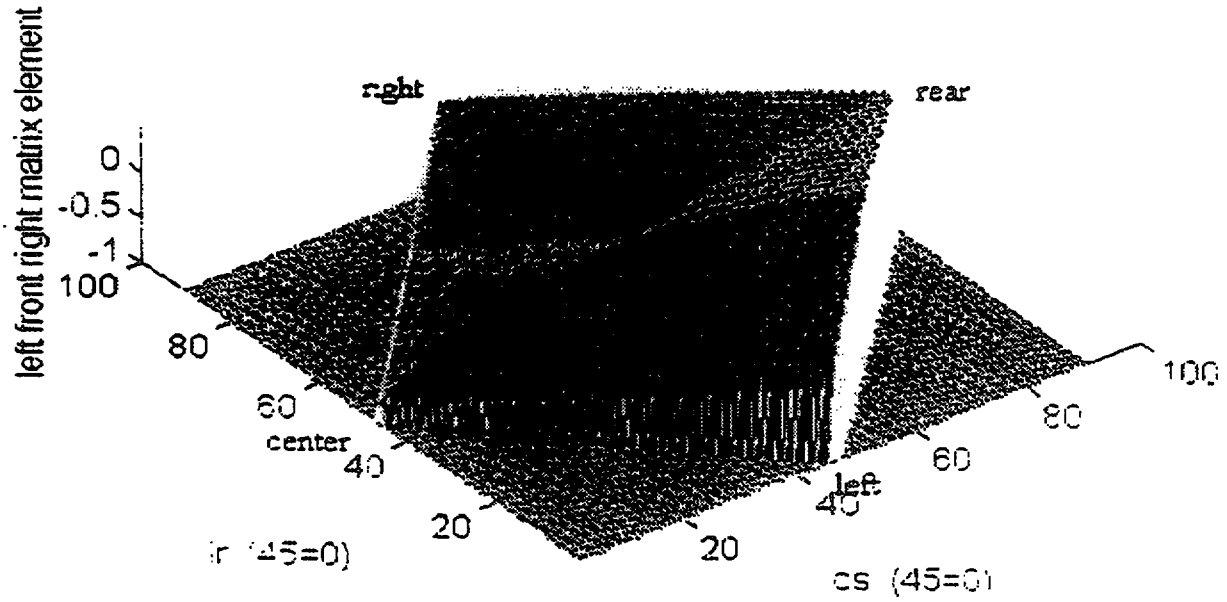
left front right matrix element
0
-0.5
-1
100
80
60
center
40
20
right
rear
100
80
60
left
40
20
cs (45=0)

FIG. 27

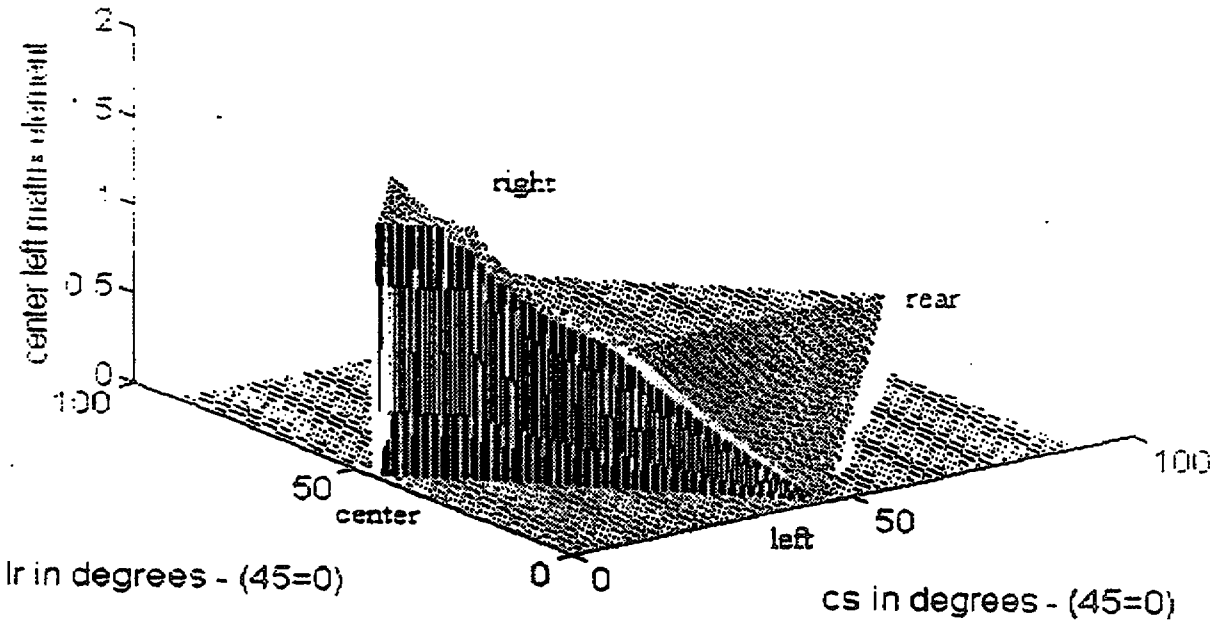
right
rear
0.5
0
100
50
center
0
0
left
50
100
lr in degrees - (45=0)
cs in degrees - (45=0)

