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(54) BINAURAL RENDERING OF A MULTI-CHANNEL AUDIO SIGNAL

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RENDU BINAURAL DE SIGNAL AUDIO MULTICANAUX

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(56) References cited:
WO-A-2007/078254 WO-A-2007/083952
WO-A-2008/069593

- **ENGDEGORD J ET AL: "Spatial Audio Object Coding (SAOC) - The Upcoming MPEG Standard on Parametric Object Based Audio Coding" 124TH AES CONVENTION, AUDIO ENGINEERING SOCIETY, PAPER 7377,, 17 May 2008 (2008-05-17), pages 1-15, XP002541458**

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Description

[0001] The present application relates to binaural rendering of a multi-channel audio signal.

[0002] Many audio encoding algorithms have been proposed in order to effectively encode or compress audio data of one channel, i.e., mono audio signals. Using psychoacoustics, audio samples are appropriately scaled, quantized or even set to zero in order to remove irrelevancy from, for example, the PCM coded audio signal. Redundancy removal is also performed.

[0003] As a further step, the similarity between the left and right channel of stereo audio signals has been exploited in order to effectively encode/compress stereo audio signals.

[0004] However, upcoming applications pose further demands on audio coding algorithms. For example, in teleconferencing, computer games, music performance and the like, several audio signals which are partially or even completely uncorrelated have to be transmitted in parallel. In order to keep the necessary bit rate for encoding these audio signals low enough in order to be compatible to low-bit rate transmission applications, recently, audio codecs have been proposed which downmix the multiple input audio signals into a downmix signal, such as a stereo or even mono downmix signal. For example, the MPEG Surround standard downmixes the input channels into the downmix signal in a manner prescribed by the standard. The downmixing is performed by use of so-called OTT^{-1} and TTT^{-1} boxes for downmixing two signals into one and three signals into two, respectively. In order to downmix more than three signals, a hierarchic structure of these boxes is used. Each OTT^{-1} box outputs, besides the mono downmix signal, channel level differences between the two input channels, as well as inter-channel coherence/cross-correlation parameters representing the coherence or cross-correlation between the two input channels. The parameters are output along with the downmix signal of the MPEG Surround coder within the MPEG Surround data stream. Similarly, each TTT^{-1} box transmits channel prediction coefficients enabling recovering the three input channels from the resulting stereo downmix signal. The channel prediction coefficients are also transmitted as side information within the MPEG Surround data stream. The MPEG Surround decoder upmixes the downmix signal by use of the transmitted side information and recovers, the original channels input into the MPEG Surround encoder.

[0005] However, MPEG Surround, unfortunately, does not fulfill all requirements posed by many applications. For example, the MPEG Surround decoder is dedicated for upmixing the downmix signal of the MPEG Surround encoder such that the input channels of the MPEG Surround encoder are recovered as they are. In other words, the MPEG Surround data stream is dedicated to be played back by use of the loudspeaker configuration having been used for encoding, or by typical configurations like stereo.

[0006] However, according to some applications, it would be favorable if the loudspeaker configuration could be changed at the decoder's side freely.

[0007] In order to address the latter needs, the spatial audio object coding (SAOC) standard is currently designed. Each channel is treated as an individual object, and all objects are downmixed into a downmix signal. That is, the objects are handled as audio signals being independent from each other without adhering to any specific loudspeaker configuration but with the ability to place the (virtual) loudspeakers at the decoder's side arbitrarily. The individual objects may comprise individual sound sources as e.g. instruments or vocal tracks. Differing from the MPEG Surround decoder, the SAOC decoder is free to individually upmix the downmix signal to replay the individual objects onto any loudspeaker configuration. In order to enable the SAOC decoder to recover the individual objects having been encoded into the SAOC data stream, object level differences and, for objects forming together a stereo (or multi-channel) signal, inter-object cross correlation parameters are transmitted as side information within the SAOC bitstream. Besides this, the SAOC decoder/transcoder is provided with information revealing how the individual objects have been downmixed into the downmix signal. Thus, on the decoder's side, it is possible to recover the individual SAOC channels and to render these signals onto any loudspeaker configuration by utilizing user-controlled rendering information.

[0008] However, although the afore-mentioned codecs, i.e. MPEG Surround and SAOC, are able to transmit and render multi-channel audio content onto loudspeaker configurations having more than two speakers, the increasing interest in headphones as audio reproduction system necessitates that these codecs are also able to render the audio content onto headphones. In contrast to loudspeaker playback, stereo audio content reproduced over headphones is perceived inside the head. The absence of the effect of the acoustical pathway from sources at certain physical positions to the eardrums causes the spatial image to sound unnatural since the cues that determine the perceived azimuth, elevation and distance of a sound source are essentially missing or very inaccurate. Thus, to resolve the unnatural sound stage caused by inaccurate or absent sound source localization cues on headphones, various techniques have been proposed to simulate a virtual loudspeaker setup. The idea is to superimpose sound source localization cues onto each loudspeaker signal. This is achieved by filtering audio signals with so-called head-related transfer functions (HRTFs) or binaural room impulse responses (BRIRs) if room acoustic properties are included in these measurement data. However, filtering each loudspeaker signal with the just-mentioned functions would necessitate a significantly higher amount of computation power at the decoder/reproduction side. In particular, rendering the multi-channel audio signal onto the "virtual" loudspeaker locations would have to be performed first wherein, then, each loudspeaker signal thus obtained is filtered with the respective transfer function or impulse response to obtain the left and right channel of the binaural output signal. Even worse: the thus obtained binaural output signal would have a poor audio quality due to the fact that in order to achieve the virtual loudspeaker signals, a relatively large amount of synthetic decorrelation signals would have to be mixed into the upmixed signals in order to compensate for the correlation between originally uncorrelated audio input signals, the correlation resulting from downmixing the plurality of audio input signals into the downmix signal.

[0009] In the current version of the SAOC codec, the SAOC parameters within the side information allow the user-interactive spatial rendering of the audio objects using any playback setup with, in principle, including headphones. Binaural rendering to headphones allows spatial control of virtual object positions in 3D space using head-related transfer function (HRTF) parameters. For example, binaural rendering in SAOC could be realized by restricting this case to the mono downmix SAOC case where the input signals are mixed into the mono channel equally. Unfortunately, mono downmix necessitates all audio signals to be mixed into one common mono downmix signal so that the original correlation properties between the original audio signals are maximally lost and therefore, the rendering quality of the binaural rendering output signal is non-optimal.

[0010] In Engdegard et al.: Spatial Audio Object Coding (SAOC) - The Upcoming MPEG Standard on Parametric Object Based Coding, 124th AES convention, paper 7377, May 17, 2008, pages 1-15, SAOC is described as an extension of the MPEG surround standard. In particular, the chosen reference model architecture, the association between the different operational modes and applications, and the status of the standardization process of 2008 is described.

[0011] WO 2008/069593 A1 describes a method for processing an audio signal comprising receiving a downmix signal, a first multi-channel information and an object information, processing the downmix signal using the object information and a mix information, and transmitting one of the first multi-channel information and a second multi-channel information according to the mix information, wherein the second channel information is generated using the object information and the mix information.

[0012] WO 2007/078254 A2 describes a personalized decoding of multi-channel surround sound. A parametric multi-channel surround audio bitstream is received in a multi-channel decoder. The received spatial parameters are transformed into a new set of spatial parameters which are used in order to obtain a decoding of the multi-channel surround sound that is not a simple equivalent of the original input multi-channel surround signal but e.g. may be personalized by making the transformation based on the representation of user head related filters. Such personalized spatial parameters may be obtained by combining the received spatial parameters and the representation of the user head related filter with a set of additional rendering parameters that, for example, are interactively determined by the user and thus, are time dependent.

[0013] WO 2007/083952 A1 describes a method and an apparatus for processing a media signal, by which the media signal can be converted to a surround signal by using spatial information of the media signal. Source mapping information corresponding to each source of multi-sources is generated by using spatial information indicating features between the multi-sources. At least one rendering information is generated by using the source mapping information and filter information having a surround effect. Smoothing is performed by using neighbor rendering information of the at least one rendering information.

[0014] Thus, it is the object of the present invention to provide a scheme for binaural rendering a multi-channel audio signal such that the binaural rendering result is improved with, concurrently, avoiding a restriction in the freedom of composing the downmix signal from the original audio signals.

[0015] This object is achieved by an apparatus according to claim 1 and a method according to claim 10.

[0016] One of the basic ideas underlying the present invention is that starting binaural rendering of a multi-channel audio signal from a stereo downmix signal is advantageous over starting binaural rendering of the multi-channel audio signal from a mono downmix signal thereof in that, due to the fact that few objects are present in the individual channels of the stereo downmix signal, the amount of decorrelation between the individual audio signals is better preserved, and in that the possibility to choose between the two channels of the stereo downmix signal at the encoder side enables that the correlation properties between audio signals in different downmix channels is partially preserved. In other words, due to the encoder downmix, the inter-object coherences are degraded which has to be accounted for at the decoding side where the inter-channel coherence of the binaural output signal is an important measure for the perception of virtual sound source width, but using stereo downmix instead of mono downmix reduces the amount of degrading so that the restoration/generation of the proper amount of inter-channel coherence by binaural rendering the stereo downmix signal achieves better quality.

[0017] A further main idea of the present application is that the afore-mentioned ICC (ICC = inter-channel coherence) control may be achieved by means of a decorrelated signal forming a perceptual equivalent to a mono downmix of the downmix channels of the stereo downmix signal with, however, being decorrelated to the mono downmix. Thus, while the use of a stereo downmix signal instead of a mono downmix signal preserves some of the correlation properties of the plurality of audio signals, which would have been lost when using a mono downmix signal, the binaural rendering may be based on a decorrelated signal being representative for both, the first and the second downmix channel, thereby reducing the number of decorrelations or synthetic signal processing compared to separately decorrelating each stereo downmix channel.

[0018] Referring to the figures, preferred embodiments of the present application are described in more detail. Among these figures,

Fig. 1 shows a block diagram of an SAOC encoder/decoder arrangement in which the embodiments of the present invention may be implemented;

Fig. 2 shows a schematic and illustrative diagram of a spectral representation of a mono audio signal;

Fig. 3 shows a block diagram of an audio decoder capable of binaural rendering according to an embodiment of the present invention;

Fig. 4 shows a block diagram of the downmix pre-processing block of Fig. 3 according to an embodiment of the present invention;

Fig. 5 shows a flow-chart of steps performed by SAOC parameter processing unit 42 of Fig. 3 according to a first alternative; and

Fig. 6 shows a graph illustrating the listening test results.

[0019] Before embodiments of the present invention are described in more detail below, the SAOC codec and the SAOC parameters transmitted in an SAOC bit stream are presented in order to ease the understanding of the specific embodiments outlined in further detail below.

[0020] Fig. 1 shows a general arrangement of an SAOC encoder 10 and an SAOC decoder 12. The SAOC encoder 10 receives as an input N objects, i.e., audio signals 14_1 to 14_N. In particular, the encoder 10 comprises a downmixer 16 which receives the audio signals 14_1 to 14_N and downmixes same to a downmix signal 18. In Fig. 1, the downmix signal is exemplarily shown as a stereo downmix signal. However, the encoder 10 and decoder 12 may be able to operate in a mono mode as well in which case the downmix signal would be a mono downmix signal. The following description, however, concentrates on the stereo downmix case. The channels of the stereo downmix signal 18 are denoted LO and RO.

[0021] In order to enable the SAOC decoder 12 to recover the individual objects 14_1 to 14_N, downmixer 16 provides the SAOC decoder 12 with side information including SAOC-parameters including object level differences (OLD), inter-object cross correlation parameters (IOC), downmix gains values (DMG) and downmix channel level differences (DCLD). The side information 20 including the SAOC-parameters, along with the downmix signal 18, forms the SAOC output data stream 21 received by the SAOC decoder 12.

[0022] The SAOC decoder 12 comprises an upmixing 22 which receives the downmix signal 18 as well as the side information 20 in order to recover and render the audio signals 14_1 and 14_N onto any user-selected set of channels 24_1 to $24_{M'}$, with the rendering being prescribed by rendering information 26 input into SAOC decoder 12 as well as HRTF parameters 27 the meaning of which is described in more detail below. The following description concentrates on binaural rendering, where M'=2 and, the output signal is especially dedicated for headphones reproduction, although decoding 12 may be able to render onto other (non-binaural) loudspeaker configuration as well, depending on commands within the user input 26.

[0023] The audio signals 14_1 to 14_N may be input into the downmixer 16 in any coding domain, such as, for example, in time or spectral domain. In case, the audio signals 14_1 to 14_N are fed into the downmixer 16 in the time domain, such as PCM coded, downmixer 16 uses a filter bank, such as a hybrid QMF bank, e.g., a bank of complex exponentially modulated filters with a Nyquist filter extension for the lowest frequency bands to increase the frequency resolution therein, in order to transfer the signals into spectral domain in which the audio signals are represented in several subbands associated with different spectral portions, at a specific filter bank resolution. If the audio signals 14_1 to 14_N are already in the representation expected by downmixer 16, same does not have to perform the spectral decomposition.

[0024] Fig. 2 shows an audio signal in the just-mentioned spectral domain. As can be seen, the audio signal is represented as a plurality of subband signals. Each subband signal 30_1 to 30_P consists of a sequence of subband values indicated by the small boxes 32. As can be seen, the subband values 32 of the subband signals 30_1 to 30_P are synchronized to each other in time so that for each of consecutive filter bank time slots 34, each subband 30_1 to 30_P comprises exact one subband value 32. As illustrated by the frequency axis 35, the subband signals 30_1 to 30_P are associated with different frequency regions, and as illustrated by the time axis 37, the filter bank time slots 34 are consecutively arranged in time.

[0025] As outlined above, downmixer 16 computes SAOC-parameters from the input audio signals 14_1 to 14_N. Downmixer 16 performs this computation in a time/frequency resolution which may be decreased relative to the original time/frequency resolution as determined by the filter bank time slots 34 and subband decomposition, by a certain amount, wherein this certain amount may be signaled to the decoder side within the side information 20 by respective syntax elements bsFrameLength and bsFreqRes. For example, groups of consecutive filter bank time slots 34 may form a frame 36, respectively. In other words, the audio signal may be divided-up into frames overlapping in time or being immediately adjacent in time, for example. In this case, bsFrameLength may define the number of parameter time slots 38 per frame, i.e. the time unit at which the SAOC parameters such as OLD and IOC, are computed in an SAOC frame 36 and bsFreqRes may define the number of processing frequency bands for which SAOC parameters are computed, i.e. the number of bands into which the frequency domain is subdivided and for which the SAOC parameters are determined and transmitted. By this measure, each frame is divided-up into time/frequency tiles exemplified in Fig. 2 by dashed lines 39.

[0026] The downmixer 16 calculates SAOC parameters according to the following formulas. In particular, downmixer 16 computes object level differences for each object i as

$$OLD_i = \frac{\sum_n \sum_{k \in m} x_i^{n,k} x_i^{n,k*}}{\max_j \left(\sum_n \sum_{k \in m} x_j^{n,k} x_j^{n,k*} \right)}$$

wherein the sums and the indices n and k, respectively, go through all filter bank time slots 34, and all filter bank subbands 30 which belong to a certain time/frequency tile 39. Thereby, the energies of all subband values x_i of an audio signal or object i are summed up and normalized to the highest energy value of that tile among all objects or audio signals.

[0027] Further the SAOC downmixer 16 is able to compute a similarity measure of the corresponding time/frequency tiles of pairs of different input objects 14_1 to 14_N. Although the SAOC downmixer 16 may compute the similarity measure between all the pairs of input objects 14_1 to 14_N, downmixer 16 may also suppress the signaling of the similarity measures or restrict the computation of the similarity measures to audio objects 14_1 to 14_N which form left or right channels of a common stereo channel. In any case, the similarity measure is called the inter-object cross correlation parameter $IOC_{i,j}$. The computation is as follows

$$IOC_{i,j} = IOC_{j,i} = \mathrm{Re}\left\{ \frac{\sum_n \sum_{k \in m} x_i^{n,k} x_j^{n,k*}}{\sqrt{\sum_n \sum_{k \in m} x_i^{n,k} x_i^{n,k*} \sum_n \sum_{k \in m} x_j^{n,k} x_j^{n,k*}}} \right\}$$

with again indexes n and k going through all subband values belonging to a certain time/frequency tile 39, and i and j denoting a certain pair of audio objects 14_1 to 14_N.

[0028] The downmixer 16 downmixes the objects 14_1 to 14_N by use of gain factors applied to each object 14_1 to 14_N.

[0029] In the case of a stereo downmix signal, which case is exemplified in Fig. 1, a gain factor $D_{1,i}$ is applied to object i and then all such gain amplified objects are summed-up in order to obtain the left downmix channel L0, and gain factors $D_{2,i}$ are applied to object i and then the thus gain-amplified objects are summed-up in order to obtain the right downmix channel R0. Thus, factors $D_{1,i}$ and $D_{2,i}$ form a downmix matrix D of size 2xN with

$$D = \begin{pmatrix} D_{1,1} & \cdots & D_{1N} \\ D_{2,1} & \cdots & D_N \end{pmatrix} \text{ and } \begin{pmatrix} LO \\ RO \end{pmatrix} = D \cdot \begin{pmatrix} Obj_1 \\ \vdots \\ Obj_N \end{pmatrix}.$$

[0030] This downmix prescription is signaled to the decoder side by means of down mix gains DMG_i and, in case of a stereo downmix signal, downmix channel level differences $DCLD_i$.

