

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



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.

5 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
10 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.

15 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
20 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
25 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,
30 decorrelated signals, will be reproduced with equal loudness in all output



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



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.

5 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
10 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
15 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,
20 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
25 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
30 right rear, with a reduced bandwidth sixth audio channel intended for very low



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



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



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

5 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,

10 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

15 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.

20 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

25 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.

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



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
5 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
10 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
15 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
20 level change in the front loudspeakers relative to the rear when a signal is panned 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
25 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
30 Griesinger U.S. Patent Application No. 08/742,460, as disclosed in the Provisional

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

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.

5 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.

10 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.

15 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.

20 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
25 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
30 when forming the left front output, and this low value is maintained until cs



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
5 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
10 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
15 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
20 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
25 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
30 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 passive matrix Dolby surround decoder
5 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
10 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 soundtracks including 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



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

5 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.

10 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;

15 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;

20 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 ;

25 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;

30 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



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;



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

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, according 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;

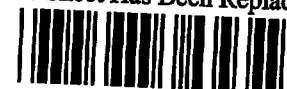
FIG. 37 shows the left front right (LFR) matrix element with the correction for center level along the $lr=0$ axis, indicating that the value is zero in the middle of the plane (no steering) and remains zero as cs increases to 22.5 degrees along the $lr=0$ axis, falling off to match the previous value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



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.

5 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
10 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
15 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.

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
25 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



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

$$\text{Left matrix element} = \cos(a/2) \quad \dots(1)$$

$$\text{Right matrix element} = \sin(a/2) \quad \dots(2)$$

5 Thus for $a = 90^\circ$, 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 $a = 0$ (full left) to $a = 180^\circ$ (full right), where the sign of the matrix element for left changes. For the left rear quadrant, a 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, $a = 270^\circ$ 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 a in equations (1) and (2) as $[-90^\circ, 270^\circ)$, 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 ($a = -90^\circ$) and left rear ($a = -45^\circ$) will have a separation given by:

$$\begin{aligned} \text{Separation} &= \cos(45^\circ) * L / (\cos(22.5^\circ) * L) = 0.77 \\ &= 2.3\text{dB} \quad \dots(3) \end{aligned}$$

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



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



power criterion.

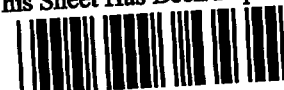
D. The transition between the reproduction of a decorrelated music component only, and the reproduction of a directional signal only, as their relative levels change, should occur smoothly and involve no shifts in the apparent direction of the sound. This criterion is also violated in various significant 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



with a requirement:

5 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
10 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
15 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.

15 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.

20 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
25 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
30 as the directional component of the output is steered to the front hemisphere.



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
5 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,
10 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.

15 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
20 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:

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

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

25 where

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

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
30 equations:



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

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

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

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.

10 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.

15 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, $t = 0^\circ$, to center, $t = 45^\circ$, the output amplitudes should be:

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

$$20 \quad \text{Center output} = \sin 2t \quad \dots(10)$$

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

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

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

$$\text{Center output} = \sin(2t - 90^\circ) = -\cos 2t \quad \dots(13)$$

$$25 \quad \text{Right output} = \cos(2t - 90^\circ) = \sin 2t \quad \dots(14)$$

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.

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



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

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

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

- 5 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.

10 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.

15 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
20 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.

- 25 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,
30 the standard passive matrix, described above with reference to FIG. 1, has a



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



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.

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

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

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

10 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.

15 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 $\phi-90^\circ$. The right surround signal 65 from input terminal 64
20 is applied to a $\phi-90^\circ$ 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.

25 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:

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

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

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

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.

5 For a monaural signal which has been distributed with the simplified encoder between the two input channels such that $A = \cos t$ and $B = \pm \sin t$, 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:

$$10 \quad t = 90^\circ - \arctan(10^{((l/r) / 20)}) \quad \dots(21)$$

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.

15 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.

20 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.

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



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

$$ts = \arctan(10 \wedge ((c/s) / 20)) - 45^\circ \quad \dots(22)$$

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 the simplified 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:

$$\begin{aligned} \text{Left output} &= LL \cos ts + LR \sin ts \\ &= FL(ts) = \cos(2ts) \end{aligned} \quad \dots(23)$$

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

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

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,

$$LR = \sin ts \cos(2ts) \pm \cos ts \sin(2ts) \quad \dots(25)$$

$$LL = \cos ts \cos(2ts) \mp \sin ts \sin(2ts) \quad \dots(26)$$

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

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

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

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

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

Once again, the decorrelated music should be constant, so

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



and these lead by similar reasoning to the result

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

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

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.

$$tl = 90^\circ - \arctan(10^{(l/r)/20}) \quad \dots(33)$$

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 $\sin(2tl)$ in this embodiment. By the above method,

$$\begin{aligned} \text{Center output} &= CL \cos tl + CR \sin tl = FC(tl) \\ &= \sin(2tl) \end{aligned} \quad \dots(34)$$