FIG. 28

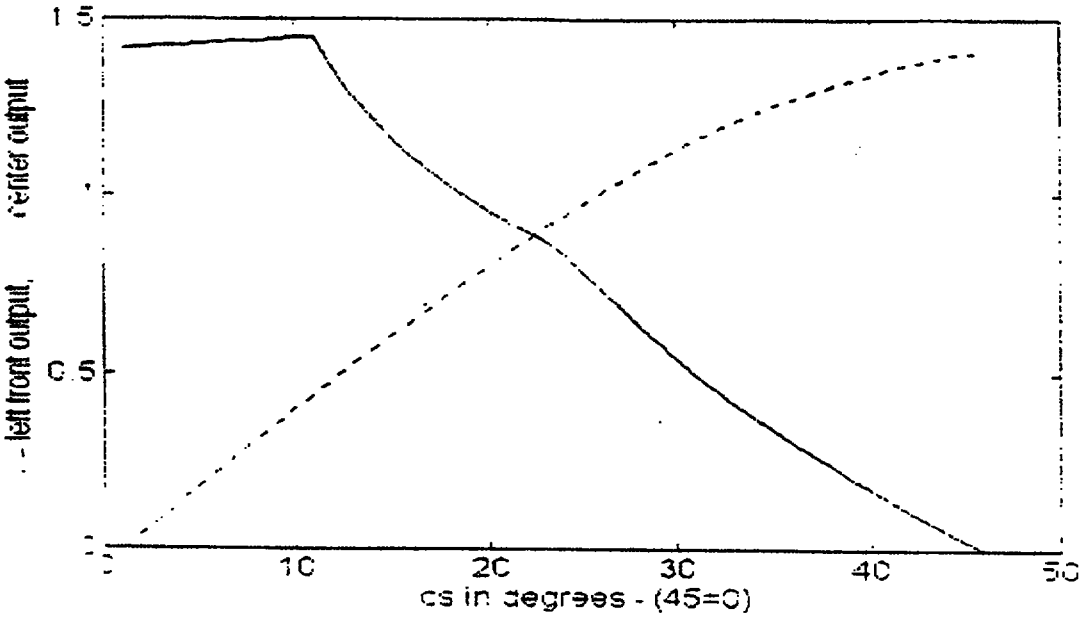
1.5
0.5
10
20
30
40
50
- left front output, center output
cs in degrees - (45=0)

FIG. 29

2000000923-3
Ref. Only
PA(1)

5-2-5 MATRIX ENCODER AND DECODER SYSTEM

Cross-Reference to Related Applications

This application is based upon the U.S. Provisional Patent Application No. 60/058,169 entitled "5-2-5 Matrix Encoder and Decoder System," filed September 5, 1997.

Field of the Invention

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, as well as the encoding of multichannel material into two channels.

Background of the Invention

The present invention concerns an improved set of design criteria and their solution to create a decoding matrix having optimum psychoacoustic

REPLACEMENT SHEET
02 MAY 2000

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.

Additionally, this invention concerns an improved set of design criteria and their solution to create an encoding circuit for the encoding of multi-channel sound onto two channels for reproduction in standard two channel receivers and by decoders according to the invention.

The present invention is part of a continuing effort to refine means of encoding multichannel audio signals into two separate channels, and then means for separating the resulting two channels back into the multichannel signals from which they were derived. One of the goals of this encode/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.