[0031] The downmix gains are calculated according to:

$$DMG_i = 10\log_{10}\left(D_{1,i}^2 + D_{2,i}^2 + \varepsilon\right),$$

where ε is a small number such as 10^{-9} or 96dB below maximum signal input.

[0032] For the $DCLD_s$ the following formula applies:

$$\mathrm{DCLD}_1 = 10\log_{10}\left(\frac{\mathrm{D}_{1,i}^2}{\mathrm{D}_{2,i}^2}\right).$$

[0033] The downmixer 16 generates the stereo downmix signal according to:

$$\begin{pmatrix} L0 \\ R0 \end{pmatrix} = \begin{pmatrix} D_1 \\ D_2 \end{pmatrix} \cdot \begin{pmatrix} Obj_1 \\ \vdots \\ Obj_N \end{pmatrix}$$

[0034] Thus, in the above-mentioned formulas, parameters OLD and IOC are a function of the audio signals and parameters DMG and DCLD are a function of D. By the way, it is noted that D may be varying in time.

[0035] In case of binaural rendering, which mode of operation of the decoder is described here, the output signal naturally comprises two channels, i.e. M'=2. Nevertheless, the aforementioned rendering information 26 indicates as to how the input signals 14_1 to 14_N are to be distributed onto virtual speaker positions 1 to M where M might be higher than 2. The rendering information, thus, may comprise a rendering matrix ***M*** indicating as to how the input objects obj_i are to be distributed onto the virtual speaker positions j to obtain virtual speaker signals vs_j with j being between 1 and M inclusively and i being between 1 and N inclusively, with

$$\begin{pmatrix} vs_1 \\ \vdots \\ vs_M \end{pmatrix} = \boldsymbol{M} \cdot \begin{pmatrix} Obj_1 \\ \vdots \\ Obj_N \end{pmatrix}$$

[0036] The rendering information may be provided or input by the user in any way. It may even possible that the rendering information 26 is contained within the side information of the SAOC stream 21 itself. Of course, the rendering information may be allowed to be varied in time. For instance, the time resolution may equal the frame resolution, i.e. ***M*** may be defined per frame 36. Even a variance of ***M*** by frequency may be possible. For example, ***M*** could be defined for each tile 39. Below, for example, $\mathbf{M}_{ren}^{l,m}$ will be used for denoting ***M***, with ***m*** denoting the frequency band and 1 denoting the parameter time slice 38.

[0037] Finally, in the following, the HRTFs 27 will be mentioned. These HRTFs describe how a virtual speaker signal j is to be rendered onto the left and right ear, respectively, so that binaural cues are preserved. In other words, for each virtual speaker position j, two HRTFs exist, namely one for the left ear and the other for the right ear. AS will be described in more detail below, it is possible that the decoder is provided with HRTF parameters 27 which comprise, for each virtual speaker position j, a phase shift offset Φ_j describing the phase shift offset between the signals received by both ears and stemming from the same source j, and two amplitude magnifications/attenuations $P_{i,\mathrm{R}}$ and $P_{i,\mathrm{L}}$ for the right and left ear, respectively, describing the attenuations of both signals due to the head of the listener. The HRTF parameter 27 could be constant over time but are defined at some frequency resolution which could be equal to the SAOC parameter resolution, i.e. per frequency band. In the following, the HRTF parameters are given as Φ_j^m, $P_{j,R}^m$ and $P_{j,L}^m$ with *m* denoting the frequency band.

[0038] Fig. 3 shows the SAOC decoder 12 of Fig. 1 in more detail. As shown therein, the decoder 12 comprises a downmix pre-processing unit 40 and an SAOC parameter processing unit 42. The downmix pre-processing unit 40 is configured to receive the stereo downmix signal 18 and to convert same into the binaural output signal 24. The downmix pre-processing unit 40 performs this conversion in a manner controlled by the SAOC parameter processing unit 42. In particular, the SAOC parameter processing unit 42 provides downmix pre-processing unit 40 with a rendering prescription information 44 which the SAOC parameter processing unit 42 derives from the SAOC side information 20 and rendering information 26.

[0039] Fig. 4 shows the downmix pre-processing unit 40 in accordance with an embodiment of the present invention in more detail. In particular, in accordance with Fig. 4, the downmix pre-processing unit 40 comprises two paths connected in parallel between the input at which the stereo downmix signal 18, i.e. $\boldsymbol{X}^{n,k}$ is received, and an output of unit 40 at which the binaural output signal $\hat{\boldsymbol{X}}^{n,k}$ is output, namely a path called dry rendering path 46 into which a dry rendering unit is serially connected, and a wet rendering path 48 into which a decorrelation signal generator 50 and a wet rendering unit 52 are connected in series, wherein a mixing stage 53 mixes the outputs of both rendering paths 46 and 48 to obtain the final result, namely the binaural output signal 24.

[0040] As will be described in more detail below, the dry rendering unit 47 is configured to compute a preliminary binaural output signal 54 from the stereo downmix signal 18 with the preliminary binaural output signal 54 representing the output of the dry rendering path 46 - also called sometimes "dry binaural signal" or just "dry signal" in the following. The dry rendering unit 47 performs its computation based on a dry rendering prescription presented by the SAOC parameter processing unit 42. In the specific embodiment described below, the rendering prescription is defined by a dry rendering matrix $\boldsymbol{G}^{n,k}$. The just-mentioned provision is illustrated in Fig. 4 by means of a dashed arrow.

[0041] The decorrelated signal generator 50 is configured to generate a decorrelated signal $\boldsymbol{X}_d^{n,k}$ from the stereo downmix signal 18 by downmixing such that same is a perceptual equivalent to a mono downmix of the right and left channel of the stereo downmix signal 18 with, however, being decorrelated to the mono downmix. As shown in Fig. 4, the decorrelated signal generator 50 may comprise an adder 56 for summing the left and right channel of the stereo downmix signal 18 at, for example, a ratio 1:1 or, for example, some other fixed ratio to obtain the respective mono downmix 58, followed by a decorrelator 60 for generating the afore-mentioned decorrelated signal $\boldsymbol{X}_d^{n,k}$. The decorrelator 60 may, for example, comprise one or more delay stages in order to form the decorrelated signal $\boldsymbol{X}_d^{n,k}$ from the delayed version or a weighted sum of the delayed versions of the mono downmix 58 or even a weighted sum over the mono downmix 58 and the delayed version(s) of the mono downmix. Of course, there are many alternatives for the decorrelator 60. In effect, the decorrelation performed by the decorrelator 60 and the decorrelated signal generator 50, respectively, tends to lower the inter-channel coherence between the decorrelated signal 62 and the mono downmix 58 when measured by the above-mentioned formula corresponding to the inter-object cross correlation, with substantially maintaining the object level differences thereof when measured by the above-mentioned formula for object level differences.

[0042] The wet rendering unit 52 is configured to compute a corrective binaural output signal 64 from the decorrelated signal 62, the thus obtained corrective binaural output signal 64 representing the output of the wet rendering path 48- also called sometimes "wet binaural signal" or just "wet signal" in the following. The wet rendering unit 52 bases its computation on a wet rendering prescription which, in turn, depends on the dry rendering prescription used by the dry rendering unit 47 as desribed below. Accordingly, the wet rendering prescription which is indicated as $P_2^{n,k}$ in Fig. 4, is obtained from the SAOC parameter processing unit 42 as indicated by the dashed arrow in Fig. 4.

[0043] The mixing stage 53 mixes both binaural output signals 54 and 64 of the dry and wet rendering paths 46 and 48 to obtain the final binaural output signal 24. As shown in Fig. 4, the mixing stage 53 is configured to mix the left and right channels of the binaural output signals 54 and 64 individually and may, accordingly, comprise an adder 66 for summing the left channels thereof and an adder 68 for summing the right channels thereof, respectively.

[0044] After having described the structure of the SAOC decoder 12 and the internal structure of the downmix pre-processing unit 40, the functionality thereof is described in the following. In particular, the detailed embodiments described below present different alternatives for the SAOC parameter processing unit 42 to derive the rendering prescription information 44 thereby controlling the inter-channel coherence of the binaural object signal 24. In other words, the SAOC parameter processing unit 42 not only computes the rendering prescription information 44, but concurrently controls the mixing ratio by which the preliminary and corrective binaural output signals 55 and 64 are mixed into the final binaural output signal 24.

[0045] In accordance with a first alternative, the SAOC parameter processing unit 42 is configured to channel the just-mentioned mixing ratio as shown in Fig. 5. In particular, in a step 80, an actual binaural inter-channel coherence value of the preliminary binaural output signal 54 is determined or estimated by unit 42. In a step 82, SAOC parameter processing unit 42 determines a target binaural inter-channel coherence value. Based on these thus determined inter-channel coherence values, the SAOC parameter processing unit 42 sets the afore-mentioned mixing ratio in step 84. In particular, step 84 may comprise the SAOC parameter processing unit 42 appropriately computing the dry rendering prescription used by dry rendering unit 42 and the wet rendering prescription used by wet rendering unit 52, respectively, based on the inter-channel coherence values determined in steps 80 and 82, respectively.

[0046] In the following, the afore-mentioned alternatives will be described on a mathematical basis. The alternatives differ from each other in the way the SAOC parameter processing unit 42 determines the rendering prescription information 44, including the dry rendering prescription and the wet rendering prescription with inherently controlling the mixing ratio between dry and wet rendering paths 46 and 48. In accordance with the first alternative depicted in Fig. 5, the SAOC parameter processing unit 42 determines a target binaural inter-channel coherence value. As will be described in more detail below, unit 42 may perform this determination based on components of a target coherence matrix $\boldsymbol{F}=\boldsymbol{A}\cdot\boldsymbol{E}\cdot\boldsymbol{A}^*$, with "*" denoting conjugate transpose, A being a target binaural rendering matrix relating the objects/audio signals 1...N to the right and left channel of the binaural output signal 24 and preliminary binaural output signal 54, respectively, and being derived from the rendering information 26 and HRTF parameters 27, and $\boldsymbol{E}$ being a matrix the coefficients of which are derived from the $IOC_{ij}^{l,m}$ and object level differences $OLD_i^{l,m}$. The computation may be performed in the spatial/temporal resolution of the SAOC parameters, i.e. for each *(l,m)*. However, it is further possible to perform the computation in a lower resolution with interpolating between the respective results. The latter statement is also true for the subsequent computations set out below.

[0047] As the target binaural rendering matrix $\boldsymbol{A}$ relates input objects 1...N to the left and right channels of the binaural output signal 24 and the preliminary binaural output signal 54, respectively, same is of size 2xN, i.e.

$$\boldsymbol{A}=\begin{pmatrix} a_{11}\cdots & a_{1N} \\ a_{21}\cdots & a_{2N} \end{pmatrix}$$

[0048] The afore-mentioned matrix $\boldsymbol{E}$ is of size NxN with its coefficients being defined as

$$e_{ij}=\sqrt{OLD_i\cdot OLD_j}\cdot max\left(IOC_{ij},0\right)$$

[0049] Thus, the matrix $\boldsymbol{E}$ with

$$\boldsymbol{E}=\begin{pmatrix} e_{11} & \cdots & e_{1N} \\ \vdots & \ddots & \vdots \\ e_{N1} & \cdots & e_{NN} \end{pmatrix}$$

has along it diagonal the object level differences, i.e.

$$e_{ii}=OLD_i$$

since $IOC_{ij}=1$ *for i = j* whereas matrix $\boldsymbol{E}$ has outside its diagonal matrix coefficients representing the geometric mean of the object level differences of objects i and j, respectively, weighted with the inter-object cross correlation measure IOC_{ij} (provided same is greater than 0 with the coefficients being set to 0 otherwise).

[0050] Compared thereto, the second and third alternatives described below, seek to obtain the rendering matrixes by finding the best match in the least square sense of the equation which maps the stereo downmix signal 18 onto the preliminary binaural output signal 54 by means of the dry rendering matrix $\boldsymbol{G}$ to the target rendering equation mapping the input objects via matrix $\boldsymbol{A}$ onto the "target" binaural output signal 24 with the second and third alternative differing from each other in the way the best match is formed and the way the wet rendering matrix is chosen.

[0051] In order to ease the understanding of the following alternatives, the afore-mentioned description of Figs. 3 and 4 is mathematically re-described. As described above, the stereo downmix signal 18 $\boldsymbol{X}^{n,k}$ reaches the SAOC decoder 12 along with the SAOC parameters 20 and user defined rendering information 26. Further, SAOC decoder 12 and SAOC parameter processing unit 42, respectively, have access to an HRTF database as indicated by arrow 27. The transmitted SAOC parameters comprise object level differences $OLD_i^{l,m}$, inter-object cross correlation values

$IOC_{ij}^{l,m}$, downmix gains $DMG_i^{l,m}$ and downmix channel level differences $DCLD_i^{l,m}$ for all N objects i, j with "*l*, *m*" denoting the respective time/spectral tile 39 with *l* specifying time and *m* specifying frequency. The HRTF parameters 27 are, exemplarily, assumed to be given as $P_{q,L}^m$, and Φ_q^m for all virtual speaker positions or virtual spatial sound source position *q*, for left (L) and right (R) binaural channel and for all frequency bands ***m***.

[0052] The downmix pre-processing unit 40 is configured to compute the binaural output $\hat{X}^{n,k}$, as computed from the stereo downmix $X^{n,k}$ and decorrelated mono downmix signal $X_d^{n,k}$ as

$$\hat{X}^{n,k} = G^{n,k}X^{n,k} + P_2^{n,k}X_d^{n,k}$$

[0053] The decorrelated signal $X_d^{n,k}$ is perceptually equivalent to the sum 58 of the left and right downmix channels of the stereo downmix signal 18 but maximally decorrelated to it according to

$$X_d^{n,k} = \text{decorrFunction}\left(\begin{pmatrix}1 & 1\end{pmatrix}X^{n,k}\right)$$

[0054] Referring to Fig. 4, the decorrelated signal generator 50 performs the function decorrFunction of the above-mentioned formula.

[0055] Further, as also described above, the downmix pre-processing unit 40 comprises two parallel rendering paths 46 and 48. Accordingly, the above-mentioned equation is based on two time/frequency dependent matrices, namely, $G^{l,m}$ for the dry and $P_2^{l,m}$ for the wet rendering path.

[0056] As shown in Fig. 4, the decorrelation on the wet rendering path may be implemented by the sum of the left and right downmix channel being fed into a decorrelator 60 that generates a signal 62, which is perceptually equivalent, but maximally decorrelated to its input 58.

[0057] The elements of the just-mentioned matrices are computed by the SAOC pre-processing unit 42. As also denoted above, the elements of the just-mentioned matrices may be computed at the time/frequency resolution of the SAOC parameters, i.e. for each time slot *l* and each processing band *m.* The matrix elements thus obtained may be spread over frequency and interpolated in time resulting in matrices $E^{n,k}$ and $P_2^{l,m}$ defined for all filter bank time slots *n* and frequency subbands *k*. However, as already above, there are also alternatives. For example, the interpolation could be left away, so that in the above equation the indices *n,k* could effectively be replaced by "*l,m*". Moreover, the computation of the elements of the just-mentioned matrices could even be performed at a reduced time/frequency resolution with interpolating onto resolution *l,m* or *n,k*. Thus, again, although in the following the indices *l,m* indicate that the matrix calculations are performed for each tile 39, the calculation may be performed at some lower resolution wherein, when applying the respective matrices by the downmix pre-processing unit 40, the rendering matrices may be interpolated until a final resolution such as down to the QMF time/frequency resolution of the individual subband values 32.

[0058] According to the above-mentioned first alternative, the dry rendering matrix $G^{l,m}$ is computed for the left and the right downmix channel separately such that

$$G^{l,m} = \begin{pmatrix} P_L^{l,m,1}\cos\left(\beta^{l,m}+\alpha^{l,m}\right)\exp\left(j\frac{\phi^{l,m,1}}{2}\right) & P_L^{l,m,2}\cos\left(\beta^{l,m}+\alpha^{l,m}\right)\exp\left(j\frac{\phi^{l,m,2}}{2}\right) \\ P_R^{l,m,1}\cos\left(\beta^{l,m}-\alpha^{l,m}\right)\exp\left(-j\frac{\phi^{l,m,1}}{2}\right) & P_R^{l,m,2}\cos\left(\beta^{l,m}-\alpha^{l,m}\right)\exp\left(-j\frac{\phi^{l,m,2}}{2}\right) \end{pmatrix}$$

[0059] The corresponding gains $P_L^{l,m,x}$, $P_R^{l,m,x}$ and phase differences $\phi^{l,m,x}$ are defined as

$$P_L^{l,m,x} = \sqrt{\frac{f_{11}^{l,m,x}}{V^{l,m,x}}}, \qquad P_R^{l,m,x} = \sqrt{\frac{f_{22}^{l,m,x}}{V^{l,m,x}}},$$

$$\phi^{l,m,x} = \begin{cases} \arg\left(f_{12}^{l,m,x}\right) & if\ 0 \le m \le const_1 \wedge \frac{\left|f_{12}^{l,m,x}\right|}{\sqrt{f_{11}^{l,m,x} f_{22}^{l,m,x}}} \ge const_2 \\ 0 & else \end{cases}$$

wherein $const_1$ may be, for example, 11 and $const_2$ may be 0.6. The index x denotes the left or right downmix channel and accordingly assumes either 1 or 2.

[0060] Generally speaking, the above condition distinguishes between a higher spectral range and a lower spectral range and ,especially, is (potentially) fulfilled only for the lower spectral range. Additionally or alternatively, the condition is dependent on as to whether one of the actual binaural inter-channel coherence value and the target binaural inter-channel coherence value has a predetermined relationship to a coherence threshold value or not, with the condition being (potentially) fulfilled only if the coherence exceeds the threshold value. The just mentioned individual sub-conditions may, as indicated above, be combined by means of an and operation.