Once again, for constant loudness of the music,

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

which yields the solutions

$$CR = \sin tl \sin(2tl) \text{ } -/+ \cos tl \cos(2tl) \quad \dots(36)$$

$$CL = \cos tl \sin(2tl) \text{ } +/- \sin tl \cos(2tl) \quad \dots(37)$$

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



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:

$$ts = \arctan(c/s) - 45^\circ \quad \dots(38)$$

$$tl = \arctan(l/r) - 45^\circ \quad \dots(39)$$

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,

$$GL = ((\cos tl - \sin tl)/\cos tl) = 1 - \tan tl \quad \dots(40)$$

$$GC = 2 (\sin ts/(\cos ts + \sin ts)) \quad \dots(41)$$

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



$$GS = (\cos ts + \sin ts) \cdot \frac{e}{e} - \frac{(1 - \sin tl)}{\cos tl} \cdot \frac{(\cos tl - \sin tl)}{\cos tl} - 0.5 \cdot \frac{2 \sin ts}{(\cos ts + \sin ts)} \frac{u}{u}$$

= 0

...(42)

5 (since this is a front quadrant), and

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

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

$$= (1 - \tan tl) (\sec tl - \tan tl) = (\cos ts + \sin ts) \cdot \frac{e}{e} - \frac{(\cos tl - \sin tl)}{\cos tl} - 0.5 \cdot \frac{2 \sin ts}{(\cos ts + \sin ts)} \frac{u}{u}$$

10

...(43)

and the left and right supplemental signals are given by:

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

15

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

Thus, the coefficients LSL and LRL are given by:

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

which becomes, after some manipulation,

$$20 \quad LSL = LRL = (\cos ts + \sin ts) (\sec tl - 1) \times (\sec tl - \tan tl) - \sin ts \quad \dots(47)$$

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,

$$25 \quad LSR = LRR = (\cos ts + \sin ts) (\tan tl - 1) - \sin ts \quad \dots(49)$$

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



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:

$$\text{Right rear output} = \text{RRL} \cos ts - \text{RRR} \sin ts = 0 \quad \dots(50)$$

5 and

$$\text{RRL}^2 + \text{RRR}^2 = 1 \quad \dots(51)$$

which yield the solution:

$$\text{RRR} = \text{RSR} = \cos ts$$

$$\text{RRL} = \text{RSL} = \sin ts \quad \dots(52)$$

10 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:

$$ts = \arctan(10^{(s/c)/20}) - 45^\circ \quad \dots(53)$$

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

$$\text{LL} = \cos ts \quad \dots(54)$$

$$\text{LR} = \sin ts \quad \dots(55)$$

and

$$\text{RL} = \sin ts \quad \dots(56)$$

$$20 \quad \text{RR} = \cos ts \quad \dots(57)$$

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
25 be removed when there is strong left steering.

We use the previous definition for tl for center steering,

$$tl = 90^\circ - \arctan(10^{((l/r)/20)}) \quad \dots(58)$$

as tl varies from 0 to 22.5° . Under strong steering, the left side and left rear outputs are zero when $tl = 0^\circ$, but increase with tl according to the value $\sin 4tl$.

30 In the presence of uncorrelated music, represented by the signals $A = \cos t$, $B =$



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

$$\text{LSL} = \text{LRL} \quad \dots(59)$$

$$\text{LSR} = \text{LRR} \quad \dots(60)$$

to have equal outputs at the sides and rear, and the amplitude during steering

5 follows $\text{FS}(tl) = \sin 4tl$, so that

$$\text{LSL} \cos tl - \text{LSR} \sin tl = \text{FS}(tl) \quad \dots(61)$$

For the music to have constant level,

$$\text{LSL}^2 + \text{LSR}^2 = 1 \quad \dots(62)$$

Solving as before,

$$10 \quad -\text{LSR} = \sin tl \text{FS}(tl) \pm \cos tl \sqrt{1 - \text{FS}(tl)^2} \quad \dots(63)$$

$$\text{LSL} = \cos tl \text{FS}(tl) \mp \sin tl \sqrt{1 - \text{FS}(tl)^2} \quad \dots(64)$$

Simplifying and using the preferred sign, as before,

$$-\text{LSR} = \sin tl \sin 4tl + \cos tl \cos 4tl \quad \dots(65)$$

$$\text{LSL} = \cos tl \sin 4tl - \sin tl \cos 4tl \quad \dots(66)$$

15 which may be further reduced to:

$$-\text{LSR} = \cos 3tl \quad \dots(67)$$

$$\text{LSL} = \sin 3tl \quad \dots(68)$$

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

20 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:

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

$$\text{RSL}^2 + \text{RSR}^2 = 1 \quad \dots(70)$$

25 which yield the solution:

$$\text{RSL} = \sin ts \quad \dots(71)$$

$$\text{RSR} = \cos ts \quad \dots(72)$$

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



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
 5 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
 10 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 $tl = 22.5^\circ$ to $tl = 45^\circ$, this ratio expressed in dB decreases to zero, while the log of the center to
 15 surround ratio (c/s) becomes a large negative value.