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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.

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

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.

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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. In a further aspect, it provides active circuitry that encodes five input channels into a two channel output that will perform optimally both with standard two channel equipment, with decoders according to the invention, and with industry standard "Dolby Pro-Logic" decoders.

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.

Additionally, decoders in accordance with the invention include

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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.

Encoders in accordance with the invention mix five (or five full-range plus one low frequency) input channels into two output channels such that when the input level of a particular input is strong the energy present at that input is preserved in the output; such that the direction of a strong input is encoded in the phase/amplitude ratio of the output signals; such that strong signals can be panned between any two inputs of the encoder and the output will be correctly directionally encoded. In addition, decorrelated material applied to the two rear inputs of the encoder will be encoded to two channels in such a fashion that the left/right separation of the inputs will be preserved when the encoder output is decoded by decoders according to the invention such that in-phase inputs to the two rear channels of the encoder will produce a two channel output that will decode to the rear channels of decoders according the invention, and decoders according the Dolby standard, and such that anti-phase inputs to the two rear channels of the encoder will produce outputs that will correspond to a non-directional signal for decoders according to the invention and to decoders according to the Dolby standard, and also such that low level reverberant signals applied to the two rear inputs of the encoder will be encoded with a 3dB reduction in level in the two channel output.

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Brief Description of the Drawing

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 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,

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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

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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;

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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

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.

Experience with the decoder and encoder 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

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disclosed. This application will present the most essential features of the improved encoder and 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

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.

In mathematical equations describing the decoder and encoder 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

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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.

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 which is incorporated herein by reference.

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. Patent Application No. 08/742,460 presented mathematical formulae for these

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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

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

$$lr = 90 - \arctan(10\,\hat{}\,((l/r)/20)) \quad \ldots(1a)$$

$$cs = 90 - \arctan(10\,\hat{}\,((c/s)/20)) \quad \ldots(1b)$$

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.

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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.

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

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.)

We have found in practice that there are several advantages to applying

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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.

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.**

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.

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

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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.

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.

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4. General description of the encoder

FIG. 4 shows a block diagram of an encoder 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.

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.

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

5. Design goals for the decoder active matrix elements

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

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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."

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.

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

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V1.11 are retained in the new design, but will not be covered in this document.

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 **278** 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**.

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.

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

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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**.

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.

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

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

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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.

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.

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.

Experience with the V1.11 decoder showed that although the reduction in center channel loudness solved the spatial problem, the power balance in the

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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.

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

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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.

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.

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.

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

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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*.

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.

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

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(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

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

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;
```


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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**.

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.

9. Matrix decoders in equations and graphics

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

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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.

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

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,

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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.

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.

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

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

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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.

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.

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.

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

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standard (Dolby) decoder the surround channel is treated differently, and this issue will be discussed at length later in this application.

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.

10. The left front matrix elements in our'89 patent

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

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

In the left front quadrant:

$$LFL = 1 - 0.-5*G(cs) + 0.41*G(lr) \quad ...(3a)$$

$$LFR = -0.5*G(cs) \quad ...(3b)$$

In the right front quadrant:

$$LFL = 1 - 0.5*G(cs) \quad ...(3c)$$

$$LFR = -0.5*G(cs) \quad ...(3d)$$

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

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

$$LFR = -0.5*G(cs) \quad ...(3f)$$

In the right rear quadrant:

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$$LFL = 1 - 0.5*G(cs) \quad \ldots(3g)$$

$$LFR = -0.5*G(cs) \quad \ldots(3h)$$

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.

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.

In the application No. 08/742,460 and Reference [1], we corrected the

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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.

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:

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

$$LFR = -\sin(cs) \quad \text{...(4b)}$$

For the right front quadrant:

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

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

For the left rear quadrant:

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

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

For the right rear quadrant:

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

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

11. Improvements to the left front matrix elements

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

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separation between the front channels and the rear channels when stereo music was panned to the rear.

For the front left quadrant:

$$LFL = (\cos(cs) - 0.41*G(lr))*boost1(cs) \quad ...(5a)$$

$$LFR = (-\sin(cs))*boost\ 1(cs) \quad ...(5b)$$

For the right front quadrant:

$$LFL = (\cos(cs))*boostl(cs) \quad ...(5c)$$

$$LFR = (-\sin(cs))*boost\ 1\ (cs) \quad ...(5d)$$

For the left rear quadrant:

$$LFL = (\cos(-cs) + 0.41*G(lr))/boost(cs) \quad ...(5e)$$

$$LFR = (\sin(cs))/boost(cs) \quad ...(5f)$$

For the right rear quadrant:

$$LFL = (\cos(cs))1boost(cs) \quad ...(5g)$$

$$LFR = (\sin(cs))/boost(cs) \quad ...(5h)$$

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:

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

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-l)/(23*20)); % go up 3dB over this range
```