[0061] The scalar $V^{l,m,x}$ is computed as

$$V^{l,m,x} = \boldsymbol{D}^{l,m,x} \boldsymbol{E}^{l,m} \left(\boldsymbol{D}^{l,m,x}\right) + \varepsilon.$$

[0062] It is noted that ε may be the same as or different to the ε mentioned above with respect to the definition of the downmix gains. The matrix ***E*** has already been introduced above. The index (l,m) merely denotes the time/frequency dependence of the matrix computation as already mentioned above. Further, the matrices $\boldsymbol{D}^{l,m,x}$ had also been mentioned above, with respect to the definition of the downmix gains and the downmix channel level differences, so that $\boldsymbol{D}^{l,m,1}$ corresponds to the afore-mentioned $\boldsymbol{D}_1$ and $\boldsymbol{D}^{l,m,2}$ corresponds to the aforementioned $\boldsymbol{D}_2$.

[0063] However, in order to ease the understanding how the SAOC parameter processing unit 42 derives the dry rendering matrix $\boldsymbol{G}^{l,m}$ from the received SAOC parameters, the correspondence between channel downmix matrix $D^{l,m,x}$ and the downmix prescription comprising the downmix gains $DMG_i^{l,m}$ and $DCLD_i^{l,m}$ is presented again, in the inverse direction. In particular, the elements $d_i^{l,m,x}$ of the channel downmix matrix $\boldsymbol{D}^{l,m,x}$ of size 1xN, i.e. $\boldsymbol{D}^{l,m,x} = (d_1^{l,m,x}, \ldots d_N^{l,m,x})$ are given as

$$d_i^{l,m,1} = 10^{\frac{DMG_i^{l,m}}{20}} \sqrt{\frac{\tilde{d}_i^{l,m}}{1+\tilde{d}_i^{l,m}}}, \quad d_i^{l,m,2} = 10^{\frac{DMG_i^{l,m}}{20}} \sqrt{\frac{1}{1+\tilde{d}_i^{l,m}}}$$

with the element $\tilde{d}_i^{l,m}$ being defined as

$$\tilde{d}_i^{l,m} = 10^{\frac{DCLD_i^{l,m}}{10}}.$$

[0064] In the above equation of $G^{l,m}$, the gains $P_L^{l,m,x}$ and $P_R^{l,m,x}$ and the phase differences $\phi^{l,m,x}$ depend on coefficients f_{uv} of a channel-x individual target covariance matrix $F^{l,m,x}$, which, in turn, as will be set out in more detail below, depends on a matrix $\boldsymbol{E}^{l,m,x}$ of size NxN the elements $e_{ij}^{l,m,x}$ of which are computed as

$$e_{ij}^{l,m,x} = e_{ij}^{l,m}\left(\frac{d_i^{l,m,x}}{d_i^{l,m,1}+d_i^{l,m,2}}\right)\left(\frac{d_j^{l,m,x}}{d_j^{l,m,1}+d_j^{l,m,2}}\right)$$

[0065] The elements $e_{ij}^{l,m}$ of the matrix $\mathbf{E}^{l,m}$ of size $N\times N$ are, as stated above, given as $e_{ij}^{l,m} = \sqrt{OLD_i^{l,m}\cdot OLD_j^{l,m}}\cdot\max(IOC_{ij}^{l,m},0)$.

[0066] The just-mentioned target covariance matrix $\mathbf{F}^{l,m,x}$ of size 2x2 with elements $f_{uv}^{l,m,x}$ is, similarly to the covariance matrix $\boldsymbol{F}$ indicated above, given as

$$F^{l,m,x} = A^{l,m}E^{l,m,x}\left(A^{l,m}\right)^{*},$$

where "*" corresponds to conjugate transpose.

[0067] The target binaural rendering matrix $\mathbf{A}^{l,m}$ is derived from the HRTF parameters Φ_q^m, $P_{q,R}^m$ and $P_{q,L}^m$ for all N_{HRTF} virtual speaker positions q and the rendering matrix $\mathbf{M}_{ren}^{l,m}$ and is of size $2\times N$. Its elements $a_{ui}^{l,m}$ define the desired relation between all objects i and the binaural output signal as

$$a_{1,i}^{l,m} = \sum_{q=0}^{N_{\mathrm{HRTF}}-1} m_{q,i}^{l,m}P_{q,\mathrm{L}}^{m}\exp\left(j\frac{\phi_q^m}{2}\right), \qquad a_{2,i}^{l,m} = \sum_{q=0}^{N_{\mathrm{HRTF}}-1} m_{q,i}^{l,m}P_{q,\mathrm{R}}^{m}\exp\left(-j\frac{\phi_q^m}{2}\right).$$

[0068] The rendering matrix $\mathbf{M}_{ren}^{l,m}$ with elements $m_{qi}^{l,m}$ relates every audio object i to a virtual speaker q represented by the HRTF.

[0069] The wet rendering matrix $\boldsymbol{P}_2^{l,m}$ is calculated based on dry rendering matrix $\boldsymbol{G}^{l,m}$ as

$$\boldsymbol{P}_2^{l,m} = \begin{pmatrix} P_L^{l,m}\sin\left(\beta^{l,m}+\alpha^{l,m}\right)\exp\left(j\frac{\arg\left(c_{12}^{l,m}\right)}{2}\right) \\ P_R^{l,m}\sin\left(\beta^{l,m}-\alpha^{l,m}\right)\exp\left(-j\frac{\arg\left(c_{12}^{l,m}\right)}{2}\right) \end{pmatrix}$$

[0070] The gains $P_L^{l,m}$ and $P_R^{l,m}$ are defined as

$$P_L^{l,m} = \sqrt{\frac{c_{11}^{l,m}}{V^{l,m}}},\ P_R^{l,m} = \sqrt{\frac{c_{22}^{l,m}}{V^{l,m}}}.$$

[0071] The 2x2 covariance matrix $\boldsymbol{C}^{l,m}$ with elements $c_{u,v}^{l,m}$ of the dry binaural output signal 54 is estimated as

$$\boldsymbol{C}^{l,m} = \tilde{\boldsymbol{G}}^{l,m}\boldsymbol{D}^{l,m}\boldsymbol{E}^{l,m}\left(\boldsymbol{D}^{l,m}\right)^{*}\left(\tilde{\boldsymbol{G}}^{l,m}\right)^{*}$$

where

$$\tilde{G}^{l,m} = \begin{pmatrix} P_L^{l,m,1} \exp\left(j\frac{\phi^{l,m,1}}{2}\right) & P_L^{l,m,2} \exp\left(j\frac{\phi^{l,m,2}}{2}\right) \\ P_R^{l,m,1} \exp\left(-j\frac{\phi^{l,m,1}}{2}\right) & P_R^{l,m,2} \exp\left(-j\frac{\phi^{l,m,2}}{2}\right) \end{pmatrix}$$

[0072] The scalar $V^{l,m}$ is computed as

$$V^{l,m} = \boldsymbol{W}^{l,m} \boldsymbol{E}^{l,m} \left(\boldsymbol{W}^{l,m}\right)^* + \varepsilon.$$

[0073] The elements $w_i^{l,m}$ of the wet mono downmix matrix $\boldsymbol{W}^{l,m}$ of size 1xN are given as

$$w_i^{l,m} = d_i^{l,m,1} + d_i^{l,m,2}.$$

[0074] The elements $d_{x,i}^{l,m}$ of the stereo downmix matrix $\boldsymbol{D}^{l,m}$ of size 2xN are given as

$$d_{x,i}^{l,m} = d_i^{l,m,x}.$$

[0075] In the above-mentioned equation of $\boldsymbol{G}^{l,m}$, $\alpha^{l,m}$ and $\beta^{l,m}$ represent rotator angles dedicated for ICC control. In particular, the rotator angle $\alpha^{l,m}$ controls the mixing of the dry and the wet binaural signal in order to adjust the ICC of the binaural output 24 to that of the binaural target. When setting the rotator angels, the ICC of the dry binaural signal 54 should be taken into account which is, depending on the audio content and the stereo downmix matrix ***D***, typically smaller than 1.0 and greater than the target ICC. This is in contrast to a mono downmix based binaural rendering where the ICC of the dry binaural signal would always be equal to 1.0.

[0076] The rotator angles $\alpha^{l,m}$ and $\beta^{l,m}$ control the mixing of the dry and the wet binaural signal. The ICC $\rho_C^{l,m}$ of the dry binaural rendered stereo downmix 54 is, in step 80, estimated as

$$\rho_C^{l,m} = \min\left(\frac{\left|c_{12}^{l,m}\right|}{\sqrt{c_{11}^{l,m} c_{22}^{l,m}}}, 1\right).$$

[0077] The overall binaural target ICC $\rho_C^{l,m}$ is, in step 82, estimated as, or determined to be,

$$\rho_T^{l,m} = \min\left(\frac{\left|f_{12}^{l,m}\right|}{\sqrt{f_{11}^{l,m} f_{22}^{l,m}}}, 1\right)$$

[0078] The rotator angles $\alpha^{l,m}$ and $\beta^{l,m}$ for minimizing the energy of the wet signal are then, in step 84, set to be

$$\alpha^{l,m} = \frac{1}{2}\left(\arccos\left(\rho_T^{l,m}\right) - \arccos\left(\rho_C^{l,m}\right)\right),$$

$$\beta^{l,m} = \arctan\left(\tan\left(\alpha^{l,m}\right)\frac{P_R^{l,m} - P_L^{l,m}}{P_L^{l,m} + P_R^{l,m}}\right).$$

[0079] Thus, according to the just-described mathematical description of the functionality of the SAOC decoder 12 for generating the binaural output signal 24, the SAOC parameter processing unit 42 computes, in determining the actual binaural ICC, $\rho_C^{l,m}$ by use of the above-presented equations for $\rho_C^{l,m}$ and the subsidiary equations also presented above. Similarly, SAOC parameter processing unit 42 computes, in determining the target binaural ICC in step 82, the parameter $\rho_C^{l,m}$ by the above-indicated equation and the subsidiary equations. On the basis thereof, the SAOC parameter processing unit 42 determines in step 84 the rotator angles thereby setting the mixing ratio between dry and wet rendering path. With these rotator angles, SAOC parameter processing unit 42 builds the dry and wet rendering matrices or upmix parameters $G^{l,m}$ and $\mathbf{P}_2^{l,m}$ which, in turn, are used by downmix pre-processing unit 40 - at resolution *n,k* - in order to derive the binaural output signal 24 from the stereo downmix 18.

[0080] It should be noted that the afore-mentioned first alternative may be varied in some way. For example, the above-presented equation for the interchannel phase difference $\Phi_C^{l,m}$ could be changed to the extent that the second sub-condition could compare the actual ICC of the dry binaural rendered stereo downmix to const_2 rather than the ICC determined from the channel individual covariance matrix $F^{l,m,x}$ so that in that equation the portion $\frac{|f_{12}^{l,m,x}|}{\sqrt{f_{11}^{l,m,x} f_{22}^{l,m,x}}}$ would be replaced by the term $\frac{|c_{12}^{l,m}|}{\sqrt{c_{11}^{l,m} c_{22}^{l,m}}}$.

[0081] Further, it should be noted that, in accordance with the notation chosen, in some of the above equations, a matrix of all ones has been left away when a scalar constant such as ε was added to a matrix so that this constant is added to each coefficient of the respective matrix.

[0082] An alternative generation of the dry rendering matrix with higher potential of object extraction is based on a joint treatment of the left and right downmix channels. Omitting the subband index pair for clarity, the principle is to aim at the best match in the least squares sense of

$$\hat{X} = GX$$

to the target rendering

$$\mathbf{Y} = \mathbf{AS}\ .$$

[0083] This yields the target covariance matrix:

$$YY^* = ASS^*A^*$$

where the complex valued target binaural rendering matrix **A** is given in a previous formula and the matrix **S** contains the original objects subband signals as rows.

[0084] The least squares match is computed from second order information derived from the conveyed object and downmix data. That is, the following substitutions are performed

$$XX^* \leftrightarrow DED^*,$$

$$YX^* \leftrightarrow AED^*,$$

$$YY^* \leftrightarrow AEA^*.$$

To motivate the substitutions, recall that SAOC object parameters typically carry information on the object powers (OLD) and (selected) inter-object cross correlations (IOC). From these parameters, the NxN object covariance matrix **E** is derived, which represents an approximation to **SS*,** i.e. **E**≈**SS*,** yielding **YY*=AEA*.**

[0085] Further, **X=DS** and the downmix covariance matrix becomes:

$$XX^*=DSS^*D^*,$$

which again can be derived from E by XX*=DED*.

[0086] The dry rendering matrix **G** is obtained by solving the least squares problem

$$\min\{\text{norm}\{\text{ Y-X }\}\}.$$

$$G = G_0 = YX^*\left(XX^*\right)^{-1}$$

where **YX*** is computed as **YX*=AED*.**

[0087] Thus, dry rendering unit 42 determines the binaural output signal $\hat{X}$ form the downmix signal X by use of the 2x2 dry rendering matrix G, by $\hat{X} = GX$, and the SAOC parameter processing unit determines G by use of the above formulae to be

$$G = AED^*(DED^*)^{-1},$$

[0088] Given this complex valued dry rendering matrix, the complex valued wet rendering matrix **P -** formerly denoted P_2 - is computed in the SAOC parameter processing unit 42 by considering the missing covariance error matrix

$$\Delta R = YY^* - G_0 XX^* G_0^*.$$

[0089] It can be shown that this matrix is positive and a preferred choice of **P** is given by choosing a unit norm eigenvector **u** corresponding to the largest eigenvalue A of Δ**R** and scaling it according to

$$\mathbf{P} = \sqrt{\frac{\lambda}{V}}\mathbf{u},$$

where the scalar V is computed as noted above, i.e. $V = WE(W)^* + \varepsilon$.

[0090] In other words, since the wet rendering path is installed to correct the correlation of the obtained dry solution, $\Delta R = AEA^* - G_0 DED^* G_0^*$.represents the missing covariance error matrix, i.e. $YY^* = \hat{X}\hat{X}^* + \Delta R$ or, respectively, $\Delta R = YY^* - \hat{X}\hat{X}^*$, and, therefore, the SAOC parameter processing unit 42 stets P such that $PP^* = \Delta R$, one solution for which is given by choosing the above-mentioned unit norm eigenvector **u.**

[0091] A third method for generating dry and wet rendering matrices represents an estimation of the rendering parameters based on cue constrained complex prediction and combines the advantage of reinstating the correct complex covariance structure with the benefits of the joint treatment of downmix channels for improved object extraction. An additional opportunity offered by this method is to be able to omit the wet upmix altogether in many cases, thus paving the way for a version of binaural rendering with lower computational complexity. As with the second alternative, the third alternative presented below is based on a joint treatment of the left and right downmix channels.

[0092] The principle is to aim at the best match in the least squares sense of

$$\hat{\mathbf{X}} = \mathbf{GX}$$

to the target rendering **Y** = **AS under** the constraint of correct complex covariance

$$GXX^*G^* + VPP^* = \hat{Y}\hat{Y}^*.$$

[0093] Thus, it is the aim to find a solution for G and P, such that

1) $\hat{Y}\hat{Y}^* = YY^*$ (being the constraint to the formulation in 2); and

2) min{norm{$Y-\hat{Y}$}}, as it was requested within the second alternative.

[0094] From the theory of Lagrange multipliers, it follows that there exists a self adjoint matrix $M = M^*$, such that

$$\mathbf{MP} = \mathbf{0},$$

and

$$MGXX^* = YX^*$$

[0095] In the generic case where both $\mathbf{YX^*}$ and $\mathbf{XX^*}$ are non-singular it follows from the second equation that **M** is non-singular, and therefore $\mathbf{P} = \mathbf{0}$ is the only solution to the first equation. This is a solution without wet rendering. Setting $\mathbf{K} = \mathbf{M^{-1}}$ it can be seen that the corresponding dry upmix is given by

$$\mathbf{G} = \mathbf{KG_0}$$

where $\mathbf{G_0}$ is the predictive solution derived above with respect to the second alternative, and the self adjoint matrix **K** solves

$$\mathbf{KG_0XX^*G_0^*K^*} = \mathbf{YY^*}.$$

[0096] If the unique positive and hence selfadjoint matrix square root of the matrix $\mathbf{G_0XX^*G_0^*}$ is denoted by **Q**, then the solution can be written as

$$\mathbf{K} = \mathbf{Q^{-1}(QYY^*Q)^{1/2}Q^{-1}}.$$

[0097] Thus, the SAOC parameter processing unit 42 determines **G** to be $\mathbf{KG_0} = \mathbf{Q^{-1}(QYY^*Q)^{1/2}Q^{-1}\ G_0} = \mathbf{(G_0DED^*G_0^*)^{-1}(G_0\ DED^*G_0^*\ AEA^*\ G_0\ DED^*G_0^*)^{1/2}(G_0\ DED^*G_0^*)^{-1}\ G_0}$ **with** $\mathbf{G_0} = \mathbf{AED^*\ (DED^*)^{-1}}$.

[0098] For the inner square root there will in general be four self-adjoint solutions, and the solution leading to the best match of **X** to **Y** is chosen.

