

A SYSTEM AND METHOD FOR EXPANDING MULTI-SPEAKER PLAYBACK

Field of the Invention

[0001] The invention relates generally to audio signal post-processing, and specifically to playing back stereo audio contents on multiple speakers.

[0002] More specifically, it is related to the detecting the directional cues from the stereo audio signals and reproducing sounds with surrounding effects utilizing more than two speakers.

Background of the Invention

[0003] Multi-speaker systems are widely available on audio, video and entertainment systems today. Quadratic 4-speaker systems and 5.1 systems such as shown in FIG.1 provide listeners with more enjoyable multi-media experiences than traditional stereo systems. However, there are large quantities of stereo audio content that are in stock and also currently being produced that do not fully utilize the advantages of such multi-speaker systems. There have been difficulties in attempting to make better utilization of multi-speaker systems to give more listening pleasure than straightforward stereo playback.

[0004] Many methods have been developed in attempting to better render the stereo audio content utilizing more than two speakers to create a surrounding sound field. Among the many attempts, many follow the belief that in-phase information

between two signals should be placed at the center front and out-of-phase information should be pushed to the rear.

[0005] In US patent 3,959,590, Scheiber discloses a system for decoding stereo audio to four output signals. The system makes use of a gain-controlled first and second input channel to generate a first and second output channel. A third output channel is generated by combining the first and second input channels. A fourth output channel is generated by subtracting the first and second input channels. All the output channels are gain-controlled. The gains are derived by comparing the sum and the difference of the logarithmic values of the absolute values of the two input signals. Smoothing filters then are used to smooth out the rapid fluctuations of the gains.

[0006] Efforts have also been made on achieving better channel separation. In US patent 4,799,260, Mandell developed a system based on the assumption that there is only one dominant output signal at one time. In US patent 4,862,502, Griesinger developed a similar system with simple logarithmic operation and low pass filter to smooth the gains. In US patent 5,666,424 and 5,666,640, Fosgate uses a servo logic circuit to control the output gain of each channel. All of these developments used the logarithmic value of either the two input signals or the sum and difference of the two input signals to control the gains of each output channel. While these algorithms achieved good channel separation, the relatively high computational cost raised some concerns.

[0007] Several developments also concentrated on decoding stereo audio into multi-channel audio. Instead of aiming at better channel separation, other effects that altered the stereo content mixers' intention were included. In US patent 5,319,713, Waller mixed the left input signal with lowpassed right input signal to generate left output signal and mixed the right input signal with lowpassed left input signal to

generate right output signal. It provided a widened frontal sound image, but also made the sound effect unnatural. In US patent 6,198,826, Cowieson generated spatial effects by filtering the difference between the two input signals and output it out-of-phase to both front and rear speakers.

[0008] While there have been several developments in improving sound quality via channel separation or other sound effects, the present state of the art is still lacking in having sound systems for multi-speaker rendition of stereo audio content that achieves good channel separation with minimal and scalable computational costs. Furthermore, present developments have not been able to offer efficient solutions to input signals having more than one dominant signal.

[0009] Therefore, a need clearly exists for a method and system for expanding multiple speaker playback that overcomes the problems or at least alleviate the limitations of existing systems.

Summary of the Invention

[0010] The present invention seeks a method and system for expanding multiple speaker playback.

[0011] Accordingly, in one aspect, the present invention provides, a channel synthesis processor for a surround sound decoding unit for processing stereo audio input having a left input signal and a right input signal for playback on a plurality of channels comprising a left channel, a right channel, a center channel, a left surround channel and a right surround channel, the channel synthesis processor comprising;

a band splitting filter bank for receiving, splitting and filtering the left and right input signal into a plurality of frequency sub-bands, each of the plurality of frequency sub-bands comprising a left sub-band signal and a right sub-band signal;

a plurality of single-band synthesis structures corresponding to the plurality of frequency sub-bands, each of the single-band synthesis structure for receiving the corresponding left and right sub-band signals from each of the frequency sub-bands for processing into a plurality of sub-band channels comprising: a left sub-band channel, a right sub-band channel, a center sub-band channel, a left surround sub-band channel, and a right surround sub-band channel;

wherein each corresponding sub-band channels of the plurality of sub-band channels of each single-band synthesis structure are summed up in a mixer for reproduction in a corresponding channel of the plurality of channels of the surround sound decoding unit.