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```
  corr1(x) = corr(x),
end
for x = 25:46 % go back down for corrl over this range 24 to 45 degrees
  corr(x) = 1.41;
  corr(x) = corr(48-x);
end
```

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.

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.

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.

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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
```

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

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**

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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_tbl = sin(**a1**);

cos_tbl = cos(**a1**);

cos_tbl_plus = cos(**a1**).*corr1(**a**+1);

cos_tbl_plus = **cos_tbl_plus** – **cos_tbl**; % this is the one we use

cos_tbl_minus = cos(**a1**)./corr(**a**+1);

sin_tbl_plus = sin(**a1**).*corr1(**a**+1);

sin_tbl_plus = **sin_tbl_plus** – **sin_tbl**; % this is the one we use

sin_tbl_minus = sin(**a1**)./corr(**a**+1);

The vectors **sin_tbl_plus** and **cos_tbl_plus** are the difference between a plain sine and cosine, and the boosted sine and cosine. We now define

LFL = cos(*cs*) + 0.41*G(*lr*) + **cos_tbl_plus**(*bcs*) ...(7a)

LFR = –sin(*cs*) – **sin_tbl_plus**(*bcs*) ...(7b)

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.

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.

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

LFL = **cos_tbl_minus**(–*cs*) + 0.41*G(–*cs*) ...(8a)

LFR = **sin_tbl_minus**(–*cs*) ...(8b)

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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.

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:

$$LFL = \cos(t) * F(t) -/+ \sin(t) * (\mathrm{sqrt}(1 - F(t)\ \hat{}\ 2)) \quad ...(9a)$$

$$LFR = (\sin(t) * F(t) +/- \cos(t) * (\mathrm{sqrt}(1 - F(t)\ \hat{}\ 2))) \quad ...(9b)$$

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

$$LFL = \cos(t)*\cos(4*t) + \sin(t)*\sin(4*t) \quad ...(9c)$$

$$LFR = (\sin(t)*\cos(4*t) - \cos(t)*\sin(4*t) \quad ...(9d)$$

See FIG 11 for the plot of these coefficients LFL (solid curve) and LFR

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(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.)

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,

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

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

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

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

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

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;
```


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```
  end
```

and a new correction function which depends on *bp*:

```
for x = 1:24
  ax = 2*pi*(46-x) 360;
  front_boundary_tbl(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
```

We then define LFL and LFR in this quadrant as:

$$LFL = \cos(cs)/(\cos(cs)+\sin(cs)) - \text{front_boundary_tbl}(bp) + 0.41*G(lr) \quad \ldots(11a)$$

$$LFR = \sin(cs)/(\cos(cs)+\sin(cs)) + \text{front_boundary_tbl}(bp) \quad \ldots(11b)$$

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:

$$LFL = \cos(cs)/(\cos(cs)+\sin(cs)) = 1 - 0.5*G(cs) \quad \ldots(12a)$$

$$LFR = \sin(cs)/(\cos(cs)+\sin(cs)) = 0.5*G(cs) \quad \ldots(12b)$$

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

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

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steering progresses to full rear. Along the $lr = 0$ axis and in the right rear quadrant the function is identical to the Dolby matrix.

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.

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.

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.

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.)

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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.

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.