[0099] In practice, one has to limit the dry rendering matrix $\mathbf{G} = \mathbf{KG_0}$ to a maximum size, for instance by limiting condition on the sum of absolute values squares of all dry rendering matrix coefficients, which can be expressed as

$$\mathrm{trace}(\mathbf{GG^*}) \le g_{max}.$$

[0100] If the solution violates this limiting condition, a solution that lies on the boundary is found instead. This is achieved by adding constraint

$$\mathrm{trace}(\mathbf{GG^*}) = g_{max}$$

to the previous constraints and re-deriving the Lagrange equations. It turns out that the previous equation

$$\mathbf{MGXX^* = YX^*}$$

has to be replaced by

$$\mathbf{MGXX^* + \mu I = YX^*}$$

where μ is an additional intermediate complex parameter and **I** is the 2x2 identity matrix. A solution with nonzero wet rendering **P** will result. In particular, a solution for the wet rendering matrix can be found by $\mathbf{PP^*} = (\mathbf{YY^*} - \mathbf{GXX^*G^*}) / V = (\mathbf{AEA^*} - \mathbf{GDED^*G^*}) / V$, wherein the choice of **P** is preferably based on the eigenvalue consideration already stated above with respect to the second alternative, and V is $\mathbf{WEW^*} +_{\varepsilon}$. The latter determination of **P** is also done by the SAOC parameter processing unit 42.

[0101] The thus determined matrices **G** and **P** are then used by the wet and dry rendering units as described earlier.

[0102] If a low complexity version is required, the next step is to replace even this solution with a solution without wet rendering. A preferred method to achieve this is to reduce the requirements on the complex covariance to only match on the diagonal, such that the correct signal powers are still achieved in the right and left channels, but the cross covariance is left open.

[0103] Regarding the first alternative, subjective listening tests were conducted in an acoustically isolated listening room that is designed to permit high-quality listening. The result is outlined below.

[0104] The playback was done using headphones (STAX SR Lambda Pro with Lake-People D/A Converter and STAX SRM-Monitor). The test method followed the standard procedures used in the spatial audio verification tests, based on the "Multiple Stimulus with Hidden Reference and Anchors" (MUSHRA) method for the subjective assessment of intermediate quality audio.

[0105] A total of 5 listeners participated in each of the performed tests. All subjects can be considered as experienced listeners. In accordance with the MUSHRA methodology, the listeners were instructed to compare all test conditions against the reference. The test conditions were randomized automatically for each test item and for each listener. The subjective responses were recorded by a computer-based MUSHRA program on a scale ranging from 0 to 100. An instantaneous switching between the items under test was allowed. The MUSHRA tests have been conducted to assess the perceptual performance of the described stereo-to-binaural processing of the MPEG SAOC system.

[0106] In order to assess a perceptual quality gain of the described system compared to the mono-to-binaural performance, items processed by the mono-to-binaural system were also included in the test. The corresponding mono and stereo downmix signals were AAC-coded at 80 *kbits* per second and per channel.

[0107] As HRTF database "KEMAR_MIT_COMPACT" was used. The reference condition has been generated by binaural filtering of objects with the appropriately weighted HRTF impulse responses taking into account the desired rendering. The anchor condition is the low pass filtered reference condition (at 3.5kHz).

[0108] Table 1 contains the list of the tested audio items.

Table 1 - Audio items of the listening tests

Listening items	Nr. mono/stereo objects	object angles object gains (dB)
discol disco2	10/0	[-30, 0, -20, 40, 5,-5, 120, 0, -20, -40] [-3, -3, -3, -3, -3, -3, -3, -3, -3,-3] [-30, 0, -20, 40, 5, -5, 120, 0, -20, -40] [-12, -12, 3, 3, -12, -12, 3, -12, 3, -12]
coffee1 coffee2	6/0	[10, -20, 25, -35, 0, 120 [0, -3, 0, 0, 0, 0] [10, -20, 25, -35, 0, 120] [3, -20, -15, -15, 3, 3]
pop2	1/5	[0, 30, -30, -90, 90, 0, 0, -120, 120, -45, 45] [4, -6, -6, 4, 4, -6, -6, -6, -6, -16, -16]

[0109] Five different scenes have been tested, which are the result of rendering (mono or stereo) objects from 3 different object source pools. Three different downmix matrices have been applied in the SAOC encoder, see Table. 2.

Table 2 - Downmix types

Downmix type	Mono	Stereo	Dual mono
Matlab notation	dmx1=ones(1,N);	dmx2=zeros(2,N); dmx2(1,1:2:N)=1; smx2(2,2:2:N)=1:	dmx3=ones(2,N):

[0110] The upmix presentation quality evaluation tests have been defined as listed in Table 3.

Table 1 Table 3 - Listening test conditions

Text condition	Downmix type	Core-coder
x-1-b	Mono	AAC@80kbps
x-2-b	Stereo	AAC@160kbps
x-2-b Dual/Mono	Dual Mono	AAC@160kbps
5222	Stereo	AAC@160kbps
5222 DualMono	Dual Mono	AAC@160kbps

[0111] The "5222" system uses the stereo downmix pre-processor as described in ISO/IEC JTC 1/SC 29/WG 11 (MPEG), Document N10045, "ISO/IEC CD 23003-2:200x Spatial Audio Object Coding (SAOC)", 85th MPEG Meeting, July 2008, Hannover, Germany, with the complex valued binaural target rendering matrix $\mathbf{A}^{l,m}$ as an input. That is, no ICC control is performed. Informal listening test have shown that by taking the magnitude of $\mathbf{A}^{l,m}$ for upper bands instead of leaving it complex valued for all bands improves the performance. The improved "5222" system has been used in the test.

[0112] A short overview in terms of the diagrams demonstrating the obtained listening test results can be found in Figure 6. These plots show the average MUSHRA grading per item over all listeners and the statistical mean value over all evaluated items together with the associated 95% confidence intervals. One should note that the data for the hidden reference is omitted in the MUSHRA plots because all subjects have identified it correctly.

[0113] The following observations can be made based upon the results of the listening tests:

- "x-2-b_DualMono" performs comparable to "5222".
- "x-2-b_DualMono" performs clearly better than "5222_DualMono".
- "x-2-b_DualMono" performs comparable to "x-1-b"
- "x-2-b" implemented according to the above first alternative, performs slightly better than all other conditions.
- item "disco1" does not show much variation in the results and may not be suitable.

[0114] Thus, a concept for binaural rendering of stereo downmix signals in SAOC has been described above, that fulfils the requirements for different downmix matrices. In particular the quality for dual mono like downmixes is the same as for true mono downmixes which has been verified in a listening test. The quality improvement that can be gained from stereo downmixes compared to mono downmixes can also be seen from the listening test. The basic processing blocks of the above embodiments were the dry binaural rendering of the stereo downmix and the mixing with a wet binaural signal with a proper combination of both blocks.

- In particular, the wet binaural signal was computed using one decorrelator with mono downmix input so that the left and right powers and the IPD are the same as in the dry binaural signal.
- The mixing of the wet and dry binaural signals was controlled by the target ICC and the ICC of the dry binaural signal so that typically less decorrelation is required than for mono downmix based binaural rendering resulting in higher overall sound quality.
- Further, the above embodiments, may be easily modified for any combination of mono/stereo downmix input and mono/stereo/binaural output in a stable manner.

[0115] In other words, embodiments providing a signal processing structure and method for decoding and binaural rendering of stereo downmix based SAOC bitstreams with inter-channel coherence control were described above. All combinations of mono or stereo downmix input and mono, stereo or binaural output can be handled as special cases of the described stereo downmix based concept. The quality of the stereo downmix based concept turned out to be typically better than the mono Downmix based concept which was verified in the above described MUSHRA listening test.

[0116] In Spatial Audio Object Coding (SAOC) ISO/IEC JTC 1/SC 29/WG 11 (MPEG), Document N10045, "ISO/IEC CD 23003-2:200x Spatial Audio Object Coding (SAOC)", 85th MPEG Meeting, July 2008, Hannover, Germany, multiple audio objects are downmixed to a mono or stereo signal. This signal is coded and transmitted together with side information (SAOC parameters) to the SAOC decoder. The above embodiments enable the inter-channel coherence (ICC) of the binaural output signal being an important measure for the perception of virtual sound source width, and being, due to the encoder downmix, degraded or even destroyed, (almost) completely to be corrected.

[0117] The inputs to the system are the stereo downmix, SAOC parameters, spatial rendering information and an HRTF database. The output is the binaural signal. Both input and output are given in the decoder transform domain typically by means of an oversampled complex modulated analysis filter bank such as the MPEG Surround hybrid QMF filter bank, ISO/IEC 23003-1:2007, Information technology - MPEG audio technologies - Part 1: MPEG Surround with sufficiently low inband aliasing. The binaural output signal is converted back to PCM time domain by means of the synthesis filter bank. The system is thus, in other words, an extension of a potential mono downmix based binaural rendering towards stereo Downmix signals. For dual mono Downmix signals the output of the system is the same as for such mono Downmix based system. Therefore the system can handle any combination of mono/stereo Downmix input and mono/stereo/binaural output by setting the rendering parameters appropriately in a stable manner.

[0118] In even other words, the above embodiments perform binaural rendering and decoding of stereo downmix based SAOC bit streams with ICC control. Compared to a mono downmix based binaural rendering, the embodiments can take advantage of the stereo downmix in two ways:

- Correlation properties between objects in different downmix channels are partly preserved
- Object extraction is improved since few objects are present in one downmix channel

[0119] Thus, a concept for binaural rendering of stereo downmix signals in SAOC has been described above that fulfils the requirements for different downmix matrices. In particular, the quality for dual mono like downmixes is the same as for true mono downmixes which has been verified in a listening test. The quality improvement that can be gained from stereo downmixes compared to mono downmixes can also be seen from the listening test. The basic processing blocks of the above embodiments were the dry binaural rendering of the stereo downmix and the mixing with a wet binaural signal with a proper combination of both blocks. In particular, the wet binaural signal was computed using one decorrelator with mono downmix input so that the left and right powers and the IPD are the same as in the dry binaural signal. The mixing of the wet and dry binaural signals was controlled by the target ICC and the mono downmix based binaural rendering resulting in higher overall sound quality. Further, the above embodiments may be easily modified for any combination of mono/stereo downmix input and mono/stereo/binaural output in a stable manner. In accordance with the embodiments, the stereo downmix signal $X^{n,k}$ is taken together with the SAOC parameters, user defined rendering information and an HRTF database as inputs. The transmitted SAOC parameters are $OLD_i^{l,m}$ (object level differences), $IOC_{ij}^{l,m}$ (inter-object cross correlation), $DMG_i^{l,m}$ (downmix gains) and $DCLD_i^{l,m}$ (downmix channel level differences) for all N objects i,j. The HRTF parameters were given as $P_{q,L}^m$, $P_{q,R}^m$ and ϕ_q^m for all HRTF database index *q,* which is associated with a certain spatial sound source position.

[0120] Finally, it is noted that although within the above description, the terms "inter-channel coherence" und "inter-object cross correlation" have been constructed differently in that "coherence" is used in one term and "cross correlation" is used in the other, the latter terms may be used interchangeably as a measure for similarity between channels and objects, respectively.

[0121] Depending on an actual implementation, the inventive binaural rendering concept can be implemented in hardware or in software. Therefore, the present invention also relates to a computer program, which can be stored on a computer-readable medium such as a CD, a disk, DVD, a memory stick, a memory card or a memory chip. The present invention is, therefore, also a computer program having a program code which, when executed on a computer, performs the inventive method of encoding, converting or decoding described in connection with the above figures. Furthermore, it is noted that all steps indicated in the flow diagrams are implemented by respective means in the decoder, respectively, an that the implementations may comprise subroutines running on a CPU, circuit parts of an ASIC or the like. A similar statement is true for the functions of the blocks in the block diagrams

[0122] In other words, according to an embodiment an apparatus for binaural rendering a multi-channel audio signal (21) into a binaural output signal (24) is provided, the multi-channel audio signal (21) comprising a stereo downmix signal (18) into which a plurality of audio signals (14_1-14_N) are downmixed, and side information (20) comprising a downmix information (DMG, DCLD) indicating, for each audio signal, to what extent the respective audio signal has been mixed into a first channel (L0) and a second channel (R0) of the stereo downmix signal (18), respectively, as well as object level information (OLD) of the plurality of audio signals and inter-object cross correlation information (IOC) describing similarities between pairs of audio signals of the plurality of audio signals, the apparatus comprising means (47) for computing, based on a first rendering prescription ($\boldsymbol{G}^{l,m}$) depending on the inter-object cross correlation information, the object level information, the downmix information, rendering information relating each audio signal to a virtual speaker position and HRTF parameters, a preliminary binaural output signal (54) from the first and second channels of the stereo downmix signal (18); means (50) for generating a decorrelated signal ($\boldsymbol{X}_d^{n,k}$) as an perceptual equivalent to a mono downmix (58) of the first and second channels of the stereo downmix signal (18) being, however, decorrelated to the mono downmix (58); means (52) for computing, depending on a second rendering prescription ($\boldsymbol{P}_2^{l,m}$) depending on the inter-object cross correlation information, the object level information, the downmix information, the rendering information and the HRTF parameters, a corrective binaural output signal (64) from the decorrelated signal (62); and means (53) for mixing the preliminary binaural output signal (54) with the corrective binaural output signal (64) to obtain the binaural output signal (24).

References

[0123]

ISO/IEC JTC 1/SC 29/WG 11 (MPEG), Document N10045, "ISO/IEC CD 23003-2:200x Spatial Audio Object Coding (SAOC)", 85th MPEG Meeting, July 2008, Hannover, Germany

EBU Technical recommendation: "MUSHRA-EBU Method for Subjective Listening Tests of Intermediate Audio Quality", Doc. B/AIM022, October 1999.

ISO/IEC 23003-1:2007, Information technology - MPEG audio technologies - Part 1: MPEG Surround

ISO/IEC JTC1/SC29/WG11 (MPEG), Document N9099: "Final Spatial Audio Object Coding Evaluation Procedures and Criterion". April 2007, San Jose, USA

Jeroen, Breebaart, Christof Faller: Spatial Audio Processing. MPEG Surround and Other Applications. Wiley & Sons, 2007.

Jeroen, Breebaart et al.: Multi-Channel goes Mobile : MPEG Surround Binaural Rendering. AES 29th International Conference, Seoul, Korea, 2006.

Claims

1. Apparatus for binaural rendering a multi-channel audio signal (21) into a binaural output signal (24), the multi-channel audio signal (21) comprising a stereo downmix signal (18) into which a plurality of audio signals (14_1-14_N) are downmixed, and side information (20) comprising a downmix information (DMG, DCLD) indicating, for each audio signal, to what extent the respective audio signal has been mixed into a first channel (L0) and a second channel (R0) of the stereo downmix signal (18), respectively, as well as object level information (OLD) of the plurality of audio signals and inter-object cross correlation information (IOC) describing similarities between pairs of audio signals of the plurality of audio signals, the apparatus being configured to:

compute (47), based on a first rendering prescription ($\boldsymbol{G}^{l,m}$) depending on the inter-object cross correlation information, the object level information, the downmix information, rendering information relating each audio signal to a virtual speaker position and HRTF parameters, a preliminary binaural output signal (54) from the first and second channels of the stereo downmix signal (18);

generate (50), from the stereo downmix signal (18), a decorrelated signal ($\boldsymbol{X}_d^{n,k}$) as a perceptual equivalent to a mono downmix (58) of the first and second channels of the stereo downmix signal (18) being, however, decorrelated to the mono downmix (58);

compute (52), depending on a second rendering prescription ($\boldsymbol{P}_2^{l,m}$) depending on the inter-object cross correlation information, the object level information, the downmix information, the rendering information and the HRTF parameters, a corrective binaural output signal (64) from the decorrelated signal (62); and

mix (53) the preliminary binaural output signal (54) with the corrective binaural output signal (64) to obtain the binaural output signal (24).

2. Apparatus according to claim 1, wherein the apparatus is further configured to, in generating the decorrelated signal ($X_d^{n,k}$) , sum the first and second channel of the stereo downmix signal (18) and decorrelate the sum to obtain the decorrelated signal (62).

3. Apparatus to claim 1 or 2 further configured to:

estimate (80) an actual binaural inter-channel coherence value of the preliminary binaural output signal (54); determine (82) a target binaural inter-channel coherence value; and
set (84) a mixing ratio determining to which extent the binaural output signal (24) is influenced by the first and second channels of the stereo downmix signal (18) as processed by the computation (47) of the preliminary binaural output signal (54) and the first and second channels of the stereo downmix signal (18) as processed by the generation (50) of a decorrelated signal and the computation (52) of the corrective binaural output signal (64), respectively, based on the actual binaural inter-channel coherence value and the target binaural inter-channel coherence value.

4. Apparatus to claim 3 wherein the apparatus is further configured to, in setting the mixing ratio, set the mixing ratio by setting the first rendering prescription ($G^{l,m}$) and the second rendering prescription ($P_2^{l,m}$) based on the actual binaural inter-channel coherence value and the target binaural inter-channel coherence value.

5. Apparatus according to claim 3 or 4, wherein the apparatus is further configured to, in determining the target binaural inter-channel coherence value, perform the determination based on components of a target covariance matrix $F = A\,E\,A^*$, with "*" denoting conjugate transpose, A being a target binaural rendering matrix relating the audio signals to the first and second channels of the binaural output signal, respectively, and being uniquely determined by the rendering information and the HRTF parameters, and E being a matrix being uniquely determined by the inter-object cross correlation information and the object level information.