[0012] In another aspect, the present invention provides, a method for processing stereo audio input having a left input signal and a right input signal for playback on a plurality of channels comprising a left channel, a right channel, a center channel and a left surround channel and a right surround channel, the method comprising the steps: splitting the left and right input signals into a plurality of frequency sub-bands, each of the plurality of frequency sub-bands comprising a left sub-band signal and a right sub-band signal; processing the left sub-band signal and the right sub-band signal into a plurality of sub-band channels comprising: a left sub-band channel, a right sub-band channel, a center sub-band channel, a left surround sub-band channel, and a right surround sub-band channel; and summing up each of the corresponding plurality of sub-band channels for reproduction in a corresponding channel of the plurality of channels.

Brief Description of the Drawings

[0013] A preferred embodiment of the present invention will now be more fully described, with reference to the drawings of which:

[0014] FIG.1 illustrates a functional block diagram of a typical 5.1 multi-speaker system;

[0015] FIG.2 illustrates a functional block diagram of a Surround Sound Decoding System;

[0016] FIG.3 illustrates a Surround Sound Decoding unit of FIG.2;

[0017] FIG.4 illustrates a Channel Synthesis Processor of FIG.3;

[0018] FIG.5 illustrates a Single-Band Channel Synthesis Structure of FIG.4;

[0019] FIG.6 illustrates a gain control unit of FIG.5;

[0020] FIG.7A and FIG.7B illustrates a logic flow for calculating G_{LR} , G_C and G_S ;

[0021] FIG.8 illustrates a logic flow for calculating G_{RS} and G_{LS} .

Detailed description of the Drawings

[0022] A method and system for expanding multiple speaker playback with a preferred embodiment is described. In the following description, details are provided to describe the preferred embodiment. It shall be apparent to one skilled in the art, however that the preferred embodiment may be practiced without such details. Some of the details may not be described at length so as not to obscure the preferred embodiment.

[0023] This preferred embodiment addresses the problem of reproducing stereo sounds with up to 5.1 speaker systems. It not only serves an alternative method for the current methods, but also offers a solution for improved channel separation when there are more than one dominant signal. It also addresses the issue of scalability in terms of computation and memory usage.

[0024] The preferred embodiment adopts multi-band processing to allow different frequency bands to be steered separately. This advantageously achieves better channel separation when there are more than one dominant signal at any time. As the number of frequency bands increases, the ambiguity of decoding is reduced as the chances of the dominant band falling into the different frequency bands increases.

[0025] The issues of multiple dominant signals is further decreased by introducing slow decay of surround signals when there is a dominant center signal and vice versa. This is based on the assumption that not all dominant signals are active at all times. For example, a dominant signal at a center channel with dialog will have pauses between words and sentences. If a dominant signal is present in the center channel, surround channels will not be completely muted within a short period of time, rather a slow decay is used, which enables faster recovery during the inactive time of the center channel.

[0026] N-2-N matrixing systems have been claimed to be capable of mixing multi-channel audio into stereo audio, which can be decoded back into the original multi-channel format.

[0027] Considering:

$$Lt = L + 0.707C - 0.707S$$

$$Rt = R + 0.707C + 0.707S$$

where Lt and Rt are the left and right channels after the encoding, L , R , C and S are the left, right, center and surround channels. The phase of the surround channel is normally shifted by 90 degree before encoding.

[0028] The decoding process will take a matrix as

$$\begin{bmatrix} L \\ R \\ C \\ S \end{bmatrix} = \begin{bmatrix} G_{LL} & G_{LR} \\ G_{RL} & G_{RR} \\ G_{CL} & G_{CR} \\ G_{SL} & G_{SR} \end{bmatrix} \times \begin{bmatrix} Lt \\ Rt \end{bmatrix}$$

where G is the gain. On top of that, most of the decoder will have the following decoding method:

$$\begin{aligned} L &= G_L \times Lt \\ R &= G_R \times Rt \\ C &= G_C \times (Rt + Lt) \\ S &= G_S \times (Rt - Lt) \end{aligned}$$

where G_L , G_R , G_C and G_S are the gains for the left, right, center and surround channels respectively.

[0029] The gains are obtained by analysing the in-phase and out-of-phase signals between the two inputs. Mathematically, it is impossible to reproduce the exact original multi-channel audio once it has been mixed into stereo. Rather an approximation of the original audio signals is derived based on certain assumptions made. The most common assumption will be that there is only one dominant audio signal at any given time, which is often not true. If there are actually more than one dominant signal at a particular time, the assumption will lead to incorrect decoding. For example, if there exist a dominant center signal and a dominant surround signal at the same time, most of the decoder will, instead outputting a center signal and a surround signal, will output a left signal and a right signal.