Consider what happens when a directional signal at $tl = 22.5^\circ$ 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
 20 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:

$$tl = 90^\circ - \arctan(l/r) \quad \dots(73)$$

$$LRL = \cos(45^\circ - tl) \quad \dots(74)$$

$$LRR = -\sin(45^\circ - tl) \quad \dots(75)$$

25 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
 30 constant, we can express the results in a table:

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:

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

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

and similarly for the right channel,

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

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

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°.



Then

$$\text{RSBOOST} = 0.5 \sin(2(ts - 22.5^\circ)) \quad \dots(80)$$

and for the side outputs,

$$\text{LSL} = \cos(45^\circ - tl) + \text{RBOOST}(\log c/s) - \text{RSBOOST}(ts) \quad \dots(81)$$

$$5 \quad \text{LSR} = -\sin(45^\circ - tl) \quad \dots(82)$$

$$\text{RSL} = \sin(45^\circ - tl) \quad \dots(83)$$

$$\text{RSR} = \cos(45^\circ - tl) + \text{RBOOST}(\log c/s) - \text{RSBOOST}(ts) \quad \dots(84)$$

and for the rear outputs,

$$10 \quad \begin{aligned} \text{LRL} &= \cos(45^\circ - tl) + \text{RBOOST}(\log c/s) \\ &+ 0.5 \text{RSBOOST}(ts) \end{aligned} \quad \dots(85)$$

$$\text{LRR} = -\sin(45^\circ - tl) \quad \dots(86)$$

$$\text{RRL} = \sin(45^\circ - tl) \quad \dots(87)$$

$$\begin{aligned} \text{RRR} &= \cos(45^\circ - tl) + \text{RBOOST}(\log c/s) \\ &+ 0.5 \text{RSBOOST}(ts) \end{aligned} \quad \dots(88)$$

15 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
20 by a boost function depending on l/r , where we define two angles:

$$tlr = 90^\circ - \arctan(l/r) \quad \dots(89)$$

$$trl = 90^\circ - \arctan(r/l) \quad \dots(90)$$

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

$$\text{LFL} = \cos ts + \text{LFBOOST}(tlr) \quad \dots(91)$$

25 and for steering to the right,

$$\text{RFR} = \cos ts + \text{LFBOOST}(trl) \quad \dots(92)$$

Both center matrix elements are also boosted during center steering:

$$\text{CL} = \sin tl + 0.71 \text{LFBOOST}(ts) \quad \dots(93)$$

$$\text{CR} = \cos tl + 0.71 \text{LFBOOST}(ts) \quad \dots(94)$$

30 These equations completely specify the additional requirements for a film



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 $0.5/1.5 = 9.5\text{dB}$.

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:

$$\text{LFL} = 1 + \text{LFBOOST}(ts) \quad \dots(95)$$

$$\text{RFR} = 1 + \text{LFBOOST}(ts) \quad \dots(96)$$

$$\text{LFR} = \text{RFL} = 0 \quad \dots(97)$$

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



adding the appropriate boost expressions derived from the angles t_s and t_l 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.

5 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.

10 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
15 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.

20 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
25 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.



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



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
5 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
10 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
15 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
20 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
25 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



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



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 $\cos 22.5^\circ$ or 0.92 and $\sin 22.5^\circ$ 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,



when steering between left rear (-8dB) to full rear (-33dB). Again, the axis is $-(c/s \text{ in dB})$, 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 $\phi-90^\circ$ 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 $\phi-90^\circ$ and ϕ respectively in the RS channel from attenuators 258



and 260 respectively.

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

A similar matrix 252 sums the $RS(\phi)$ signal with gain $\sin \Theta_{RS}$, the $RS(\phi-90^\circ)$ signal with gain $\cos \Theta_{RS}$, the $R(\phi)$ signal, the $C(\phi)$ signal with gain 0.707, and the $S(\phi)$ 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



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 $\phi(f)-90^\circ$ 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
5 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.

10 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
15 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
20 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
25 328 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



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.

5 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
10 having the attenuation value $\cos \Theta_{RS}$, while the phase shifter 346, which may be the same shifter as 306 in FIG. 8, of phase function $\phi-90^\circ$, is passed through attenuator 348 with attenuation factor $\sin \Theta_{RS}$. 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.

15 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 $\cos \Theta_{LS}$, while in the other branch it passes through phase shifter 358 of phase function $\phi-90^\circ$ and attenuator 360 of attenuation factor $\sin \Theta_{LS}$. The signals from attenuators 356 and 360 are
20 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.