6. Apparatus according to claim 5, wherein the apparatus is further configured to, in computing the preliminary binaural output signal (54), perform the computation so that

$$\hat{X}_1 = G \cdot X$$

where X is a 2x1 vector the components of which correspond to the first and second channels of the stereo downmix signal (18), $\hat{X}_1$ is a 2x1 vector the components of which correspond to the first and second channels of the preliminary binaural output signal (54), G is a first rendering matrix representing the first rendering prescription and having a size of 2x2 with

$$G = \begin{pmatrix} P_L^1 \cos(\beta+\alpha)\exp\left(j\frac{\phi^1}{2}\right) & P_L^2 \cos(\beta+\alpha)\exp\left(j\frac{\phi^2}{2}\right) \\ P_R^2 \cos(\beta-\alpha)\exp\left(-j\frac{\phi^1}{2}\right) & P_R^2 \cos(\beta-\alpha)\exp\left(-j\frac{\phi^2}{2}\right) \end{pmatrix}$$

wherein, with $x \in \{1,2\}$,

$$P_L^x = \sqrt{\frac{f_{11}^x}{V^x}}, \qquad P_R^x = \sqrt{\frac{f_{22}^x}{V^x}},$$

$$\phi^x = \begin{cases} \arg\left(f_{12}^x\right) & \textit{if a first condition applies} \\ 0 & \textit{otherwise} \end{cases}$$

wherein f_{11}^x, f_{12}^x and f_{22}^x are coefficients of sub-target covariance matrices $\boldsymbol{F^x}$ of size 2x2 with $\boldsymbol{F^x} = \boldsymbol{A}\ \boldsymbol{E^x}\ \boldsymbol{A}^*$,

wherein $e_{ij}^x = e_{ij}\left(\frac{d_i^x}{d_i^1+d_i^2}\right)\left(\frac{d_j^x}{d_i^1+d_i^2}\right)$ are coefficients of NxN matrix E^x, N being the number of audio signals, e_{ij} are coefficients of the matrix $\boldsymbol{E}$ being of size NxN, and d_i^x are uniquely determined by the downmix information, wherein d_i^1 indicates the extent to which audio signal i has been mixed into the first channel of the stereo downmix signal (18) and d_i^2 defines to what extent audio signal i has been mixed into the second channel of the stereo output signal (18),

wherein V^x is a scalar with $V^x = \boldsymbol{D^x E}(\boldsymbol{D^x})^* + \varepsilon$ and $\boldsymbol{D^x}$ is a 1xN matrix the coefficients of which are d_i^x, wherein the apparatus is further configured to, in computing a corrective binaural output signal (64), perform the computation such that

$$\hat{X}_2 = P_2 \cdot X_d$$

where $\boldsymbol{X_d}$ is the decorrelated signal, $\hat{\boldsymbol{X}}_2$ is a 2x1 vector the components of which correspond to first and second channels of the corrective binaural output signal (64), and $\boldsymbol{P_2}$ is a second rendering matrix representing the second rendering prescription and having a size 2x2 with

$$\boldsymbol{P}_2 = \begin{pmatrix} P_L \sin(\beta + \alpha)\exp\left(j\frac{\arg(c_{12})}{2}\right) \\ P_R \sin(\beta - \alpha)\exp\left(-j\frac{\arg(c_{12})}{2}\right) \end{pmatrix}$$

wherein gains P_L and P_R are defined as

$$P_L = \sqrt{\frac{c_{11}}{V}},\ P_R = \sqrt{\frac{c_{22}}{V}}$$

wherein c_{11} and c_{21} are coefficients of a 2x2 covariance matrix $\boldsymbol{C}$ of the preliminary binaural output signal (54) with

$$\boldsymbol{C} = \tilde{\boldsymbol{G}}\boldsymbol{DED}^*\tilde{\boldsymbol{G}}^*$$

wherein V is a scalar with $V = \boldsymbol{W}\ \boldsymbol{E}\ \boldsymbol{W}^* + \varepsilon$, $\boldsymbol{W}$ is a mono downmix matrix of size 1xN the coefficients of which are uniquely determined by d_i^x, $\boldsymbol{D} = \begin{pmatrix} \boldsymbol{D}^1 \\ \boldsymbol{D}^2 \end{pmatrix}$, and $\tilde{\boldsymbol{G}}$ is

$$\tilde{\boldsymbol{G}}^{l,m} = \begin{pmatrix} P_L^1 \exp\left(j\frac{\phi^1}{2}\right) & P_L^{l,m,2} \exp\left(j\frac{\phi^2}{2}\right) \\ P_R^1 \exp\left(-j\frac{\phi^1}{2}\right) & P_R^2 \exp\left(-j\frac{\phi^2}{2}\right) \end{pmatrix},$$

wherein the apparatus is further configured to, in estimating the actual binaural inter-channel coherence value, determine the actual binaural inter-channel coherence value as

$$\rho_C = \min\left(\frac{|c_{12}|}{\sqrt{c_{11}c_{22}}},1\right)$$

wherein the apparatus is further configured to, in determining the target binaural inter-channel coherence value, determine the target binaural inter-channel coherence value as

$$\rho_T = \min\left(\frac{|f_{12}|}{\sqrt{f_{11}f_{22}}},1\right),$$

and
wherein the apparatus is further configured to, in setting the mixing ratio, determine rotator angles α and β according to

$$\alpha = \frac{1}{2}\left(\arccos(\rho_T) - \arccos(\rho_C)\right),$$

$$\beta = \arctan\left(\tan(\alpha)\frac{P_R - P_L}{P_L + P_R}\right),$$

with ε denoting a small constant for avoiding divisions by zero, respectively.

7. Apparatus according to claim 1, wherein the apparatus is further configured to, in computing the preliminary binaural output signal (54), perform the computation so that

$$\hat{\boldsymbol{X}}_1 = \boldsymbol{G} \cdot \boldsymbol{X}$$

where $\boldsymbol{X}$ is a 2x1 vector the components of which correspond to the first and second channels of the stereo downmix signal (18), $\hat{\boldsymbol{X}}_1$ is a 2x1 vector the components of which correspond to the first and second channels of the preliminary binaural output signal (54), $\boldsymbol{G}$ is a first rendering matrix representing the first rendering prescription and having a size of 2x2 with

$$\boldsymbol{G} = \quad \boldsymbol{AED}^*(\boldsymbol{DED}^*)^{-1},$$

where $\boldsymbol{E}$ is a matrix being uniquely determined by the inter-object cross correlation information and the object level information;
$\boldsymbol{D}$ is a 2xN matrix the coefficients d_{ij} are uniquely determined by the downmix information, wherein d_{1j} indicates the extent to which audio signal j has been mixed into the first channel of the stereo downmix signal (18) and d_{2j} defines to what extent audio signal j has been mixed into the second channel of the stereo output signal (18);
$\boldsymbol{A}$ is a target binaural rendering matrix relating the audio signals to the first and second channels of the binaural output signal, respectively, and is uniquely determined by the rendering information and the HRTF parameters,
wherein the apparatus is further configured to, in computing a corrective binaural output signal (64), perform the computation such that

$$\hat{\boldsymbol{X}}_2 = \boldsymbol{P} \cdot \boldsymbol{X}_d$$

where $\boldsymbol{X}_d$ is the decorrelated signal, $\hat{\boldsymbol{X}}_2$ is a 2x1 vector the components of which correspond to first and second channels of the corrective binaural output signal (64), and $\boldsymbol{P}$ is a second rendering matrix representing the second rendering prescription and having a size 2x2 and is determined such that **PP*=**Δ***Rb***, with Δ***R*** = ***A****EA**-***G***$_0$*DED**$\boldsymbol{G}_0$* with $\boldsymbol{G}_0$ **=** ***G***.

8. Apparatus according to claim 1, wherein the apparatus is further configured to, in computing the preliminary binaural output signal (54), perform the computation so that

$$\hat{\boldsymbol{X}}_1 = \boldsymbol{G} \cdot \boldsymbol{X}$$

where ***X*** is a 2x1 vector the components of which correspond to the first and second channels of the stereo downmix signal (18), $\hat{\boldsymbol{X}}_1$, is a 2x1 vector the components of which correspond to the first and second channels of the preliminary binaural output signal (54), ***G*** is a first rendering matrix representing the first rendering prescription and having a size of 2x2 with

$$\boldsymbol{G} = (\mathbf{G_0 DED^* G_0^*})^{-1}(\mathbf{G_0\ DED^* G_0^*\ AEA^*\ G_0\ DED^* G_0^*})^{1/2}(\mathbf{G_0\ DED^* G_0^*})^{-1}\ \mathbf{G_0}$$
$$\text{with } \mathbf{G_0} = \mathbf{AED^*}\,(\mathbf{DED^*})^{-1}$$

where ***E*** is a matrix being uniquely determined by the inter-object cross correlation information and the object level information;
D is a 2xN matrix the coefficients d_{ij} are uniquely determined by the downmix information, wherein d_{1j} indicates the extent to which audio signal j has been mixed into the first channel of the stereo downmix signal (18) and d_{2j} defines to what extent audio signal j has been mixed into the second channel of the stereo output signal (18);
A is a target binaural rendering matrix relating the audio signals to the first and second channels of the binaural output signal, respectively, and is uniquely determined by the rendering information and the HRTF parameters,
wherein the apparatus is further configured to, in computing a corrective binaural output signal (64), perform the computation such that

$$\hat{\boldsymbol{X}}_2 = \boldsymbol{P} \cdot \boldsymbol{X}_d$$

where $\boldsymbol{X}_d$ is the decorrelated signal, $\hat{\boldsymbol{X}}_2$ is a 2x1 vector the components of which correspond to first and second channels of the corrective binaural output signal (64), and ***P*** is a second rendering matrix representing the second rendering prescription and having a size 2x2 and is determined such that **PP*=(AEA*** - **GDED*G***) / *V* with *V* being a scalar.

9. Apparatus according to ay of the preceding claims, wherein the downmix information (DMG, DCLD) is time-dependent, and the object level information (OLD) and the inter-object cross correlation information (IOC) are time and frequency dependent.

10. Method for binaural rendering a multi-channel audio signal (21) into a binaural output signal (24), the multi-channel audio signal (21) comprising a stereo downmix signal (18) into which a plurality of audio signals (14_1-14_N) are downmixed, and side information (20) comprising a downmix information (DMG, DCLD) indicating, for each audio signal, to what extent the respective audio signal has been mixed into a first channel (L0) and a second channel (R0) of the stereo downmix signal (18), respectively, as well as object level information (OLD) of the plurality of audio signals and inter-object cross correlation information (IOC) describing similarities between pairs of audio signals of the plurality of audio signals, the method comprising:

computing, based on a first rendering prescription ($\boldsymbol{G}^{l,m}$) depending on the inter-object cross correlation information, the object level information, the downmix information, rendering information relating each audio signal to a virtual speaker position and HRTF parameters, a preliminary binaural output signal (54) from the first and second channels of the stereo downmix signal (18);

generating, from the stereo downmix signal (18), a decorrelated signal ($\boldsymbol{X}_d^{n,k}$) as an perceptual equivalent to a mono downmix (58) of the first and second channels of the stereo downmix signal (18) being, however, decorrelated to the mono downmix (58);

computing, depending on a second rendering prescription ($P_2^{l,m}$) depending on the inter-object cross correlation information, the object level information, the downmix information, the rendering information and the HRTF parameters, a corrective binaural output signal (64) from the decorrelated signal (62); and
mixing the preliminary binaural output signal (54) with the corrective binaural output signal (64) to obtain the binaural output signal (24).

11. Computer program having instructions for performing, when running on a computer, a method according to claim 10.

Patentansprüche

1. Vorrichtung zum binauralen Wiedergeben eines Mehrkanalaudiosignals (21) in ein binaurales Ausgangssignal (24), wobei das Mehrkanalaudiosignal (21) ein Stereoabwärtsmischsignal (18), in das eine Mehrzahl von Audiosignalen (14_1 - 14_N) abwärtsgemischt sind, und Nebeninformationen (20) aufweist, die (DMG, DCLD) aufweisen, die für jedes Audiosignal angeben, in welchem Ausmaß das jeweilige Audiosignal in einen ersten Kanal (L0) beziehungsweise einen zweiten Kanal (R0) des Stereoabwärtsmischsignals (18) gemischt wurde, sowie Objektpegelinformationen (OLD) der Mehrzahl von Audiosignalen und Zwischen-Objekt-Kreuzkorrelation-Informationen (IOC), die Ähnlichkeiten zwischen Paaren von Audiosignalen der Mehrzahl von Audiosignalen beschreiben, wobei die Vorrichtung dazu konfiguriert ist:

auf der Basis einer ersten Wiedergabevorschrift ($G^{l,m}$), die von den Zwischen-Objekt-Kreuzkorrelation-Informationen, den Objektpegelinformationen, den Abwärtsmischinformationen, den Wiedergabeinformationen, die jedes Audiosignal mit einer Virtueller-Lautsprecher-Position in Verbindung bringen, und HRTF-Parametern abhängt, ein vorläufiges binaurales Ausgangssignal (54) von dem ersten und dem zweiten Kanal des Stereoabwärtsmischsignals (18) zu berechnen (47);

aus dem Stereoabwärtsmischsignal (18) ein dekorreliertes Signal $(\mathbf{X}_d^{n,k})$ als Wahrnehmungsäquivalent zu einer Monoabwärtsmischung (58) des ersten und des zweiten Kanals des Stereoabwärtsmischsignals (18) zu erzeugen (50), das jedoch bezüglich der Monoabwärtsmischung (58) dekorreliert ist;
in Abhängigkeit von einer zweiten Wiedergabevorschrift ($P_2^{l,m}$), die von den Zwischen-Objekt-Kreuzkorrelation-Informationen, den Objektpegelinformationen,
den Abwärtsmischinformationen, den Wiedergabeinformationen und den HRTF-Parametern abhängt, aus dem dekorrelierten Signal (62) ein korrigierendes binaurales Ausgangssignal (64) zu berechnen (52); und
das vorläufige binaurale Ausgangssignal (54) mit dem korrigierenden binauralen Ausgangssignal (64) zu mischen (53), um das binaurale Ausgangssignal (24) zu erhalten.

2. Vorrichtung gemäß Anspruch 1, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Erzeugen des dekorrelierten Signals $(\mathbf{X}_d^{n,k})$ den ersten und den zweiten Kanal des Stereoabwärtsmischsignals (18) zu summieren und die Summe zu dekorrelieren, um das dekorrelierte Signal (62) zu erhalten.

3. Vorrichtung gemäß Anspruch 1 oder 2, die ferner dazu konfiguriert ist:

einen binauralen Zwischen-Kanal-Kohärenz-Istwert des vorläufigen binauralen Ausgangssignals (54) zu schätzen (80);
einen binauralen Zwischen-Kanal-Kohärenz-Sollwert zu bestimmen (82); und
ein Mischungsverhältnis einzustellen (84), das auf der Basis des binauralen Zwischen-Kanal-Kohärenz-Istwerts und des Zwischen-Kanal-Kohärenz-Sollwerts bestimmt, in welchem Ausmaß das binaurale Ausgangssignal (24) durch den ersten und den zweiten Kanal des Stereoabwärtsmischsignals (18), wie es durch die Berechnung (47) des vorläufigen binauralen Ausgangssignals (54) verarbeitet wird,
und durch den ersten und den zweiten Kanal des Stereoabwärtsmischsignals (18),
wie es durch die Erzeugung (50) eines dekorrelierten Signals beziehungsweise die Berechnung (52) des korrigierenden binauralen Ausgangssignals (64) verarbeitet wird, beeinflusst wird.

4. Vorrichtung gemäß Anspruch 3, wobei die Vorrichtung dazu konfiguriert ist, beim Einstellen des Mischverhältnisses das Mischverhältnis einzustellen, indem sie die erste Wiedergabevorschrift ($G^{l,m}$) und die zweite Wiedergabevorschrift ($P_2^{l,m}$) auf der Basis des Zwischen-Kanal-Kohärenz-Istwerts und des Zwischen-Kanal-Kohärenz-Sollwerts einstellt.

5. Vorrichtung gemäß Anspruch 3 oder 4, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Bestimmen des binauralen Zwischen-Kanal-Kohärenz-Istwerts die Bestimmung auf der Basis von Komponenten einer Kovarianz-Sollmatrix **F** = ***A E A**** durchzuführen, wobei "*" eine konjugierte Transponierte bezeichnet, **A** eine binaurale Wiedergabe-Sollmatrix ist, die die Audiosignale mit dem ersten beziehungsweise dem zweiten Kanal des binauralen Ausgangssignals in Bezug setzt und durch die Wiedergabeinformationen und die HRTF-Parameter eindeutig bestimmt wird, und ***E*** eine Matrix ist, die durch die Zwischen-Objekt-Kreuzkorrelation-Informationen und die Objektpegelinformationen eindeutig bestimmt wird.