[0030] The preferred embodiment splits the input signals into a number of frequency sub-bands. Each frequency sub-band is then processed by a synthesis structure. This synthesis structure can be reused by all the frequency sub-bands. By summing the outputs from the synthesis structure for all the sub-bands, the multi-channel output is then sent to the speaker system.

[0031] In each of the single band synthesis structure, the left and right outputs are obtained by the left and right inputs with a gain; the center output is obtained by the sum of left and right inputs with a gain; the surround outputs are obtained by the difference of right and left inputs with a gain. Gains are further applied to the two surround outputs to steer the surround information between left surround and right surround. The synthesis structure contains signal detection and feedback mechanisms to control all these gains. The gain of each frequency sub-band for each output is regulated through gradual increment and decrement to avoid rapid fluctuations.

[0032] Referring to FIG.1, a functional block diagram of a typical 5.1 multi-speaker system 10 is shown. The audio video source 30 is split into its audio content 32 and video content 33 and sent to their respective audio 35 and video 36 decoders. The output from the video decoder 36 is then sent to a display unit 20 for providing the visual images. The output from the audio decoder 35 which comprises the left input signal 5 and right input signal 6 from the decoded stereo audio content is then sent to the Surround Sound Decoding System (SSDS) 50.

[0033] The SSDS 50 then processes the left and right input signals and produce a plurality of channels each for playing on a respective speaker in the 5.1 multi-speaker system. It is the intention that listener 3 sitting at an optimum location will be able to enjoy the stereo audio content played through such a 5.1 multi-speaker system.

[0034] The plurality of channels that would be played by the respectively speakers are: left channel 11, right channel 12, center channel 13, left surround channel 14, right surround channel 15 and Low Frequency Effects (LFE) channel 16 (subwoofer).

[0035] Referring to FIG.2, the SSDS 50 further comprises a Surround Sound Decoding Unit (SSDU) 52 that takes as input the left 5 and right 6 input signals and produces the plurality of channels: the left channel 11, the right channel 12, the center channel 13, the left surround channel 14, the right surround channel 15 and the low frequency effect channel 16

[0036] Referring to FIG.3, the SSDU 52 comprises a Channel Synthesis Processor 60 which receives the left input signal 5 and right input signal 6 via a pair of High Pass Filters 63. The left input signal 5 and right input signal 6 are also summed up in a mixer before filtering in a Low Pass filter 65 to produce the Low Frequency Effects. The cutoff frequency can vary from 50 Hz to 200 Hz depending on the speaker and subwoofer characteristics. A attenuation of 3dB may be performed to normalize the loudness of the LFE channel 16.

[0037] Referring to FIG.4, the Channel Synthesis Processor 60 comprises of a Band-Splitting Filter Bank 70 which receives the left input signal 5 and right input signal 6 for splitting and filtering into a plurality of frequency sub-bands. Each of the frequency sub-bands comprises a left sub-band signals 5a and a right sub-band signals 6a.

[0038] The number of frequency sub-bands can vary depending on the processing or computational power allocated or available. The more frequency sub-bands the system has, the higher the quality of the reproduction.

[0039] The left sub-band signals 5a and the right sub-band signals 6a of each sub-band is processed by a single band synthesis structure 72 to produce a plurality of sub-band channel signals. The sub-band channel signals are: left sub-band channel 11a, right sub-band channel 12a, center sub-band channel 13a, left surround sub-band channel 14a, and right surround sub-band channel 15a. The sum of all corresponding sub-band channel signals from each single band synthesis structure 72 is sent to the respective output channels to be reproduced by the respective speakers.

[0040] Each of the single-band synthesis structures 72 are controlled by predetermined control parameters. This splitting up into a plurality of frequency sub-bands and having each frequency sub-band further processed for a plurality of sub-band channel signals can be referred to as Band based steering. The preferred embodiment utilizes Band based steering to achieve better channel separation and also allows for ease of scalability with limitations of computational power by scaling down on the number of frequency sub-bands.

[0041] Referring to FIG.5, the single band synthesis structure 72 takes the left sub-band signals 5a and the right sub-band signals 6a of each sub-band, processes them and produces the sub-band channel signals which are: the left sub-band channel 11a, the right sub-band channel 12a, the center sub-band channel 13a, the left surround sub-band channel 14a, and the right surround sub-band channel 15a.