12. Rear matrix elements during front steering

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

$$LRL = 0.71*(1 - G(lr)) \quad \text{...(13a)}$$

$$LRR = 0.71*(-1) \quad \text{...(13b)}$$

For the rear left quadrant

$$LRL = 0.71*(1 - G(lr) + .41*G(-cs)) \quad \text{...(13c)}$$

$$LRR = -0.71*(1 + 0.41*G(-cs)) \quad \text{...(13d)}$$

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

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

For the front left quadrant

$$LRL = 1 - G(lr) \quad \text{...(14a)}$$

$$LRR = -1 \quad \text{...(14b)}$$

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For the rear left

$$LRL = 1 - G(lr) \quad \text{...(14c)}$$

$$LRR = -1 \quad \text{...(14d)}$$

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

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

13. A brief digression on the surround level in Dolby Pro-Logic

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.

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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.

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.

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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.

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

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

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.

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.

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.

In FIG 17, assuming equal power ratios to stereo given the matrix

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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*.

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

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

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

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

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

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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.

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.)

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.

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.

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

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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:

$$LRL = \cos(cs) - GS(lr) \quad ...(17a)$$

$$LRR = -\sin(cs) - GR(lr) \quad ...(17b)$$

We want the sum of the squares to be one along the $cs = 0$ axis,,

$$(1 - GS(lr))^2 + (GR(lr))^2 = 1 \quad ...(18)$$

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

$$LRL*\cos(t) + LRR*\sin(t) = 0 \quad ...(19)$$

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.

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)
```


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```
    xymin = y;
  end
  if (xymin > 23)
    xymin = 23.
  end
% note that xymin varies from zero to 22.5 degrees.
```

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

In the front left quadrant

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

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

In the front right quadrant,

$$LRL = \cos(cs) \quad ...(20c)$$

$$LRR = -\sin(cs) \quad ...(20d)$$

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.

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

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

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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.

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

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.

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.

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:

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$$LRL = \cos(-cs) = sri(-cs) \quad ...(21a)$$

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

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

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):

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

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

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.

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

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

The behavior of the Left Rear Left and Left Rear Right elements is much

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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.

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.

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*).

$$LRL = \cos(45-lr)*\sin(4*(45-lr))-\sin(45-lr)*\cos(4*(45-lr))$$
$$= sra(lr) \quad ...(23a)$$

$$LRR = -(\sin(45-lr).*\sin(4*(45-lr)) + \cos(45-lr).*\cos(4*(45-lr)))$$
$$= -srac(lr) \quad ...(23b)$$

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

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:

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

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

The two functions sra(*x*) and srac(*x*) – are now defined for 0 < *lr* < 45.

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19. March 1997 version

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.

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).

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.

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));
```


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

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

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)
```

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

The '89 patent and Dolby Pro-Logic both have the following matrix elements:

For front steering:

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

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

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For rear steering:

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

$$CR = 1 \quad \text{...(24d)}$$

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.

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.

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

For front steering:

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

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

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.

In the version of March 1997 the boost function of *cs* starts at zero as

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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.

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.

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.

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23. Center channel in the new design

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?

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.

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.

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

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

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?

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

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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.

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.

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.

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.

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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).

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.

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

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?

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

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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.

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

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

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

For rear steering

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

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

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.

In MATLAB notation:

```
center_max = 0.65;
center_rate = 0.75;
```


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```
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
```

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..

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.

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We assume that:

$$LFL = GP(cs) \quad ...(28a)$$

$$LFR = GF(cs) \quad ...(28b)$$

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

$$CR = 0.42 + GC(cs) \quad ...(28d)$$

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

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

Power from the center is:

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

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,

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

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

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

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

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

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

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

...(31a)

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

For normal stereo, GC=0, GP=1, and GF=0. The center to LR power ratio

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is then:

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

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

$$((GP(cs)-GF(cs))^2 + 2*(GC(cs)^2 + 0.5*(\cos(cs)-\sin(cs))^2)$$
$$= ((GP(cs)^2 + GC(cs)^2 + GF(cs)^2 + 0.5) \quad ...(33)$$

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.

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.

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 $cs \leq 22.5$ 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);
```


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```
  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
```

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
```


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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.

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

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

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

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

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.

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,

$$CL = 0.42 - 0.42*G(lr) + GC(cs) + \text{center_fix_table}(xymin) \quad \text{...(35a)}$$

$$CR = 0.42 + GC(cs) + \text{center_fix_table}(xymin) \quad \text{...(35b)}$$

See FIG. 28 for a three-dimensional representation of the CL matrix

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element. While not perfect, this correction works well in practice.

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

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

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.

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

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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.

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.

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

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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.

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.

26. Explanation of the design

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.