6. Vorrichtung gemäß Anspruch 5, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Berechnen des vorläufigen binauralen Ausgangssignals (54) die Berechnung so durchzuführen, dass

$$\hat{X}_1 = G \cdot X$$

wobei X ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des Stereoabwärtsmischsignals (18) entsprechen, $\hat{X}_1$ ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des vorläufigen binauralen Ausgangssignal (54) entsprechen, ***G*** eine erste Wiedergabematrix ist, die die erste Wiedergabevorschrift darstellt und eine Größe von 2x2 aufweist, wobei

$$\mathbf{G} = \begin{pmatrix} P_L^1 \cos(\beta+\alpha)\exp\left(j\frac{\varphi^1}{2}\right) & P_L^2 \cos(\beta+\alpha)\exp\left(j\frac{\varphi^2}{2}\right) \\ P_R^2 \cos(\beta-\alpha)\exp\left(-j\frac{\varphi^1}{2}\right) & P_R^2 \cos(\beta-\alpha)\exp\left(-j\frac{\varphi^2}{2}\right) \end{pmatrix}$$

wobei, bei $x \in \{1,2\}$,

$$P_L^x = \sqrt{\frac{f_{11}^x}{V^x}}, \qquad P_R^x = \sqrt{\frac{f_{22}^x}{V^x}},$$

$$\varphi^x = \begin{cases} \arg\left(f_{12}^x\right) & \textit{falls eine erste Bedingung gilt} \\ 0 & \textit{andernfalls} \end{cases}$$

wobei f_{11}^*, f_{12}^* und f_{22}^* Koeffizienten von Sub-Soll-Kovarianzmatrizen $\boldsymbol{F^x}$ der Größe 2x2 bei $\boldsymbol{F^x} = \boldsymbol{A\, E^x\, A^*}$ sind, wobei $e_{ij}^x = e_{ij}\left(\frac{d_i^x}{d_i^1+d_i^2}\right)\left(\frac{d_j^x}{d_i^1+d_i^2}\right)$ Koeffizienten der NxN-Matrix $\boldsymbol{E^x}$ sind, N die Anzahl von Audiosignalen ist, e_{ij} Koeffizienten der Matrix **E** sind, die die Größe NxN aufweist, und d_i^x durch die eindeutig bestimmt werden, wobei d_i^1 das Ausmaß angibt, in dem das Audiosignal i in den ersten Kanal des Stereoabwärtsmischsignals (18) gemischt wurde, und d_i^2 definiert, in welchem Ausmaß das Audiosignal i in den zweiten Kanal des Stereoausgangssignals (18) gemischt wurde, wobei V^x ein Skalar mit $V^x = D^x E\,(D^x)^* + \varepsilon$ ist und $\boldsymbol{D^x}$ eine 1xN-Matrix ist, deren Koeffizienten d_i^x sind, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Berechnen eines korrigierenden binauralen Ausgangssignals (64) die Berechnung derart durchzuführen, dass

$$\hat{X}_2 = P_2 \cdot X_d\,,$$

wobei X_d das dekorrelierte Signal ist, $\hat{X}_2$ ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des korrigierenden binauralen Ausgangssignals (64) entsprechen, und $\boldsymbol{P_2}$ eine zweite Wiedergabematrix ist, die die zweite Wiedergabevorschrift darstellt und eine Größe 2x2 bei

$$\mathbf{P}_2 = \begin{pmatrix} P_L \sin\left(\beta + \alpha\right) \exp\left(j \frac{\arg(c_{12})}{2}\right) \\ P_R \sin\left(\beta - \alpha\right) \exp\left(-j \frac{\arg(c_{12})}{2}\right) \end{pmatrix}$$

aufweist, wobei Verstärkungen P_L und P_R als

$$P_L = \sqrt{\frac{c_{11}}{V}},\ P_R = \sqrt{\frac{c_{22}}{V}}$$

definiert sind, wobei C_{11} und C_{22} Koeffizienten einer 2x2-Kovarianzmatrix $\boldsymbol{C}$ des vorläufigen binauralen Ausgangssignals (54) mit

$$\mathbf{C} = \tilde{\mathbf{G}}\mathbf{DED}^*\tilde{\mathbf{G}}^*$$

sind, wobei V ein Skalar mit $V = \mathbf{W\ E\ W^*} + \varepsilon$ ist, $\boldsymbol{W}$ eine Monoabwärtsmischmatrix der Größe 1xN ist, deren Koeffizienten durch d_i^x, $\mathbb{D} = \begin{pmatrix} D^1 \\ D^2 \end{pmatrix}$ eindeutig bestimmt werden, und **G**

$$\tilde{\mathbf{G}}^{l,m} = \begin{pmatrix} P_L^1 \exp\left(j \frac{\varphi^1}{2}\right) & P_L^{l,m,2} \exp\left(j \frac{\varphi^2}{2}\right) \\ P_R^1 \exp\left(-j \frac{\varphi^1}{2}\right) & P_R^2 \exp\left(-j \frac{\varphi^2}{2}\right) \end{pmatrix}$$

ist, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Schätzen des Zwischen-Kanal-Kohärenz-Istwerts den Zwischen-Kanal-Kohärenz-Sollwert als

$$\rho_C = \min\left(\frac{|c_{12}|}{\sqrt{c_{11}c_{22}}}, 1\right)$$

zu bestimmen, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Bestimmen des Zwischen-Kanal-Kohärenz-Istwerts den Zwischen-Kanal-Kohärenz-Sollwert als

$$\rho_T = \min\left(\frac{|f_{12}|}{\sqrt{f_{11}fl_{22}}}, 1\right)$$

zu bestimmen, und wobei die Vorrichtung ferner dazu konfiguriert ist, beim Einstellen des Mischverhältnisses Rotatorwinkel α und β gemäß

$$\alpha = \frac{1}{2}\left(\arccos\left(\rho_T\right) - \arccos\left(\rho_C\right)\right),$$

$$\beta = \arctan\left(\tan(\alpha)\frac{P_R - P_L}{P_L + P_R}\right)$$

zu bestimmen, wobei ε eine kleine Konstante zum Vermeiden von jeweiligen Divisionen durch null bezeichnet.

7. Vorrichtung gemäß Anspruch 1, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Berechnen des vorläufigen binauralen Ausgangssignals (54) die Berechnung so durchzuführen, dass

$$\hat{X}_1 = \boldsymbol{G} \cdot \boldsymbol{X}$$

wobei ***X*** ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des Stereoabwärtsmischsignals (18) entsprechen, $\hat{\mathbf{X}}_1$ ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des vorläufigen binauralen Ausgangssignals (54) entsprechen, ***G*** eine erste Wiedergabematrix ist, die die erste Wiedergabevorschrift darstellt und eine Größe von 2x2 aufweist, wobei

$$\boldsymbol{G} = \quad \boldsymbol{AED}^*(\boldsymbol{DED}^*)^{-1},$$

wobei ***E*** eine Matrix ist, die durch die Zwischen-Objekt-Kreuzkorrelation-Informationen und die Objektpegelinformationen eindeutig bestimmt wird;
D eine 2xN-Matrix ist, deren Koeffizienten d_{ij} durch die Abwärtsmischinformationen eindeutig bestimmt werden, wobei $d_{1,j}$ das Ausmaß angibt, in dem das Audiosignal j in den ersten Kanal des Stereoabwärtsmischsignals (18) gemischt wurde, und $d_{2,j}$ definiert, in welchem Ausmaß das Audiosignal j in den zweiten Kanal des Stereoausgangssignals (18) gemischt wurde;
A eine binaurale Wiedergabe-Sollmatrix ist, die die Audiosignale mit dem ersten beziehungsweise dem zweiten Kanal des binauralen Ausgangssignals in Bezug setzt und durch die Wiedergabeinformationen und die HRTF-Parameter eindeutig bestimmt wird,
wobei die Vorrichtung ferner dazu konfiguriert ist, beim Berechnen eines korrigierenden binauralen Ausgangssignals (64) die Berechnung derart durchzuführen, dass

$$\hat{X}_2 = \boldsymbol{P} \cdot \boldsymbol{X}_d$$

wobei $\boldsymbol{X_d}$ das dekorrelierte Signal ist, $\hat{\boldsymbol{X}}_2$ ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des korrigierenden binauralen Ausgangssignals (64) entsprechen, und ***P*** eine zweite Wiedergabematrix ist, die die zweite Wiedergabevorschrift darstellt und eine Größe 2x2 aufweist und derart bestimmt wird, dass PP* = Δ**R**, wobei $\Delta\mathbf{R} = AEA^* - \mathbf{G}_0 DED^* \mathbf{G}_0^*$, wobei $\mathbf{G_0} = \mathbf{G}$.

8. Vorrichtung gemäß Anspruch 1, wobei die Vorrichtung ferner dazu konfiguriert ist, beim Berechnen des vorläufigen binauralen Ausgangssignals (54) die Berechnung so durchzuführen, dass

$$\hat{X}_1 = \boldsymbol{G} \cdot \boldsymbol{X}$$

wobei ***X*** ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des Stereoabwärtsmischsignals (18) entsprechen, $\hat{\mathbf{X}}_1$ ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des vorläufigen binauralen Ausgangssignals (54) entsprechen, ***G*** eine erste Wiedergabematrix ist, die die erste Wiedergabevorschrift darstellt und eine Größe von 2x2 aufweist, wobei

$$\boldsymbol{G} = (\mathrm{G_0}\mathbf{DED}^*\mathrm{G_0}^*)^{-1}(\mathrm{G_0}\,\mathbf{DED}^*\mathrm{G_0}^*\;\mathbf{AEA}^*\;\mathrm{G_0}\,\mathbf{DED}^*\mathrm{G_0}^*)^{1/2}(\mathrm{G_0}\,\mathbf{DED}^*\mathrm{G_0}^*)^{-1}\,\mathrm{G_0}$$
$$\text{wobei } \mathrm{G_0} = \mathbf{AED}^*\,(\mathbf{DED}^*)^{-1}$$

wobei ***E*** eine Matrix ist, die durch die Zwischen-Objekt-Kreuzkorrelation-Informationen und die Objektpegelinfor-

mationen eindeutig bestimmt wird;
D eine 2xN-Matrix ist, deren Koeffizienten d_{ij} durch die Abwärtsmischinformationen eindeutig bestimmt werden, wobei $d_{1,j}$ das Ausmaß angibt, in dem das Audiosignal j in den ersten Kanal des Stereoabwärtsmischsignals (18) gemischt wurde, und $d_{2,j}$ definiert, in welchem Ausmaß das Audiosignal j in den zweiten Kanal des Stereoausgangssignals (18) gemischt wurde;
A eine binaurale Wiedergabe-Sollmatrix ist, die die Audiosignale mit dem ersten beziehungsweise dem zweiten Kanal des binauralen Ausgangssignals in Bezug setzt und durch die Wiedergabeinformationen und die HRTF-Parameter eindeutig bestimmt wird,
wobei die Vorrichtung ferner dazu konfiguriert ist, beim Berechnen eines korrigierenden binauralen Ausgangssignals (64) die Berechnung derart durchzuführen, dass

$$\hat{\boldsymbol{X}}_2 = \boldsymbol{P} \cdot \boldsymbol{X}_d$$

wobei $\boldsymbol{X_d}$ das dekorrelierte Signal ist, $\hat{\mathbf{X}}_\mathbf{2}$ ein 2x1-Vektor ist, dessen Komponenten dem ersten und dem zweiten Kanal des korrigierenden binauralen Ausgangssignals (64) entsprechen, und ***P*** eine zweite Wiedergabematrix ist, die die erste Wiedergabevorschrift darstellt und eine Größe 2x2 aufweist und derart bestimmt wird, dass **PP* = (AEA* - GDED*G*)** / *V*, wobei *V* ein Skalar ist.

9. Vorrichtung gemäß einem der vorhergehenden Ansprüche, bei der die (DMG, DCLD) zeitabhängig sind und die Objektpegelinformationen (OLD) und die Zwischen-Objekt-Kreuzkorrelation-Informationen (IOC) zeit- und frequenzabhängig sind.

10. Verfahren zum binauralen Wiedergeben eines Mehrkanalaudiosignals (21) in ein binaurales Ausgangssignal (24), wobei das Mehrkanalaudiosignal (21) ein Stereoabwärtsmischsignal (18), in das eine Mehrzahl von Audiosignalen (14_1 - 14_N) abwärtsgemischt sind, und Nebeninformationen (20) aufweist, die (DMG, DCLD) aufweisen, die für jedes Audiosignal angeben, in welchem Ausmaß das jeweilige Audiosignal in einen ersten Kanal (L0) beziehungsweise einen zweiten Kanal (R0) des Stereoabwärtsmischsignals (18) gemischt wurde, sowie Objektpegelinformationen (OLD) der Mehrzahl von Audiosignalen und Zwischen-Objekt-Kreuzkorrelation-Informationen (IOC), die Ähnlichkeiten zwischen Paaren von Audiosignalen der Mehrzahl von Audiosignalen beschreiben, wobei das Verfahren folgende Schritte aufweist:

Berechnen, auf der Basis einer ersten Wiedergabevorschrift ($\boldsymbol{G}^{l,m}$), die von den Zwischen-Objekt-Kreuzkorrelation-Informationen, den Objektpegelinformationen, den Abwärtsmischinformationen, den Wiedergabeinformationen, die jedes Audiosignal mit einer Virtueller-Lautsprecher-Position in Verbindung bringen, und HRTF-Parametern abhängt, eines vorläufigen binauralen Ausgangssignals (54) von dem ersten und dem zweiten Kanal des Stereoabwärtsmischsignals (18);

Erzeugen, aus dem Stereoabwärtsmischsignal (18), eines dekorrelierten Signals ($\mathbf{X}_d^{n,k}$) als Wahrnehmungsäquivalent zu einer Monoabwärtsmischung (58) des ersten und des zweiten Kanals des Stereoabwärtsmischsignals (18), das jedoch bezüglich der Monoabwärtsmischung (58) dekorreliert ist;

Berechnen, in Abhängigkeit von einer zweiten Wiedergabevorschrift ($\boldsymbol{P}_2^{l,m}$), die von den Zwischen-Objekt-Kreuzkorrelation-Informationen, den Objektpegelinformationen, den Abwärtsmischinformationen, den Wiedergabeinformationen und den HRTF-Parametern abhängt, eines korrigierenden binauralen Ausgangssignals (64) aus dem dekorrelierten Signal (62); und
Mischen des vorläufigen binauralen Ausgangssignals (54) mit dem korrigierenden binauralen Ausgangssignal (64), um das binaurale Ausgangssignal (24) zu erhalten.

11. Computerprogramm, das Anweisungen zum Durchführen, wenn es auf einem Computer läuft, eines Verfahrens gemäß Anspruch 10 aufweist.

Revendications

1. Appareil pour rendu binaural d'un signal audio multicanal (21) en un signal de sortie binaural (24), le signal audio multicanal (21) comprenant un signal de mélange descendant stéréo (18) dans lequel sont mélangés vers le bas une pluralité de signaux audio (14_1 à 14_N), et des informations latérales (20) comprenant une information de mélange descendant (DMG, DCLD) indiquant, pour chaque signal audio, la mesure dans laquelle le signal audio respectif a été mélangé dans respectivement un premier canal (L0) et un deuxième canal (R0) du signal de mélange descendant stéréo (18), ainsi que des informations de niveau d'objet (OLD) de la pluralité de signaux audio et des informations de corrélation croisée entre objets (IOC) décrivant les similitudes entre paires de signaux audio de la pluralité de signaux audio, l'appareil étant configuré pour:

calculer (47), sur base d'une première prescription de rendu ($G^{l,m}$) qui dépend des informations de corrélation croisée entre objets, des informations de niveau d'objet, des informations de mélange descendant, des informations de rendu mettant en rapport chaque signal audio avec une position de haut-parleur virtuel, et des paramètres HRTF, un signal de sortie binaural préliminaire (54) des premier et deuxième canaux du signal de mélange descendant stéréo (18);

générer (50), à partir du signal de mélange descendant stéréo (18), un signal décorrélé $\left(X_d^{n,k}\right)$ comme un équivalent perceptuel à un mélange descendant mono (58) des premier et deuxième canaux du signal de mélange descendant stéréo (18) qui est toutefois décorrélé au mélange descendant mono (58);
calculer (52), on fonction d'une deuxième prescription de rendu ($P_2^{l,m}$) qui dépend des informations de corrélation croisée entre objets, des informations de niveau d'objet, des informations de mélange descendant, des informations de rendu, et des paramètres HRTF, un signal de sortie binaural de correction (64) à partir du signal décorrélé (62); et
mélanger (53) le signal de sortie binaural préliminaire (54) avec le signal de sortie binaural de correction (64), pour obtenir le signal de sortie binaural (24).

2. Appareil selon la revendication 1, dans lequel l'appareil est par ailleurs configuré pour additionner, lors de la génération du signal décorrélé $\left(X_d^{n,k}\right)$, le premier et le deuxième canal du signal de mélange descendant stéréo (18) et décorréler la somme, pour obtenir le signal décorrélé (62).

3. Appareil selon la revendication 1 ou 2, configuré par ailleurs pour:

estimer (80) une valeur de cohérence entre canaux binaurale réelle du signal de sortie binaural préliminaire (54);
déterminer (82) une valeur de cohérence entre canaux binaurale cible; et
régler (84) un rapport de mélange déterminant la mesure dans laquelle le signal de sortie binaural (24) est influencé par les premier et deuxième canaux du signal de mélange descendant stéréo (18) traité respectivement par le calcul (47) du signal de sortie binaural préliminaire (54) et les premier et deuxième canaux du signal de mélange descendant stéréo (18) traité par la génération (50) d'un signal décorrélé et le calcul (52) du signal de sortie binaural de correction (64), sur base de la valeur de cohérence entre canaux binaurale réelle et de la valeur de cohérence entre canaux binaurale cible.

4. Appareil selon la revendication 3, dans lequel l'appareil est par ailleurs configuré pour régler, lors du réglage du rapport de mélange, le rapport de mélange en réglant la première prescription de rendu ($G^{l,m}$) et la deuxième prescription de rendu ($P_2^{l,m}$) sur base de la valeur de cohérence entre canaux binaurale réelle et de la valeur de cohérence entre canaux binaurale cible.