[0042] The sub-band channel signals are generated by Voltage controlled amplifiers 83 amplifying the left sub-band signal 5a and right sub-band signal with a plurality of different predetermined gains from a gain control unit.

[0043] The left sub-band channel 11a is generated using the left sub-band signal 5a and left right gain G_{LR} . The right sub-band channel 12a is generated using the right sub-band signal 6a and the left right gain G_{LR} .

[0044] The center sub-band channel 13a is generated using the sum of the left sub-band signal 5a and the right sub-band signal 6a with center gain G_C . The resulting signal can further be attenuated by 3 dB before being consolidated by the mixers.

[0045] The left surround sub-band channel 14a is generated using the sum of the left sub-band signal 5a and the right sub-band signal 6a amplified with surround gain G_S and then further amplified using right surround gain G_{RS} . The signal may also be first attenuated by 3 dB before being amplified by the right surround gain G_{RS} . The right surround sub-band channel 15a is generated using the sum of the left sub-band signal 5a and the right sub-band signal 6a amplified with surround gain G_S and then further amplified using left surround gain G_{LS} . The signal may also be first attenuated by 3 dB before being amplified by the left surround gain G_{LS} .

[0046] The 3 dB attenuation of some of the signals are to normalize the amplitudes of the signals as a result of summing the left sub-band signal 5a and right sub-band signal.

[0047] The gains from the gain control unit 80 are generated by calculations performed on the left sub-band signal 5a and right sub-band signal. Referring to FIG.6, the gains are generated by a Threshold Detection and Gain Adjustment Calculation (TDGC) unit 85.

[0048] For calculating the G_{LS} and G_{RS} , the left sub-band signal 5a and right sub-band signal are buffered using an initial buffer having a predetermined algorithm. The signals are further buffered using buffer lengths varying typically between 5ms

to 0.1s. The buffered signals are then further windowed using any of predetermined windows. Rectangular windows may be used for low computational power uses or more complicated windows such as hamming windows may be applied if computational power is not an issue. The buffered and windowed signals are then accumulated and sent to the TDGC unit 85. The signals sent to the TDGC unit 85 are the left accumulation value M_L 5b and the right accumulation value M_R 6b.

[0049] The left accumulation value M_L 5b and the right accumulation value M_R 6b undergo a threshold detection and gain adjustment calculation process in the TDGC unit 85 to generate the gains G_{LS} and G_{RS} . These gains G_{LS} and G_{RS} are further generated by using a first set of control parameters 91 in control units. The first set of control parameters 91 comprising at least a increment and/or decrement size D .

[0050] The initial buffer used for buffering the left sub-band signal 5a may further be represented by $|L + \alpha R|$ for the left accumulation value M_L 5b. While, the initial buffer used for buffering the right sub-band signal 6a and may be represented by $|R + \alpha L|$ for the right accumulation value M_R 6b. Where α is a control parameter varying from 0 to 0.4.

[0051] For calculating the G_{LR} , G_C and G_S , the left sub-band signal 5a and right sub-band signal 6a are first used to generate a Total Correlation value A_T 17 and a Differential Correlation Value A_D 18. The total correlation value A_T 17 and the differential correlation value A_T 18 are then sent to the TDGC unit 85 where the threshold detection and gain adjustment calculation process is used to calculate G_{LR} , G_C and G_S . These together with a second set of control parameters 92 are used in control units to the gains G_{LR} , G_C and G_S .

[0052] The left sub-band signal 5a and right sub-band signal 6a are first summed up before undergoing buffering of buffer lengths varying typically between 5ms to 0.1s. The buffered signals are then further windowed using any of predetermined windows. Rectangular windows may be used for low computational power uses or more complicated hamming windows may be applied if computational power is not an issue. The summed, buffered and windowed signals are then autocorrelated to generate the Total Correlation value A_T 17.

[0053] For generating the Differential Correlation Value A_D 18, the left sub-band signal 5a is subtracted from the right sub-band signal 6a, then buffered and windowed in a similar manner before undergoing auto correlation.

[0054] Referring to FIG.7A and FIG.7B, the detailed calculations for calculating the gains G_{LR} , G_C and G_S is shown. The gain value is gradually increased to a maximum when there is presence of a strong dominant signal, whereas if there is no clear dominant signal, ideal gain values are calculated and the gain values are gradually increased towards the ideal gain values. Slower decay is introduced for surround channel when there is a strong center signal and vice versa.