25 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



controlled by the inverse of l_s/r_s , which we call r_s/l_s , so that when r_s/l_s exceeds 3dB, the value of Θ_{rs} begins to increase from 0° , moving towards an asymptote at -65° when r_s/l_s 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
 5 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 be reduced by combining the real signals before sending them through the ϕ phase shifter, thus resulting in only two ϕ and two $\phi-90^\circ$ phase
 10 shift networks. The description of Θ_{ls} and Θ_{rs} is also simplified. Θ_{ls} approaches 90° when l_s/r_s is greater than 3dB, and otherwise is zero (just as in the decoder design). Likewise, Θ_{rs} approaches 90° when r_s/l_s is greater than 3dB, and otherwise is zero.

In FIG. 10, elements corresponding to those in the right half of FIG. 7,
 15 namely the attenuators 254-260 and the $\phi-90^\circ$ 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
 20 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
 25 passes as before through attenuator element 256 and $\phi-90^\circ$ phase shifter 234, then passing through the actively controlled attenuator 270 having attenuation factor $\cos \Theta_{ls}$, 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.



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 $\cos \Theta_{RS}$, 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 $\sin \Theta_{LS}$ 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/l_s (or $-l_s/rs$) 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



value of l_s/r_s , 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 G_L , G_C and G_R . 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 G_L and G_R respectively



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
5 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
10 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
15 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
20 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
25 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
30 signal, and this can be accomplished by reducing the value of GS whenever there



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 tl/s in attenuators 270 and 272, and its inverse, rs/lr , controls the steering angle tr/s 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



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



control signals are

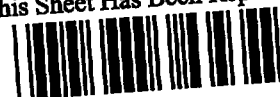
$$l/r = 20 * \log_{10}(\cos t / \sin t) \quad \dots(98)$$

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

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

5 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-
10 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
15 "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
20 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
25 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
30 discontinuity was due to a programming error in the decoder version 1.11. The



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,

$$\begin{aligned} \text{LFL} &= 1 - 0.5 * G(cs) + 0.41 * G(lr) & \dots(100) \\ \text{LFR} &= -0.5 * G(cs) & \dots(101) \end{aligned}$$

In the right front quadrant,

$$\text{LFL} = 1 - 0.5 * G(cs) \quad \dots(102)$$

$$\text{LFR} = -0.5 * G(cs) \quad \dots(103)$$

In the left rear quadrant,

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

$$\text{LFR} = 0.5 * G(-cs) \quad \dots(105)$$

In the right rear quadrant,

$$\text{LFL} = 1 - 0.5 * G(-cs) \quad \dots(106)$$

$$\text{LFR} = 0.5 * G(-cs) \quad \dots(107)$$

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



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
 5 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
 10 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
 15 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:

20 For the left front quadrant

$$\text{LFL} = \cos(cs) + 0.41 * G(lr) \quad \dots(108)$$

$$\text{LFR} = -\sin(cs) \quad \dots(109)$$

For the right front quadrant

$$\text{LFL} = \cos(cs) \quad \dots(110)$$

25 $\text{LFR} = -\sin(cs) \quad \dots(111)$

For the left rear quadrant

$$\text{LFL} = \cos(-cs) + 0.41 * G(lr) \quad \dots(112)$$

$$\text{LFR} = \sin(-cs) \quad \dots(113)$$

For the right rear quadrant

30 $\text{LFL} = \cos(-cs) \quad \dots(114)$



$$\text{LFR} = \sin(-cs) \quad \dots(115)$$

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,

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

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

For the right front quadrant,

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

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

For the left rear quadrant,

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

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

For the right rear quadrant,

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

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

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 $\text{boost1}(cs)$ 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.) $\text{Boost}(cs)$ is



given by $\text{corr}(x)$ in the code below, in which comments are preceded by the % symbol:

```

5      % 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
10      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 performance can 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



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
 5 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:

```

10      % find the bounded c/s
      if (cs < 24)
          bcs = cs-(lr-1);
          if (bcs<1) % this limits the maximum value
              bcs = 1;
15      end
      else
          bcs = 47-cs-(lr-1);
          if (bcs <1)
              bcs = 1;
20      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.

25 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..)



$a = 0:45$ % 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

5 front

$\sin_tbl = \sin(a1)$;

$\cos_tbl = \cos(a1)$;

$\cos_tbl_plus = \cos(a1).*\text{corr1}(a+1)$;

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

10

$\cos_tbl_minus = \cos(a1)./ \text{corr}(a+1)$;

$\sin_tbl_plus = \sin(a1).*\text{corr1}(a+1)$;

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

$\sin_tbl_minus = \sin(a1)./ \text{corr}(a+1)$;

$\sin_tbl_plus$ and $\cos_tbl_plus$ are the difference between a plain sine and

15 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.

20 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

25 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

30 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 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.