5. Appareil selon la revendication 3 ou 4, dans lequel l'appareil est par ailleurs configuré pour effectuer, lors de la détermination de la valeur de cohérence entre canaux binaurale cible, la détermination sur base de composantes d'une matrice de covariance cible $\boldsymbol{F} = \boldsymbol{A}\,\boldsymbol{E}\,\boldsymbol{A}^*$, "*" désignant la transposée conjuguée, $\boldsymbol{A}$ étant une matrice de rendu binaurale cible mettant en rapport les signaux audio avec respectivement les premier et deuxième canaux du signal de sortie binaural, et étant déterminée de manière unique par les informations de rendu et les paramètres HRTF, et $\boldsymbol{E}$ étant une matrice déterminée de façon unique par les informations de corrélation croisée entre objets et les informations de niveau d'objet.

6. Appareil selon la revendication 5, dans lequel l'appareil est par ailleurs configuré pour effectuer le calcul, lors du calcul du signal de sortie binaural préliminaire (54), de sorte que

$$\hat{X}_1 = G \cdot X$$

où X est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de mélange descendant stéréo (18), $\hat{X}_1$ est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de sortie binaural préliminaire (54), ***G*** est une première matrice de rendu qui représente la première prescription de rendu et présentant une grandeur de 2x2 avec

$$G = \begin{pmatrix} P_L^1 cos(\beta + a) exp\left(j\frac{\phi^1}{2}\right) & P_L^2 cos(\beta + a) exp\left(j\frac{\phi^2}{2}\right) \\ P_R^2 cos(\beta - a) exp\left(-j\frac{\phi^1}{2}\right) & P_R^2 cos(\beta - a) exp\left(-j\frac{\phi^2}{2}\right) \end{pmatrix}$$

où, avec x ∈ {1,2},

$$P_L^x = \sqrt{\frac{f_{11}^x}{V^x}}, \qquad P_R^x = \sqrt{\frac{f_{22}^x}{V^x}},$$

$$\phi^x = \begin{cases} arg(f_{12}^x) & si\ une\ première\ condition\ s'applique \\ 0 & autrememt \end{cases}$$

où f_{11}^x, f_{12}^x et f_{22}^x sont des coefficients de matrice de covariance sous-cible ***F****** de grandeur 2x2, où $F^x = AE^xA^*$,

où $e_{ij}^x = e_{ij}\left(\frac{d_i^x}{d_i^1+d_i^2}\right)\left(\frac{d_j^x}{d_i^1+d_i^2}\right)$ sont des coefficients de matrice NxN E^x, N étant le nombre de signaux audio, e_{ij} sont des coefficients de la matrice ***E*** de grandeur NxN, et d_i^x sont déterminés de manière unique par les informations de mélange descendant, où d_i^1 indique la mesure dans laquelle le signal audio i a été mélangé dans le premier canal du signal de mélange descendant stéréo (18) et d_i^2 définit la mesure dans laquelle le signal audio i a été mélangé dans le deuxième canal du signal de sortie stéréo (18),

où V^x est une mesure scalaire, où $V^x = D^xE(D^x)^* + \varepsilon$ et D^x est une matrice 1xN dont les coefficients sont d_i^x, dans lequel l'appareil est par ailleurs configuré pour effectuer le calcul, lors du calcul d'un signal de sortie binaural de correction (64), de sorte que

$$\hat{X}_2 = P_2 \cdot X_d$$

où X_d est le signal décorrélé, $\hat{X}_2$ est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de sortie binaural de correction (64), et $\boldsymbol{P}_2$ est une deuxième matrice de rendu qui représente la deuxième prescription de rendu et présente une grandeur de 2x2, avec

$$P_2 = \begin{pmatrix} P_L sin(\beta + a) exp\left(j\frac{arg(c_{12})}{2}\right) \\ P_R sin(\beta - a) exp\left(-j\frac{arg(c_{12})}{2}\right) \end{pmatrix}$$

où les gains P_L et P_R sont définis comme

$$P_L = \sqrt{\frac{c_{11}}{V}}, P_R = \sqrt{\frac{c_{22}}{V}}$$

où c_{11} et c_{22} sont des coefficients d'une matrice ***C*** de covariance 2x2 du signal de sortie binaural préliminaire (54), avec

$$\boldsymbol{C} = \boldsymbol{\widetilde{G}DED^{*}\widetilde{G}^{*}}$$

où V est une mesure scalaire, où $V = WEW^{*} + \varepsilon$, ***W*** est une matrice de mélange descendant mono de grandeur 1xN dont les coefficients sont déterminés de manière unique par d_i^x, $D = \begin{pmatrix} D^1 \\ D^2 \end{pmatrix}$ et $\boldsymbol{\tilde{G}}$ est

$$\tilde{G}^{l,m} = \begin{pmatrix} P_L^1 exp\left(j\frac{\phi^1}{2}\right) & P_L^{l,m,2} exp\left(j\frac{\phi^2}{2}\right) \\ P_R^1 exp\left(-j\frac{\phi^1}{2}\right) & P_R^2 exp\left(-j\frac{\phi^2}{2}\right) \end{pmatrix},$$

dans lequel l'appareil est par ailleurs configuré pour déterminer, lors de l'estimation de la valeur de cohérence entre canaux binaurale réelle, la valeur de cohérence entre canaux binaurale réelle comme

$$\rho_C = min\left(\frac{|C_{12}|}{\sqrt{C_{11}C_{22}}}, 1\right)$$

dans lequel l'appareil est par ailleurs configuré pour déterminer, lors de la détermination de la valeur de cohérence entre canaux binaurale cible, la valeur de cohérence entre canaux binaurale cible comme

$$\rho_T = min\left(\frac{|f_{12}|}{\sqrt{f_{11}fl_{22}}}, 1\right),$$

et
dans lequel l'appareil est par ailleurs configuré pour déterminer, lors du réglage du rapport de mélange, les angles de rotateur α et β selon

$$\alpha = \frac{1}{2}\left(arcos(\rho_T) - arcos(\rho_C)\right),$$

$$\beta = arctan\left(tan(\alpha)\frac{P_R - P_L}{P_L + P_R}\right),$$

ε désignant une petite constante pour éviter les divisions par zéro, respectivement.

7. Appareil selon la revendication 1, dans lequel l'appareil est par ailleurs configuré pour effectuer le calcul, lors du calcul du signal de sortie binaural préliminaire (54), de sorte que

$$\boldsymbol{\hat{X}_1 = G \cdot X}$$

où ***X*** est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de mélange descendant stéréo (18), $\boldsymbol{\hat{X}}_1$ est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de sortie binaural préliminaire (54), ***G*** est une première matrice de rendu représentant la première prescription de rendu et présentant une grandeur de 2x2, avec

$$\boldsymbol{G = AED^{*}(DED^{*})^{-1}},$$

où ***E*** est une matrice déterminée de manière unique par les informations de corrélation croisée entre objets et les informations de niveau d'objet;
D est une matrice 2xN dont les coefficients d_{ij} sont déterminés de manière unique les informations de mélange descendant, où d_{1j} indique la mesure dans laquelle le signal audio j a été mélangé dans le premier canal du signal de mélange descendant stéréo (18) et d_{2j} définit la mesure dans laquelle le signal audio j a été mélangé dans le deuxième canal du signal de sortie stéréo (18);
A est une matrice de rendu binaurale cible mettant en rapport les signaux audio avec respectivement les premier et deuxième canaux du signal de sortie binaural, et est déterminée de manière unique par les informations de rendu et les paramètres HRTF,
dans lequel l'appareil est par ailleurs configuré pour effectuer le calcul, lors du calcul d'un signal de sortie binaural de correction (64), de sorte que

$$\hat{X}_2 = P \cdot X_d$$

où X_d est le signal décorrélé, $\hat{X}_2$ est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de sortie binaural de correction (64), et ***P*** est une deuxième matrice de rendu qui représente la deuxième prescription de rendu et présentant une grandeur de 2x2 et qui est déterminée de sorte que PP* = $\Delta\boldsymbol{R}$, où $\Delta\boldsymbol{R} = AEA^* - \boldsymbol{G}_0 DED^* \boldsymbol{G}_0^*$, où $\boldsymbol{G}_0 = \boldsymbol{G}$.

8. Appareil selon la revendication 1, dans lequel l'appareil est par ailleurs configuré pour effectuer le calcul, lors du calcul du signal de sortie binaural préliminaire (54), de sorte que

$$\hat{X}_1 = G \cdot X$$

où ***X*** est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de mélange descendant stéréo (18), $\hat{X}_1$ est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de sortie binaural préliminaire (54), ***G*** est une première matrice de rendu qui représente la première prescription de rendu et présente une grandeur de 2x2, avec

$$G = (G_0\, DED^* G_0^*)^{-1} (G_0\, DED^* G_0^*\, AEA^*\, G_0\, DED^* G_0^*)^{1/2} (G_0\, DED^* G_0^*)^{-1}$$
$$G_0 \text{ avec } G_0 = AED^*\, (DED^*)^{-1}$$

où ***E*** est une matrice déterminée de manière unique par les informations de corrélation croisée entre objets et les informations de niveau d'objet;
D est une matrice 2xN dont les coefficients d_{ij} sont déterminés de manière unique par les informations de mélange descendant, où d_{1j} indique la mesure dans laquelle le signal audio j a été mélangé dans le premier canal du signal de mélange descendant stéréo (18) et d_{2j} définit la mesure dans laquelle le signal audio j a été mélangé dans le deuxième canal du signal de sortie stéréo (18);
A est une matrice de rendu binaurale cible mettant en rapport les signaux audio pour respectivement les premier et deuxième canaux du signal de sortie binaural, et est déterminée de manière unique par les informations de rendu et les paramètres HRTF,
dans lequel l'appareil est par ailleurs configuré pour effectuer le calcul, lors du calcul d'un signal de sortie binaural de correction (64), de sorte que

$$\hat{X}_2 = P \cdot X_d$$

où $\boldsymbol{X_d}$ est le signal décorrélé, $\hat{\boldsymbol{X}}_2$ est un vecteur 2x1 dont les composantes correspondent aux premier et deuxième canaux du signal de sortie binaural de correction (64), et ***P*** est une deuxième matrice de rendu qui représente la deuxième prescription de rendu et qui présente une grandeur de 2x2 et est déterminée de sorte que **PP* = (AEA* -GDED*G*)**/ *V*, où *V* est une mesure scalaire.

9. Appareil selon l'une quelconque des revendications précédentes, dans lequel les informations de mélange descendant (DMG, DCLD) est fonction du temps, et les informations de niveau d'objet (OLD) et les informations de corrélation croisée entre objets (IOC) sont fonction du temps et de la fréquence.

10. Procédé pour le rendu binaural d'un signal audio multicanal (21) en un signal de sortie binaural (24), le signal audio multicanal (21) comprenant un signal de mélange descendant stéréo (18) dans lequel sont mélangés vers le bas une pluralité de signaux audio (14_1 à 14_N), et des informations latérales (20) comprenant une information de mélange descendant (DMG, DCLD) indiquant, pour chaque signal audio, la mesure dans laquelle le signal audio respectif a été mélangé dans respectivement un premier canal (L0) et un deuxième canal (R0) du signal de mélange descendant stéréo (18), ainsi que des informations de niveau d'objet (OLD) de la pluralité de signaux audio et des informations de corrélation croisée entre objets (IOC) décrivant les similitudes entre paires de signaux audio de la pluralité de signaux audio, le procédé comprenant le fait de:

calculer, sur base d'une première prescription de rendu ($G^{l,m}$) qui dépend des informations de corrélation croisée entre objets, des informations de niveau d'objet, des informations de mélange descendant, des informations de rendu mettant en rapport chaque signal audio avec une position de haut-parleur virtuel, et des paramètres HRTF, un signal de sortie binaural préliminaire (54) des premier et deuxième canaux du signal de mélange descendant stéréo (18);

générer, à partir du signal de mélange descendant stéréo (18), un signal décorrélé $\left(X_d^{n,k}\right)$ comme un équivalent perceptuel à un mélange descendant mono (58) des premier et deuxième canaux du signal de mélange descendant stéréo (18) qui est toutefois décorrélé au mélange descendant mono (58);
calculer, en fonction d'une deuxième prescription de rendu ($P_2^{l,m}$) qui dépend des informations de corrélation croisée entre objets, des informations de niveau d'objet, des informations de mélange descendant, des informations de rendu, et des paramètres HRTF, un signal de sortie binaural de correction (64) à partir du signal décorrélé (62); et
mélanger le signal de sortie binaural préliminaire (54) avec le signal de sortie binaural de correction (64), pour obtenir le signal de sortie binaural (24).

11. Programme d'ordinateur présentant des instructions pour réaliser, lorsqu'il est exécuté sur un ordinateur, un procédé selon la revendication 10.

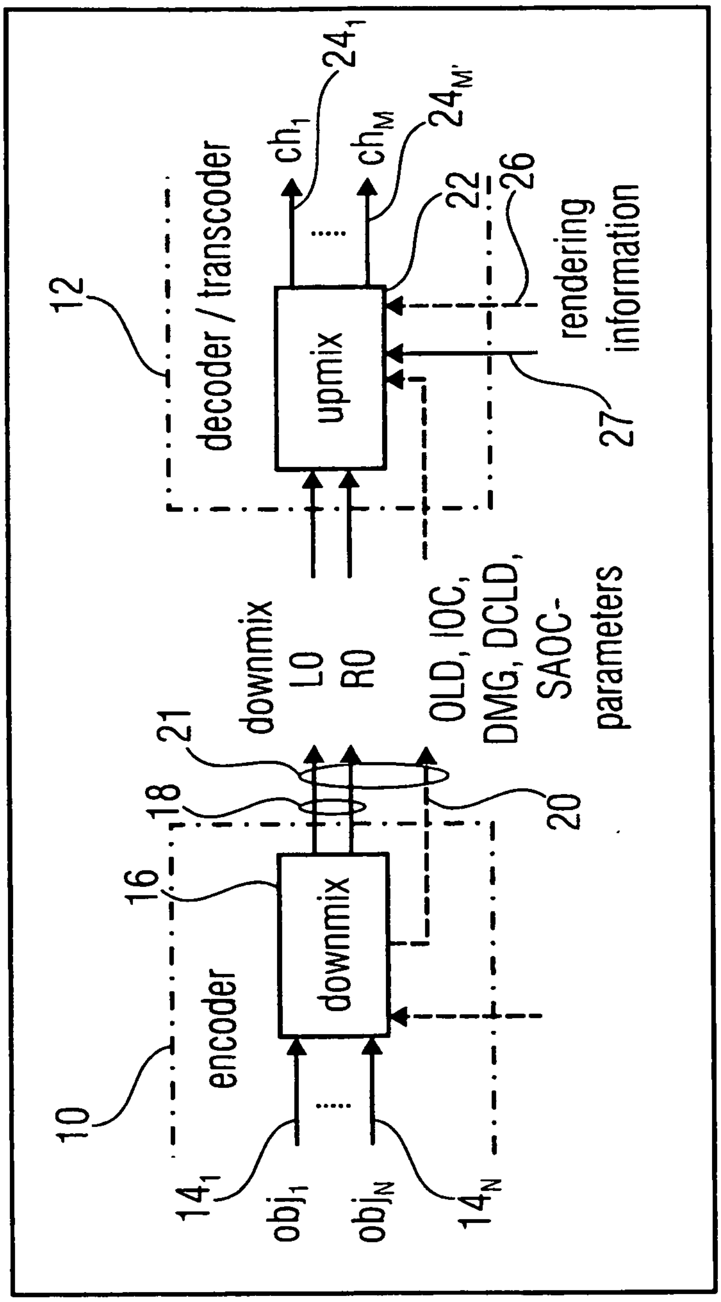

10
encoder
16
downmix
18
21
20
downmix
L0
R0
OLD, IOC,
DMG, DCLD,
SAOC-
parameters
12
decoder / transcoder
upmix
22
26
27
rendering
information
14_1
14_N
obj_1
obj_N
ch_1
ch_M
24_1
$24_{M'}$