[0055] The second set of control parameters 92 comprises of:

I_C	:	value of increment for the center channel
D_C	:	value of decrement for the center channel
D_{SC}	:	value of decrement for the center channel with slow decay, which can be 2-5 times smaller than D_C
U_C	:	upper threshold value for center channel
L_C	:	lower threshold value for center channel
S_C	:	difference between the upper and lower threshold value for center channel
I_S	:	value of increment for the surround channel
D_S	:	value of decrement for the surround channel
		D_{SS} :the value of decrement for the surround channel with slow decay, which can be 2-5 times smaller than D_S
U_S	:	upper threshold value for surround channel
L_S	:	lower threshold value for surround channel

S_S : difference between the upper and lower threshold value for surround channel
 I_{LR} : value of increment for the left and right channel
 D_{LR} : value of decrement for the left and right channel

[0056] The logic flow in FIG.7A and FIG.7B determines the next gain value by taking the current gain values and adjusting them after analyzing the output from the auto-correlators: A_T 17 and A_D 18.

[0057] Referring to FIG.7A and FIG.7B, the logic flow of the calculations are as shown:

If $A_T/A_D > U_C$ If $G_C < 1$ $G_C = G_C + I_C$ End If $G_{LR} > 0$ $G_{LR} = G_{LR} - D_{LR}$ End If $G_S > 0$ $G_S = G_S - D_{SS}$ End	Condition 1
Else if $L_C < A_T/A_D \leq U_C$ If $A_T/A_D > U_C - G_{LR} * S_C$ If $G_{LR} > 0$ $G_{LR} = G_{LR} - (1.207 - G_{LR}) * D_{LR}$ End	Condition 2 Condition 2a
$G_{LR} = G_{LR} - (1.207 - G_{LR}) * D_{LR}$	Eqn. 2.1
Else if $A_T/A_D < U_C - G_{LR} * S_C$ If $G_{LR} < 1$ $G_{LR} = G_{LR} + (1.207 - G_{LR}) * I_{LR}$ End	Condition 2b
$G_{LR} = G_{LR} + (1.207 - G_{LR}) * I_{LR}$	Eqn. 2.2
End If $A_T/A_D > U_C - G_C * S_C$ If $G_C > 0$ $G_C = G_C - (1.207 - G_C) * D_C$ End	Condition 2c
$G_C = G_C - (1.207 - G_C) * D_C$	Eqn. 2.3
Else if $A_T/A_D < U_C - G_C * S_C$ If $G_C < 1$ $G_C = G_C + (1.207 - G_C) * I_C$ End	Condition 2d
$G_C = G_C + (1.207 - G_C) * I_C$	Eqn. 2.4
End If $G_S > 0$	

$G_S = G_S - D_{SS}$	
End	
Else if $A_D/A_T > U_S$	Condition 3
If $G_S < 1$	
$G_S = G_S + I_S$	
End	
If $G_{LR} > 0$	
$G_{LR} = G_{LR} - D_{LR}$	
End	
If $G_C > 0$	
$G_C = G_C - D_{SC}$	
End	
Else if $L_S < A_D/A_T \leq U_S$	Condition 4
If $A_D/A_T > U_S - G_{LR} * S_S$	Condition 4a
If $G_{LR} > 0$	
$G_{LR} = G_{LR} - (1.207 - G_{LR}) * D_{LR}$	Eqn. 4.1
End	
Else if $A_D/A_T < U_S - G_{LR} * S_S$	Condition 4b
If $G_{LR} < 1$	
$G_{LR} = G_{LR} + (1.207 - G_{LR}) * I_{LR}$	Eqn. 4.2
End	
End	
If $A_D/A_T > U_S - G_S * S_S$	Condition 4c
If $G_S > 0$	
$G_S = G_S - (1.207 - G_S) * D_S$	Eqn. 4.3
end	
Else if $A_D/A_T < U_S - G_S * S_S$	Condition 4d
If $G_S < 1$	
$G_S = G_S + (1.207 - G_S) * D_S$	Eqn. 4.4
End	
End	
If $G_C > 0$	
$G_C = G_C - D_{SC}$	
End	
Else	
If $G_{LR} < 1$	
$G_{LR} = G_{LR} + I_{LR}$	
End	
If $G_C > 0$	
$G_C = G_C - D_C$	
End	
If $G_S > 0$	
$G_S = G_S - D_S$	
End	

End

[0058] Summarizing the logic flow for calculating the gains G_{LR} , G_C and G_S , when one of these sub-band channels are dominant, the gain for that sub-band channel is incremented while the other two gains are decremented.