5 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, 10 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:

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

$$15 \quad \text{LFR} = -(\sin(t)*F(t) +/- \cos(t)*(\text{sqrt}(1-F(t)^2))) \quad \dots(129)$$

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

$$\text{LFL} = \cos(t)*\cos(4*t) + \sin(t)*\sin(4*t) \quad \dots(130)$$

$$\text{LFR} = -(\sin(t)*\cos(4*t) - \cos(t)*\sin(4*t)) \quad \dots(131)$$

20 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,

$$25 \quad \text{LFL} = -\sin(t) \quad \dots(132)$$

$$\text{LFR} = \cos(t) \quad \dots(133)$$

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

$$\text{LFL} = \cos(cs) \quad \dots(134)$$

$$30 \quad \text{LFR} = \sin(cs) \quad \dots(135)$$

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 decoders, where multiplies are expensive, a better strategy is to define a new variable – the minimum of *lr* and *cs*:

% new - find the boundary parameter

```
10      bp = x;
      if (bp > y)
          bp = y;
      end
```

and a new correction function which depends on *bp*:

```
15      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
20          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:

$$\text{LFL} = \cos(cs)/(\cos(cs)+\sin(cs)) - \text{front_boundary_table}(bp) + 0.41 \cdot G(lr) \quad \dots(136)$$

$$\text{LFR} = \sin(cs)/(\cos(cs)+\sin(cs)) + \text{front_boundary_table}(bp) \quad \dots(137)$$

Note the correction of $\cos(cs)+\sin(cs)$. When we divide $\cos(cs)$ by this factor we get the function $1 - 0.5 \cdot G(cs)$, which is the same as the Dolby matrix in this quadrant.

30 In the right rear quadrant,

This Sheet Has Been Replaced



$$\text{LFL} = \cos(cs)/(\cos(cs)+\sin(cs)) \quad \dots(138)$$

$$\text{LFR} = \sin(cs)/(\cos(cs)+\sin(cs)) \quad \dots(139)$$

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
 5 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
 10 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
 15 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
 20 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.

25 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
 30 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.

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:

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

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

For the rear left:

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

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

(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:

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

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

For the rear left:

$$\text{LRL} = .71 - .71 * G(lr) \quad \dots(146)$$

$$\text{LRR} = .71 * G(lr) \quad \dots(147)$$

(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



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:

$$\text{LRL} = 1 - G(\text{lr}) \quad \dots(148)$$

$$\text{LRR} = -1 \quad \dots(149)$$

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 $.71 \cdot (A_{in} + B_{in})$, 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



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.

$$A_{in} = L_{in} + .71 * C_{in} \quad \dots(150)$$

$$B_{in} = R_{in} + .71 * C_{in} \quad \dots(151)$$

When A_{in} and B_{in} are played in stereo, 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 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. 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

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.

5 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.

10 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
15 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:

$$20 \quad \text{LRL} = 1 - \text{GS}(lr) - 0.5 * \text{G}(cs) \quad \dots(152)$$

$$\text{LRR} = - 0.5 * \text{G}(cs) - \text{G}(lr) \quad \dots(153)$$

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 $\text{G}(lr)$ term in LRR and a $\text{GS}(lr)$ term in LRL. $\text{G}(lr)$ was included to add signals from the B input
25 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 $\text{GS}(lr)$, using the criterion that there should be no signal output with a fully steered signal moving from left to center. The formula for $\text{GS}(lr)$ turned out to be equal to $\text{G}^2(lr)$, although a more complicated formula is given in the 1991 patent
30 (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



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:

$$LRL = \cos(cs) - GS(lr) \quad \dots(154)$$

$$LRR = -\sin(cs) - GR(lr) \quad \dots(155)$$

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

$$(1 - GS(lr))^2 + (GR(lr))^2 = 1 \quad \dots(156)$$

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

$$LRL \cdot \cos(t) + LRR \cdot \sin(t) = 0 \quad \dots(157)$$

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

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.

5 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
10 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;
15      if (xymin > y)
        xymin = y;
      end
      if (xymin > 23)
        xymin = 23;
20      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

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

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

In the front right quadrant,

$$\text{LRL} = \cos(cs) \quad \dots(160)$$

$$\text{LRR} = -\sin(cs) \quad \dots(161)$$

30 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



tvcorr($|lr| + |cs|$). Tvcrr($|lr| + |cs|$) 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($|lr|$). Tvcrr($|lr|$) 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

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
5 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
10 new functions which are simply equal to $\sin$ and $\cos$ of cs over this range.

$$LRL = \cos(-cs) = sri(-cs) \quad \dots(162)$$

$$LRR = \sin(-cs) = sric(-cs) \quad \dots(163)$$

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
15 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
20 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.

$$25 \quad LRL = \cos(-cs) * tvcorr(|lr| + |cs|) = sri(-cs) * tvcorr(|lr| + |cs|) \quad \dots(164)$$

$$LRR = \sin(-cs) * tvcorr(|lr| + |cs|) = sric(-cs) * tvcorr(|lr| + |cs|) \quad \dots(165)$$

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
30 steering.

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



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)

$$5 \quad \text{LRL} = (\cos(45+cs) + \text{rboost}(-cs)) = (\text{sri}(-cs) + \text{rboost}(-cs)) \quad \dots(166)$$

$$\text{LRR} = \sin(45+cs) = \text{sric}(-cs) \quad \dots(167)$$

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
10 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
15 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
20 equations for the Left Rear matrix elements over the range $22.5 < lr < 45$:
(remember that $t = 45 - lr$)

$$\begin{aligned} \text{LRL} &= \cos(45-lr) * \sin(4*(45-lr)) - \sin(45-lr) * \cos(4*(45-lr)) \\ &= \text{sra}(lr) \end{aligned} \quad \dots(168)$$

$$\begin{aligned} \text{LRR} &= -(\sin(45-lr) * \sin(4*(45-lr)) + \cos(45-lr) * \cos(4*(45-lr))) \\ &= -\text{srac}(lr) \end{aligned} \quad \dots(169)$$

If $cs \leq 22.5$, 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.

$$\text{LRL} = \cos(lr) = \text{sra}(lr) \quad \dots(170)$$

$$\text{LRR} = -\sin(lr) = -\text{srac}(lr) \quad \dots(171)$$

30 The two functions $\text{sra}(x)$ and $\text{srac}(x)$ are defined for $0 < lr < 45$.

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



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

5 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;
```

```
10 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 $0 < x < 45$