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.

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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.

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.

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

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want.

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.

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.

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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.

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.

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

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is panning from a front input to the corresponding rear input.

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.

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

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.

27. Frequency dependent circuits in the decoder

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.

28. The background control signal

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

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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.

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.

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

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ordinary two channel recording will have no net rearward bias. Cs for this reverberation will be zero or slightly forward.

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.

29. Frequency dependent circuits: five channel version

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.

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.

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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.

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.

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.

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

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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.

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.

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.

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.

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30. Frequency dependent circuits: the seven channel version

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.

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.

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.

While the preferred embodiments of the invention have been described and illustrated herein, many other possible embodiments exist, and these and

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other modifications and variations will be apparent to those skilled in the art, without departing from the spirit of the invention.

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WHAT IS CLAIMED IS:

1. A surround sound decoder for redistributing a pair of left and right audio input signals including directionally encoded and non-directional components into a plurality of output channels 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 for receiving said corresponding left and right audio input signals;

left and right delay means for producing delayed left and right audio signals from said left and right audio input signals;

a plurality of multiplier mean 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 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 components in outputs which are not directly involved in reproducing them in the intended direction and enhance

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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 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.

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 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 and additional delay means 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 control signals responding to the presence of surround or background

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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 components contained in said left and right audio input signals during periods when no strongly steered signals are present.

6. The decoder of claims 1 through 4 wherein at least two different modes of operation are provided and wherein the matrix coefficients are controlled differently by said steering signals in the different modes of operation.

7. The decoder of claim 6 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.

8. The decoder of claim 7 wherein for decoding film sources the matrix values for the left and right front outputs are so constructed to eliminate as much as possible 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 decoders, and reduces rapidly as the center/surround steering signal becomes more positive,

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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.

9. The decoder of claim 7 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 decoder, and reduces gradually to the maximum value for a standard 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.

10. 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.

11. 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

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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.

12. 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.

13. The decoder of claim 12 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.

14. The decoder of claim 12 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

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ordinary two channel material is played.

15. The decoder of claim 12 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.

16. 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.

17. 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 signal is zero or positive the filter has its maximum effect, and as the center-surround steering signal varies from zero to –15 degrees the filter action is reduced to zero, and as the center-surround steering become further negative the filter once again acts maximally,

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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.

18. An encoder circuit for automatically mixing five channels of audio input signals down into two output channels such that the energy ratios in the input signals are preserved in the output signals, the direction of the input signals is preserved in the phase and amplitude relationships of different components in the output signals, and such that as an input signal is panned from any input channel to any other input channel the phase/amplitude relationships of the output signals decode as closely as possible to the original direction through a decoder according to claim 1, and also as closely as possible to the original direction when decoded through a standard film decoder, said encoder comprising:

five input terminals for receiving said five audio input signals respectively identified as left, center, right, left surround and right surround;

two output terminals each for providing one of said two output channels to external apparatus;

means for determining the amplitude/phase relationships between some of the said five audio input signals and producing control signals therefrom; and

means for mixing either fixed or variable proportions of each said input channel into one of said two output channels, said variable proportions being responsive to said control signals.

19. The encoder of claim 18 wherein the circuitry includes means to actively correct the mixing values such that the energy of the output signals

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matches the energy of the input signals when the same signal is applied to the rear inputs either in phase or in antiphase.

20. The encoder of claim 18 wherein the circuitry includes means to actively correct the mixing values such that when the same signal is applied to the rear inputs in antiphase the outputs of the encoder have a relative phase of 90 degrees, indicating an unsteered condition to a decoder.

21. The encoder of claim 18 wherein the circuitry includes means to actively remove the phase shifting networks in the rear channels when a signal is panned between one of the front inputs and a rear input on the same side, for example from the front left input to the rear left input.

22. The encoder of claim 18 where the circuitry includes means to actively determine the presence of a common signal on all three front inputs, and to adjust the mix level of the center channel to preserve the total power of such a common signal at the outputs of the encoder.

23. The encoder of claim 18 where the circuitry includes means of comparing the level of the rear channels to the level of the front three channels, and when the rear level is less than the front levels active circuitry to reduce the mixing levels of the rear channels by up to three decibels, such circuitry enabling the encoder to optimally encode music where the rear channels contain mostly reverberation.

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