[0059] When Correlation factor A_T/A_D is greater than U_C (**Condition 1**), the center sub-band channel 13a is dominant and G_C is incremented while G_{LR} and G_S are decremented.

[0060] When value of the Correlation factor A_T/A_D is between L_C and U_C (**Condition 2**), the left sub-band channel 11a, the right sub-band channel 12a, and the center sub-band channel 13a are exhibiting dominant characteristics. The G_{LR} is then compared with an ideal G_{LR} value (**Condition 2a, 2b**) and suitably decremented (**Eqn. 2.1**) or incremented (**Eqn. 2.2**) towards that ideal value. The G_C is also compared with an ideal G_C value (**Condition 2c, 2d**) and suitably decremented (**Eqn. 2.3**) or incremented (**Eqn. 2.4**) towards that ideal value. G_S is also then decremented.

[0061] When Correlation factor A_T/A_D is less than U_S (**Condition 3**), the left surround sub-band channel 14a and the right surround sub-band channel 15a are dominant and G_S is incremented while G_{LR} and G_C are decremented.

[0062] When value of the Correlation factor A_T/A_D is between L_S and U_S (**Condition 4**), the left sub-band channel 11a, the right sub-band channel 12a, the left surround sub-band channel 14a and the right surround sub-band channel 15a are exhibiting dominant characteristics. The G_{LR} is then compared with an ideal G_{LR} value (**Condition 4a, 4b**) and suitably decremented (**Eqn. 4.1**) or incremented (**Eqn. 4.2**) towards that ideal value. The G_S is also compared with an ideal G_S value

(**Condition 4c, 4d**) and suitably decremented (**Eqn. 4.3**) or incremented (**Eqn. 4.4**) towards that ideal value. G_C is also then decremented.

[0063] When the left sub-band channel 11a and the right sub-band channel 12a are dominant, then G_{LR} is incremented while G_C and G_S are decremented.

[0064] As for calculation of G_{LS} and G_{RS} , the left accumulation value M_L 5b and the right accumulation value M_R 6b are used in conjunction with the first set of control parameters 91 which comprises of the predetermined increment and/or decrement value of D.

[0065] Essentially, each of the gains are assigned with predetermined increment and decrement step sizes, the corresponding gain being incremented till a maximum value and other gains being decremented till a minimum values when a single dominant signal is detected at an input channel. The gains of each output being incremented or decremented towards ideal values calculated when more than single dominant channels are present. Furthermore, the gain of the surround channel being decremented at a smaller step when a dominant center signal is detected and vice versa.

[0066] Referring to FIG.8, the logic flow of the calculations for G_{LS} and G_{RS} are as shown:

If $M_L * G_{RS} < M_R * G_{LS}$	Condition 5
If $G_{RS} < 0.707$	
$G_{RS} = G_{RS} + (0.853 - G_{RS}) * D$	Eqn. 5.1
End	
If $G_{LS} > 0$	
$G_{LS} = G_{LS} - (0.853 - G_{LS}) * D$	Eqn. 5.2
End	
Else if $M_L * G_{RS} > M_R * G_{LS}$	Condition 6
If $G_{LS} < 0.707$	

$$G_{LS} = G_{LS} + (0.853 - G_{LS}) * D \quad \text{Eqn. 6.1}$$

End

If $G_{RS} > 0$

$$G_{RS} = G_{RS} - (0.853 - G_{RS}) * D \quad \text{Eqn. 6.2}$$

End

End

[0067] Summarizing the logic flow for calculating the gains G_{LS} and G_{RS} , Accumulation factor M_R/M_L is compared with a ratio of G_{RS} and G_{LS} . When Accumulation factor M_R/M_L is greater than the ratio of G_{RS} and G_{LS} (**Condition 5**), G_{RS} is incremented (**Eqn. 5.1**) while G_{LS} is decremented (**Eqn. 5.2**).

[0068] When Accumulation factor M_R/M_L is less than the ratio of G_{RS} and G_{LS} (**Condition 6**), G_{RS} is decremented (**Eqn. 6.1**) while G_{LS} is incremented (**Eqn. 6.2**).