```
15 Rear_active_correct(x) = sra(x) - (1 - GSL(x)) ... (172)
```

```
LRL = (sri(cs) + sra(lr) - rear_active_correct(lr_bounded) - 1)
```

```
*tvcorr(|lr| + |cs|) ... (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

20 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.

25 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 $srac(lr)$.

30 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



is simply `sric(cs_bounded)`. This term will insure continuity across the $l r=0$ axis.

First we define `cs_bounded`

```
cs_bounded = lr - cs;
```

```
if(cs_bounded <1) % this limits the maximum value
```

```
5         cs_bounded = 0;
```

end

```
if (45-|lr| < cs_bounded) % use the smaller of the two values
```

```
cs_bounded = 45-lr;
```

end

```
10      for cs = 0 to 15
```

$$\text{LRR} = (-(\text{srac}(\text{lr}) + (\text{srac}(\text{lr}) - \text{G}(\text{lr})) * (15 - \text{cs}) / 15)$$

```
+sric(cs_bounded))*tvcorr( |lr| + |cs| );
```

for cs = 15 to 22.5

$$\text{LRR} = (-\text{srac}(\text{lr}) + \text{sric}(\text{cs_bounded})) * \text{tvcorr}(|\text{lr}| + |\text{cs}|);$$

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

```
for cs = 0 to 15
```

$$\text{LRL} = ((\text{sra}(\text{lr}) + (\text{sra}(\text{lr}) - \text{GS}(\text{lr})) * (15 - \text{cs}) / 15)$$

```
+ sri(-cs))*tvcorr( |lr| + |cs| );
```

20 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 $cs > 22.5$, $lr < 22.5$

$$25 \quad \text{LRL} = (\text{sra}(\text{lr}) + \text{sri}(\text{cs}) + \text{rboost}(\text{cs})) \quad \dots(174)$$
$$\text{LRR} = -\text{srac}(\text{lr}) + \text{srac}(\text{cs_bounded}) \quad \dots(175)$$

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.

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

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

$$5 \quad CL = 1 + .41 * G(cs) - G(lr) \quad \dots(176)$$

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

For rear steering

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

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

10 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, 15 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

$$20 \quad CL = \cos(45-lr) * \sin(2*(45-lr)) - \sin(45-lr) * \cos(2*(45-lr)) + .41 * G(cs) \quad \dots(180)$$

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

These equations were never implemented. The version 1.11 implementation 25 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 30 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



risers 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
5 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
10 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.

15 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
20 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-
25 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
30 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?



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
 5 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)
 10 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:

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

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

15 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
 20 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
 25 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
 30 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.

5 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
10 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
15 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.