[0069] The present invention provides a method of reproducing stereo audio on up to 5 speakers and a sub-woofer (5.1 speaker system). Channel separation is enhanced by processing sub-bands separately. Any sudden volume changes in the output channels are tampered by using gradual increments and decrements in the gain values of each channel. Transitions from one channel to another is also made smooth with slow decays of gain values of each channel.

[0070] The method and structure is fully scalable to meet the computational constraints of hardware used. Further more, since all the sub-bands processing share the same structure, it saves the hardware size or code size depending on the method of realizing the algorithm. Further more, the computational power required is relatively small compared with the present state of the art that employs complex logarithmic computation using DSPs.

[0071] It will be appreciated that various modifications and improvements can be made by a person skilled in the art without departing from the scope of the present invention.

Claims

1. A channel synthesis processor for a surround sound decoding unit for processing stereo audio input having a left input signal and a right input signal for playback on a plurality of channels comprising a left channel, a right channel, a center channel, a left surround channel and a right surround channel, the channel synthesis processor comprising;
 - a band splitting filter bank for receiving, splitting and filtering the left and right input signal into a plurality of frequency sub-bands, each of the plurality of frequency sub-bands comprising a left sub-band signal and a right sub-band signal;
 - a plurality of single-band synthesis structures corresponding to the plurality of frequency sub-bands, each of the single-band synthesis structure for receiving the corresponding left and right sub-band signals from each of the frequency sub-bands for processing into a plurality of sub-band channels comprising: a left sub-band channel, a right sub-band channel, a center sub-band channel, a left surround sub-band channel, and a right surround sub-band channel;
 - wherein each corresponding sub-band channels of the plurality of sub-band channels of each single-band synthesis structure are summed up in a mixer for reproduction in a corresponding channel of the plurality of channels of the surround sound decoding unit.
2. The channel synthesis processor in accordance with claim 1, wherein the left and right input signal are filtered by a high pass filter before processing.
3. The channel synthesis processor in accordance with claim 1, wherein the left and right input signal are summed and filtered by a low pass filter to produce a low frequency effects channel.

4. The channel synthesis processor in accordance with claim 1, wherein the single-band synthesis structure comprises:
 - a gain control unit for determining a plurality of gain values comprising: G_{LR} , G_C , G_S , G_{RS} and G_{LS} ;
 - a plurality of amplifiers for amplifying respectively the left sub-band signal and right sub-band signal with G_{LR} to produce the left sub-band channel and right sub-band channel respectively;
 - where,
 - G_{LR} is left right gain; G_C is center gain; G_S is surround gain; G_{RS} is right surround gain; and G_{LS} is left surround gain.
5. The channel synthesis processor in accordance with claim 4, wherein the center sub-band channel is produced by summing the left sub-band signal and right sub-band signal before amplifying with G_C .
6. The channel synthesis processor in accordance with claim 4, wherein the left surround sub-band channel is produced by the difference between the left sub-band signal and right sub-band signal being amplified by G_S and then G_{RS} .
7. The channel synthesis processor in accordance with claim 4, wherein the right surround sub-band channel is produced by the difference between the left sub-band signal and right sub-band signal being amplified by G_S and then G_{LS} .
8. The channel synthesis processor in accordance with claim 4, wherein the gain control unit comprises:
 - a threshold detection and gain adjustment calculation (TDGC) unit for calculating the plurality of gain values G_{LR} , G_C , G_S , G_{RS} and G_{LS} ;
 - means for buffering the left and right sub-band signals;

means for windowing the left and right sub-band signals after buffering;

means for accumulating the left and right sub-band signals after windowing for producing a left accumulation value M_L and a right accumulation value M_R ; and

means for auto-correlating the left and right sub-band signals after buffering for producing a total correlation value A_T and a differential correlation Value A_D .

9. The channel synthesis processor in accordance with claim 8, wherein the left sub-band signals first undergoes initial buffering of $\lceil L + \alpha R \rceil$ for producing the left accumulation value M_L ; and the right sub-band signals first undergoes initial buffering of $\lceil R + \alpha L \rceil$ for producing the left accumulation value M_L ;

where,

L is the left sub-band signal;

R is the right sub-band signal;

α is a control parameter varying from 0 to 0.4.

10. The channel synthesis processor in accordance with claim 8, wherein the left and right sub-band signals are summed before producing the total correlation value A_T ; and the left and right sub-band signals are subtracted before producing the differential correlation Value A_D .
11. The channel synthesis processor in accordance with claim 8, wherein a first set of control parameters is used for producing G_{RS} and G_{LS} .
12. The channel synthesis processor in accordance with claim 8, wherein a second set of control parameters is used for producing G_{LR} , G_C and G_S .

13. A method for processing stereo audio input having a left input signal and a right input signal for playback on a plurality of channels comprising a left channel, a right channel, a center channel and a left surround channel and a right surround channel, the method comprising the steps:
 - a. splitting the left and right input signals into a plurality of frequency sub-bands, each of the plurality of frequency sub-bands comprising a left sub-band signal and a right sub-band signal;
 - b. processing the left sub-band signal and the right sub-band signal into a plurality of sub-band channels comprising: a left sub-band channel, a right sub-band channel, a center sub-band channel, a left surround sub-band channel, and a right surround sub-band channel; and
 - c. summing up each of the corresponding plurality of sub-band channels for reproduction in a corresponding channel of the plurality of channels.
14. The method in accordance with claim 13, wherein the left and right input signals are filtered by a high pass filter before processing.
15. The method in accordance with claim 13, wherein the left and right input signal are summed and filtered by a low pass filter to produce a low frequency effects channel.
16. The method in accordance with claim 13, wherein step b. comprises:
 - b1. amplifying respectively the left sub-band signal and right sub-band signal with left right gain G_{LR} to produce the left sub-band channel and right sub-band channel.
17. The method in accordance with claim 13, wherein step b. comprises:

- b2. amplifying sum of the left sub-band signal and right sub-band signal with center gain G_C . to produce the center sub-band channel.
- 18. The method in accordance with claim 13, wherein step b. comprises:
 - b3. amplifying difference between the left sub-band signal and right sub-band signal by surround gain G_S and right surround G_{RS} to produce the left surround sub-band channel.
- 19. The method in accordance with claim 13, wherein step b. comprises:
 - b4. amplifying difference between the left sub-band signal and right sub-band signal by surround gain G_S and left surround G_{LS} to produce the right surround sub-band channel.
- 20. The method in accordance with claim 13, wherein the left and right sub-band signals are buffered, windowed and accumulated for respectively producing a left accumulation value M_L and a right accumulation value M_R ; and the left and right sub-band signals are buffered, windowed and auto-correlated for respectively producing a total correlation value A_T and a differential correlation Value A_D .
- 21. The method in accordance with claim 20, wherein the left sub-band signal first undergo an initial buffering of $\left| \frac{L}{2} + \alpha R \right|$ for producing the left accumulation value M_L ; and the right sub-band signal first undergo an initial buffering of $\left| \frac{R}{2} + \alpha L \right|$ for producing the left accumulation value M_L ;
 - where,
 - L is the left sub-band signal;
 - R is the right sub-band signal;

α is a control parameter varying from 0 to 0.4.

22. The method in accordance with claim 20, wherein the left and right sub-band signals are summed before producing the total correlation value A_T ; and the left and right sub-band signals are subtracted before producing the differential correlation Value A_D .
23. The method in accordance with claim 22, wherein one of the gains G_{LR} , G_C and G_S , is incremented when the sub-band channel corresponding to one of the gains G_{LR} , G_C and G_S is dominant while the other two gains are decremented.
24. The method in accordance with claim 23, wherein G_C is incremented when Correlation factor A_T/A_D is greater than an upper threshold value for the center sub-band channel.
25. The method in accordance with claim 23, wherein G_S is incremented when Correlation factor A_T/A_D is less than an upper threshold value for the surround sub-band channel.
26. The method in accordance with claim 23, wherein the gains G_{LR} , G_C and G_S are suitably decremented or incremented towards an ideal gain value of G_{LR} , G_C and G_S when more than one of the sub-band channels corresponding to the gains G_{LR} , G_C and G_S is dominant.
27. The method in accordance with claim 21, wherein Accumulation factor M_R/M_L is compared with a ratio of G_{RS} and G_{LS} , and G_{RS} is incremented while G_{LS} is decremented when the Accumulation factor M_R/M_L is greater than the ratio of G_{RS} and G_{LS} .

28. The method in accordance with claim 21, wherein G_{RS} is decremented while G_{LS} is incremented when the Accumulation factor M_R/M_L is less than the ratio of G_{RS} and G_{LS} .