20 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
25 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
30 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 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,

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

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

For rear steering,

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

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

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;

```

    if (cs < 12)
        gc(cs+1) = .42*10^(db*center_rate/(20));
        tmp = gc(cs+1);
    elseif (cs < 30)
5       gc(cs+1) = tmp*10^((cs-11)*center_rate3/(20));
        if (gc(cs+1) > center_max)
            gc(cs+1) = center_max;
        end
    else
10      gc(cs+1) = center_max*10^((cs-29)*center_rate2/(20));
        if (gc(cs+1) > center_max2)
            gc(cs+1) = center_max2;
        end
    end
end

```

15 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

20 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:

$$\text{LFL} = \text{GP}(cs) \quad \dots(188)$$

$$\text{LFR} = \text{GF}(cs) \quad \dots(189)$$

25 $\text{CL} = .42 - .42 * \text{G}(lr) + \text{GC}(cs) \quad \dots(190)$

$$\text{CR} = .42 + \text{GC}(cs) \quad \dots(191)$$

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

$$\text{PLR} = (\text{GP}^2 + \text{GF}^2) * (\text{Lin}^2 + \text{Rin}^2) + (\text{GP} - \text{GF})^2 * \text{Cin}^2$$

...(192)

30 Power from the center speaker is:



$$PC = GC^2 * (Lin^2 + Rin^2) + 2 * GC^2 * Cin^2 \quad \dots(193)$$

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,

$$5 \quad PREAR = (.71 * (\cos(cs) * (Lin + .71 * Cin) - \sin(cs) * (Rin + .71 * Cin)))^2 \quad \dots(194)$$

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

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

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

$$10 \quad PT = (GP^2 + GF^2 + GC^2) * (Lin^2 + Rin^2) + ((GP - GF)^2 + 2 * GC^2) * Cin^2 + PREAR \quad \dots(196)$$

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

$$15 \quad \begin{aligned} \text{RATIO} &= ((GP^2 + GF^2 + GC^2) * (Lin^2 + Rin^2) + ((GP - GF)^2 + 2 * GC^2) * Cin^2 + PREAR) / PREAR \\ &= (((gp(cs) - gf(cs))^2 + 2 * gc(cs)^2 + .5 * (\cos(cs) - \sin(cs))^2 * Cin^2 / \\ &\quad 2 * (gp(cs)^2 + gc(cs)^2 + gf(cs)^2) + 1) * Lin^2 \\ &= (Cin^2 / Lin^2) * ((gp(cs) - gf(cs))^2 + 2 * (gc(cs)^2) + .5 * (\cos(cs) \\ 20 \quad &\quad - \sin(cs))^2) / 2 * (gp(cs)^2 + gc(cs)^2 + gf(cs)^2) + 1) \quad \dots(197) \end{aligned}$$

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

$$\text{RATIO} = (Cin^2 / Lin^2) * 0.5 \quad \dots(198)$$

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

$$25 \quad \begin{aligned} &((gp(cs) - gf(cs))^2 - 2 * gc(cs)^2) + .5 * (\cos(cs) - \sin(cs))^2 \\ &= (gp(cs)^2 + gc(cs)^2 + gf(cs)^2) - .5) \quad \dots(199) \end{aligned}$$

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, $\sin(cs) * \text{corr1}$ (the previous LFR element) in the dashed curve, and



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
 5 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
 10 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);

15 for cs = 0:45;

if (gf_diff(cs) > sin(cs))

gf_diff(cs) = sin(cs);

end

if (gf_diff(cs) < 0)

20 gf_diff(cs) = 0;

end

end

% find the bounded c/s

if (y < 24)

25 bcs = y-(x-1);

if (bcs<1) % this limits the maximum value

bcs = 1;

end

else

30 bcs = 47-y-(x-1);

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```
if (bcs < 1) %> 46)
```

```
    bcs = 1; %46;
```

```
end
```

```
end
```

5 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);
```

10 else

```
    tmp = ((47-y-x)/(47-y))*gf_diff(y);
```

```
    lfr3d(47-x,47-y) = -sin_tbl(y)+tmp;
```

```
end
```

15 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.

25 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):

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

$$CR = .42 + GC(cs) \quad \dots(201)$$

30 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



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



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
5 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.

10 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
15 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.

20 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.
25 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,
30 even without the active correction, the performance is better than that of the



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 f_{cn} 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 f_{cn} 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 $f_c()$ and $f_s()$ direct the surround channels either to a path (through phase shift elements 234 and 246) with a 90 degree phase shift



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, f_c is one and f_s 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

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.

5 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
10 signal. Thus, active circuitry (not shown, but similar to that in the decoder) increases the value of f_s 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.

15 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
20 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
25 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.

30 A major function, therefore, of the function f_c 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 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 f_c 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 f_s 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, f_c is reduced to zero while f_s 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;



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

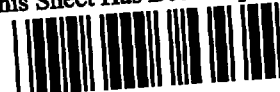
4. Increase the level and negative phase of the rear channels when their level is similar to the front channels; and

5 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,
10 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
15 departing from the spirit of the invention.

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1 1. A surround sound decoder for redistributing a pair of left and right
2 audio input signals including directionally encoded and non-directional
3 components into a plurality of output channels for reproduction through
4 loudspeakers surrounding a listening area, and incorporating means for
5 determining the directional content of said left and right audio signals and
6 generating therefrom at least a left-right steering signal and center-surround
7 steering signal, the decoder comprising:

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

10 a plurality of multiplier means equal to twice the number of said plurality
11 of output channels, organized in pairs, a first element of each said pair receiving
12 said delayed left audio signal and a second element receiving said delayed right
13 audio signal, each of said multiplier means multiplying its input audio signal by a
14 variable matrix coefficient to provide an output signal;

15 said variable matrix coefficient being controlled by one or both of said
16 steering signals; and

17 a plurality of summing means one for each of said plurality of output
18 channels each said summing means receiving the output signals of a pair of said
19 multiplier means and producing at its output one of said plurality of output
20 signals,

21 the decoder having said variable matrix values so constructed as to reduce
22 directionally encoded audio components in outputs which are not directly involved
23 in reproducing them in the intended direction; enhance directionally encoded
24 audio components in the outputs which are directly involved in reproducing them
25 in the intended direction so as to maintain constant total power for such signals;
26 while preserving high separation between the left and right channel components of
27 non-directional signals regardless of the said steering signals; and maintaining the
28 loudness defined as the total audio power level of non-directional signals
29 effectively constant whether or not directionally encoded signals are present and
30 regardless of their intended direction if present.

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

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

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

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

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

1 7. The decoder of claim 3 wherein the said variable matrix coefficients
2 determining the level of the rear output signal is maintained at a 3dB lower level
3 when the said left-right steering signal is of small magnitude and rises to the full
4 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





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

1 12. An active encoder suitable for encoding five channels of full
2 bandwidth audio onto two output channels, said five input channels being
3 respectively left front, center front, right front, left rear or surround, and right
4 rear or surround, and said output channels comprising left or A and right or B
5 channels respectively, comprising:
6 first, second, third, fourth and fifth input terminals for said left front,
7 center front, right front, left surround, and right surround channels respectively;
8 an attenuator circuit connected to said second input terminal for
9 attenuating said center front signal by a factor f_{en} ;
10 first and second summing circuits, said first summing circuit receiving the
11 attenuated center front signal from said attenuator circuit and the left front input
12 from said first input terminal directly, and said second summing circuit also
13 receiving said attenuated center front signal and the right front signal from said
14 third input terminal directly;
15 first functional attenuator with attenuation function $f_c(l, l_s)$ for attenuating
16 said left surround signal received from said fourth input terminal;
17 second functional attenuator with attenuation function $f_c(r, r_s)$ for
18 attenuating said right surround signal received from said fifth input terminal;
19 third functional attenuator with attenuation function $f_s(l, l_s)$ for attenuating
20 said left surround signal received from said fourth input terminal;
21 fourth functional attenuator with attenuation function $f_s(r, r_s)$ for
22 attenuating said right surround signal received from said fifth input terminal;
23 first, second, third and fourth cross-feed attenuators having an attenuation
24 factor $-cr_x$ each receiving the output signal from said first, second, third and
25 fourth functional attenuators, respectively;
26 first, second, third and fourth fixed attenuators having an attenuation
27 factor of 0.91 each receiving the output signal from said first, second, third and





28 fourth functional attenuators, respectively;

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

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

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

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

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

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

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

52 means responsive to the output of said first comparing means for reducing
53 the function $f_c(l,ls)$ of said first functional attenuator from one to zero and
54 increasing the function $f_s(l,ls)$ of said third functional attenuator from zero to one
55 in a complementary manner as the predominant signal direction changes from left
56 front towards left surround;

57 means responsive to the output of said second comparing means for
58 reducing the function $f_c(r,rs)$ of said second functional attenuator from one to zero



59 and increasing the function $f_s(r,rs)$ of said fourth functional attenuator from zero
60 to one in a complementary manner as the predominant signal direction changes
61 from right front towards right surround;
62 third comparing means for comparing the log amplitudes of the front signals
63 with those of the rear signals;
64 means responsive to the output signal from said third comparing means for
65 reducing the function f_{cn} by up to 3dB as the front signals exceed the rear signals
66 by up to 8dB or more;
67 fourth comparing means for comparing the phase and amplitude between
68 the left surround and right surround input signals applied to said fourth and fifth
69 input terminals; and
70 means responsive to the output signal from said fourth comparing means
71 for reducing the gain of said first and second functional attenuators by up to
72 2.2dB when the two surround signals are similar in amplitude and in phase, and
73 for increasing the gains of said third and fourth functional attenuators when the
74 two surround signals are similar in amplitude but in antiphase, thereby forcing
75 the A and B outputs to be in quadrature phase relationship representing an
76 unsteered condition.

1 13. The encoder of claim 12 further comprising:
2 a sixth input terminal for receiving a low frequency effects signal;
3 a gain stage of gain 2.0 for amplifying said low frequency effects signal;
4 said amplified low frequency effects signal being applied equally to
5 additional inputs of said first and second summing circuits so as to appear equally
6 and in phase in both A and B outputs at said first and second output terminals
7 respectively.

This Sheet Has Been Replaced