FIG 1

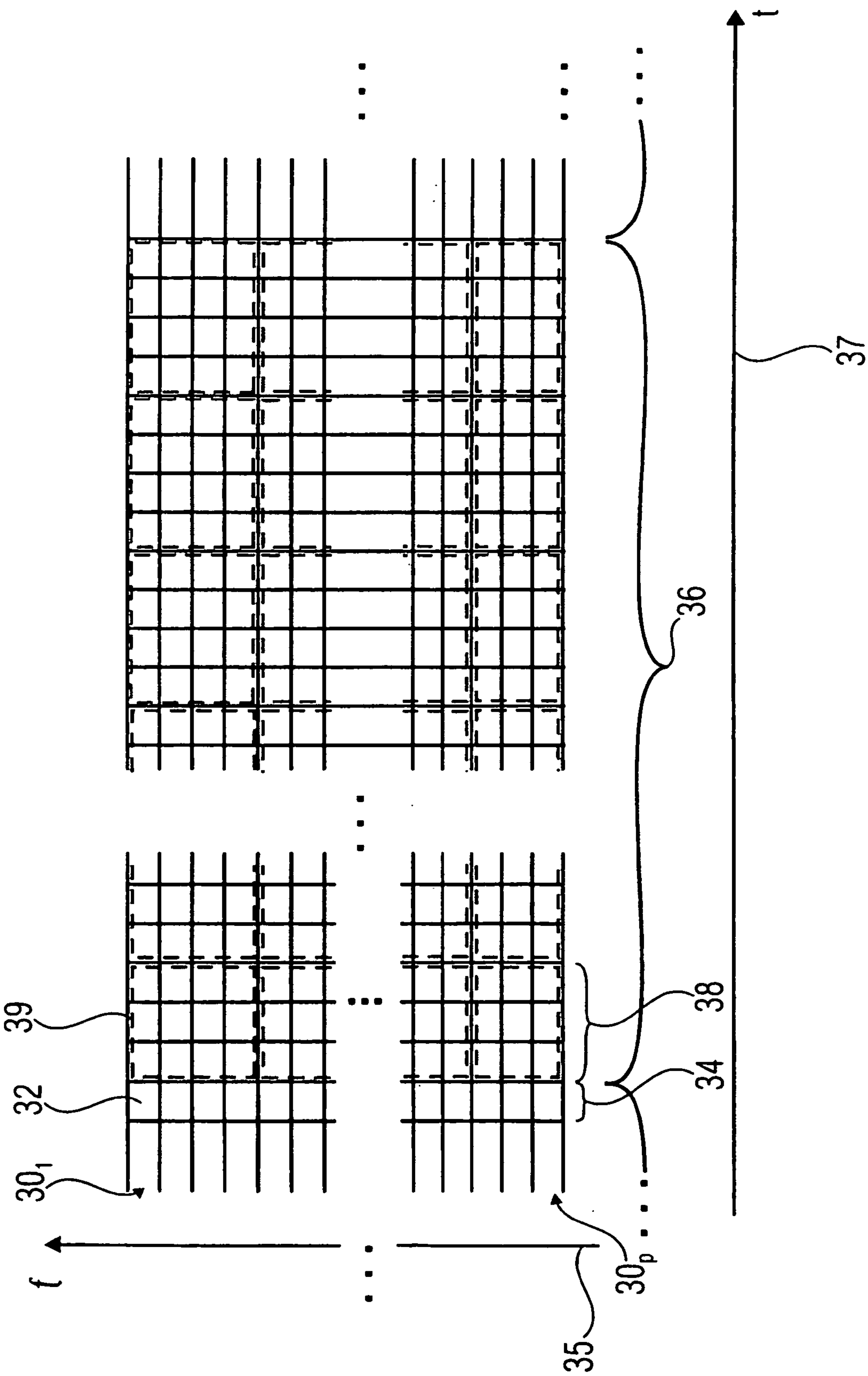


FIG 2

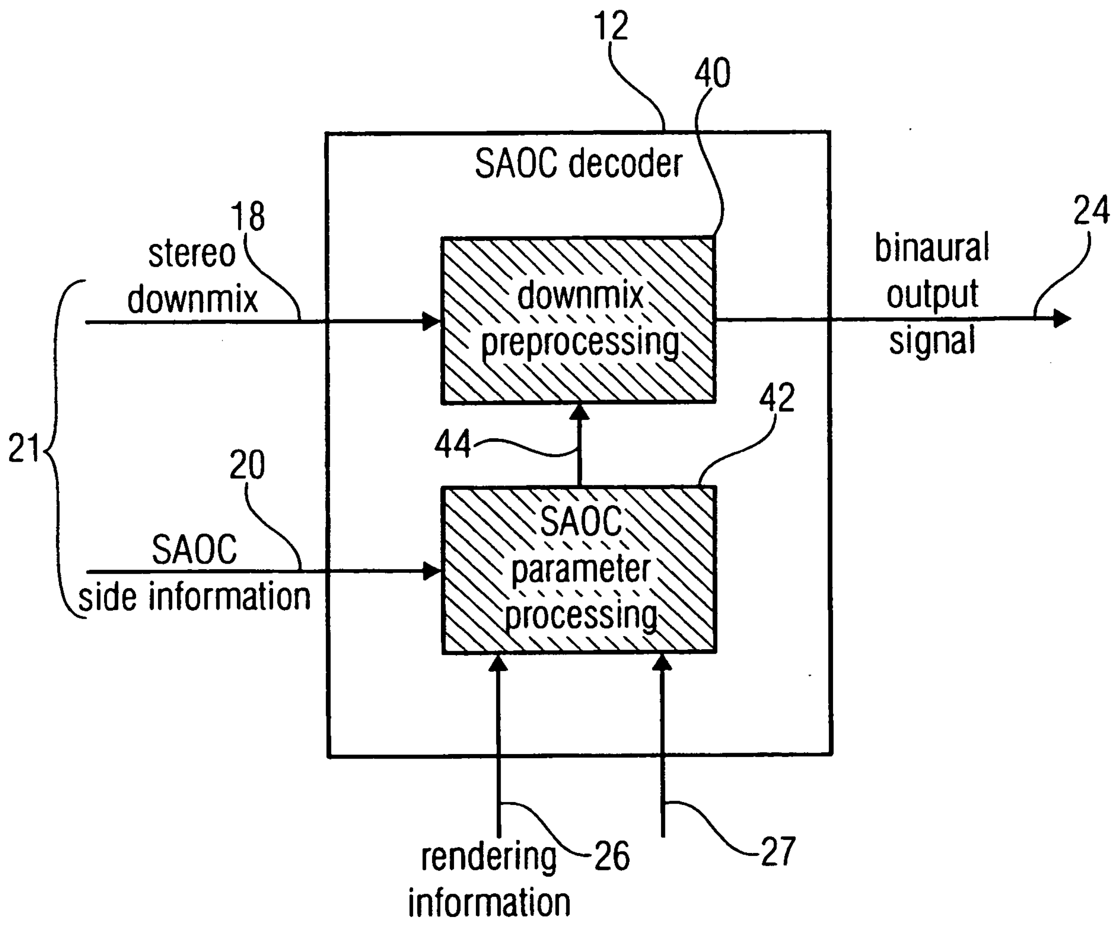

12
40
SAOC decoder
18
stereo
downmix
downmix
preprocessing
binaural
output
signal
24
21
44
42
20
SAOC
side information
SAOC
parameter
processing
rendering
information
26
27

FIG 3

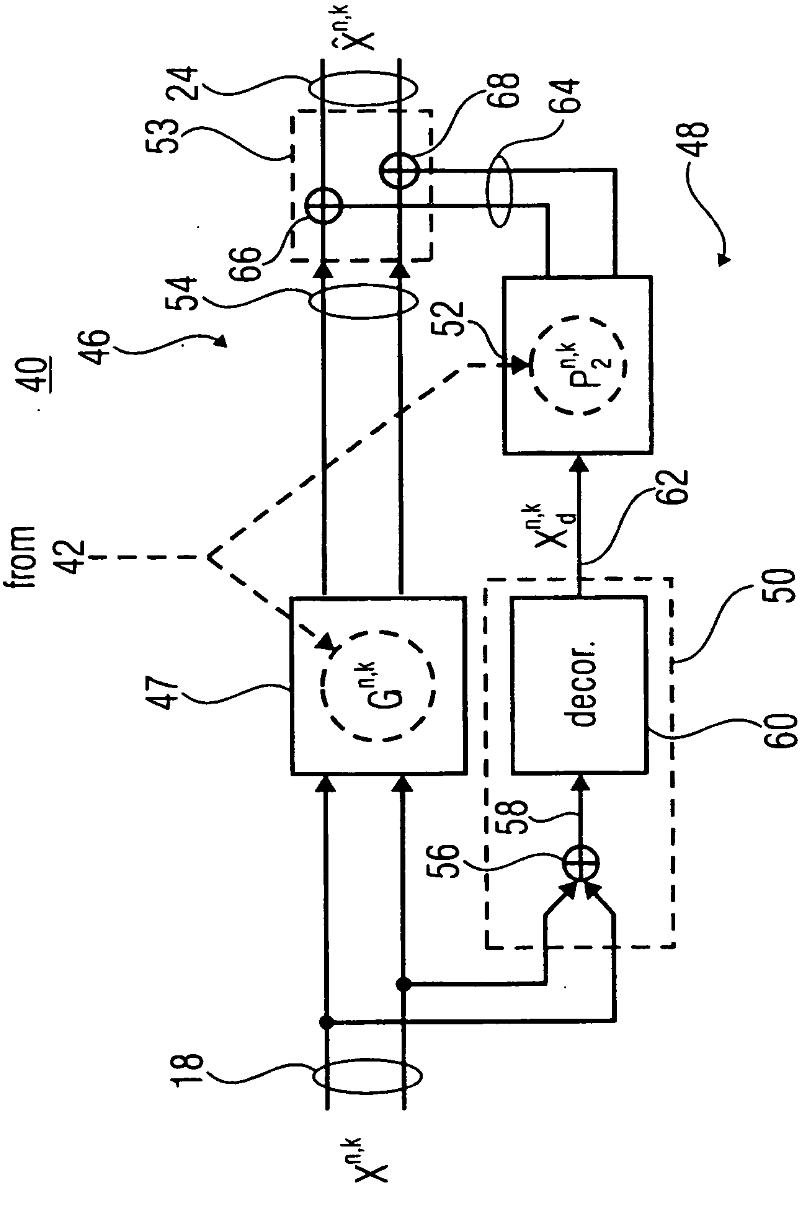


FIG 4

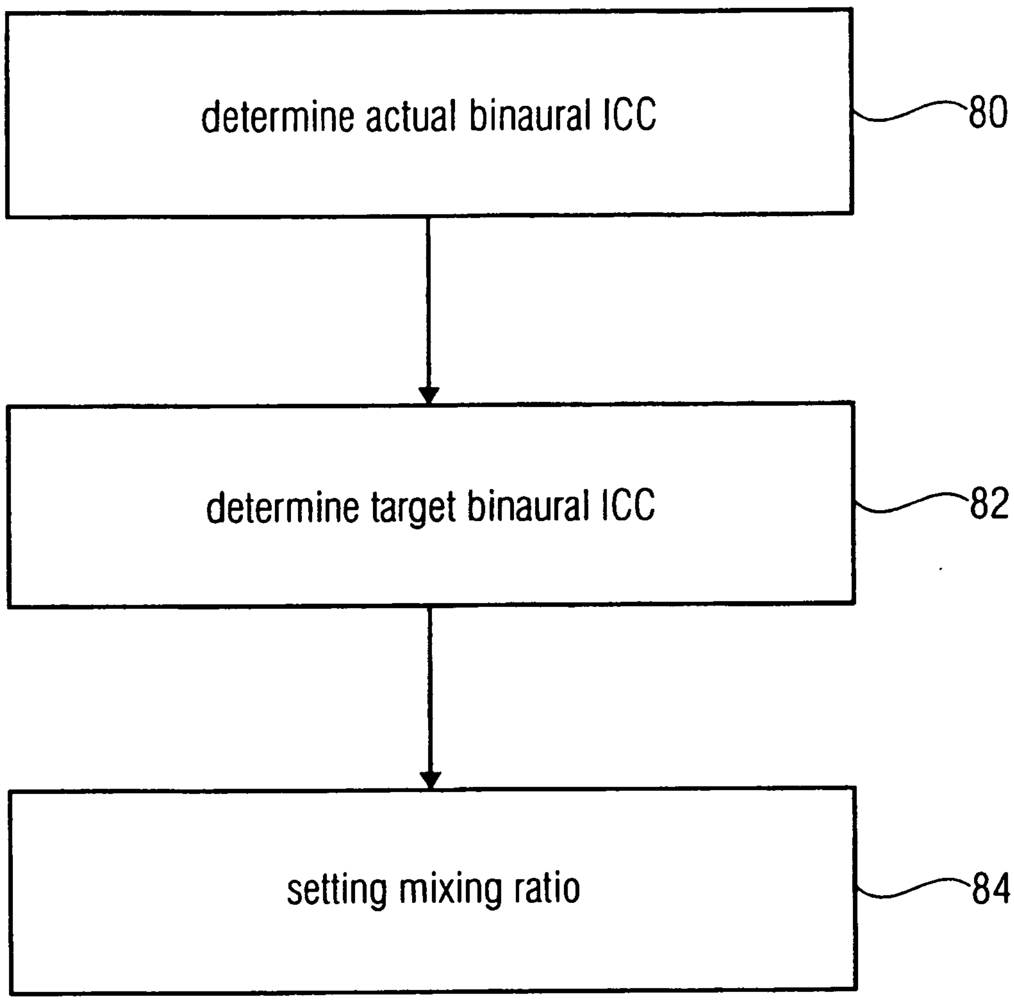
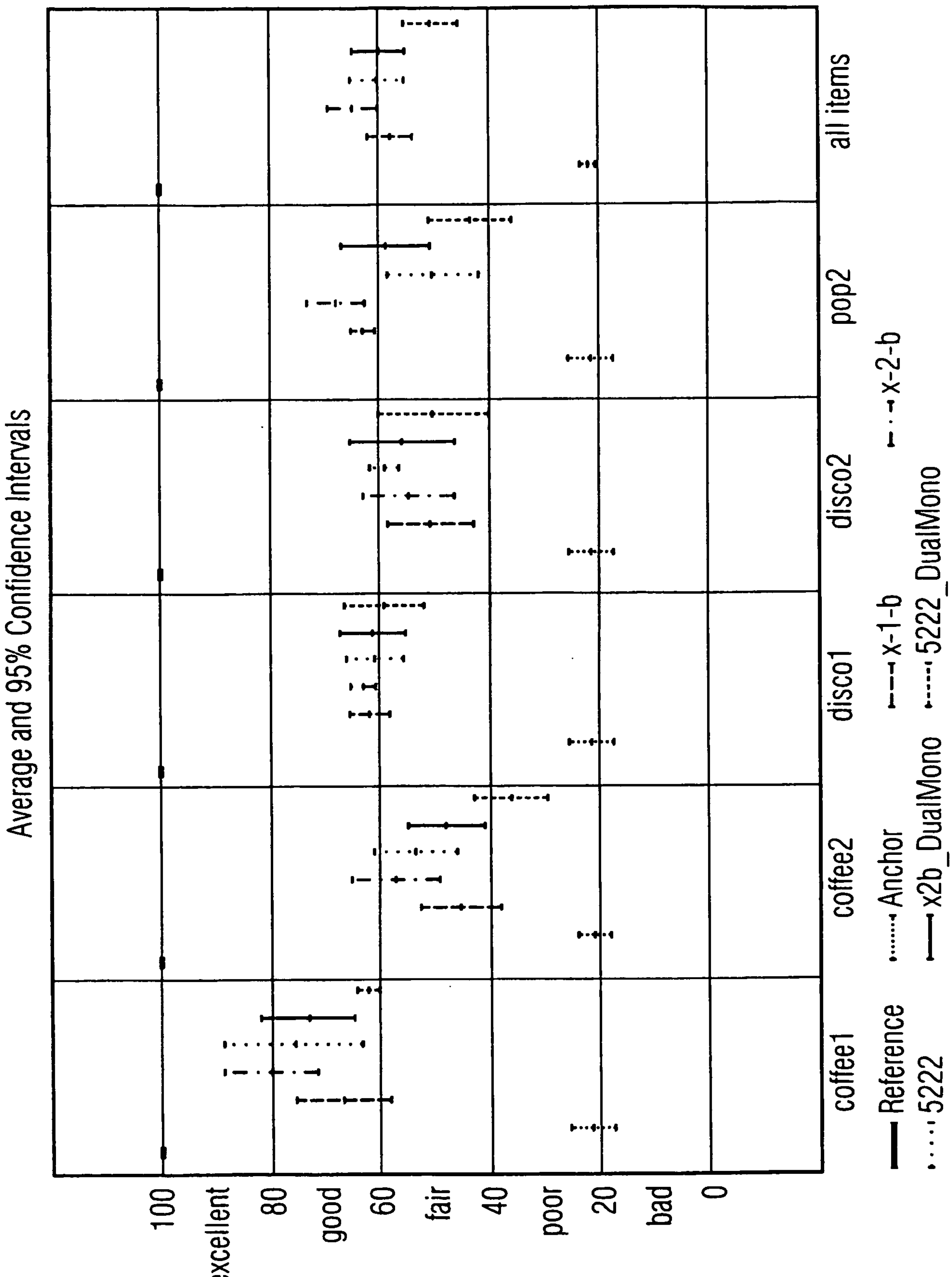

determine actual binaural ICC
80
determine target binaural ICC
82
setting mixing ratio
84

FIG 5

FIG 6


Average and 95% Confidence Intervals
100
excellent
80
good
60
fair
40
poor
20
bad
0
coffee1
coffee2
disco1
disco2
pop2
all items
Reference
Anchor
x-1-b
x-2-b
5222
x2b_DualMono
5222_DualMono

REFERENCES CITED IN THE DESCRIPTION

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Patent documents cited in the description

- WO 2008069593 A1 **[0011]**
- WO 2007078254 A2 **[0012]**
- WO 2007083952 A1 **[0013]**

Non-patent literature cited in the description

- **ENGDEGARD et al.** *Spatial Audio Object Coding (SAOC) - The Upcoming MPEG Standard on Parametric Object Based Coding, 124th AES convention, paper 7377,* 17 May 2008, 1-15 **[0010]**
- ISO/IEC CD 23003-2:200x Spatial Audio Object Coding (SAOC). *ISO/IEC JTC 1/SC 29/WG 11 (MPEG), Document N10045,* July 2008 **[0111] [0123]**
- ISO/IEC CD 23003-2:200x Spatial Audio Object Coding (SAOC). *Spatial Audio Object Coding (SAOC) ISO/IEC JTC 1/SC 29/WG 11 (MPEG), Document N10045,* July 2008 **[0116]**
- MUSHRA-EBU Method for Subjective Listening Tests of Intermediate Audio Quality. **EBU TECHNICAL RECOMMENDATION.** Doc. B/AIM022. October 1999 **[0123]**
- Information technology - MPEG audio technologies - Part 1: MPEG Surround. *ISO/IEC 23003-1:2007,* 2007 **[0123]**
- Final Spatial Audio Object Coding Evaluation Procedures and Criterion. *ISO/IEC JTC1/SC29/WG11 (MPEG), Document N9099,* April 2007 **[0123]**
- **JEROEN ; BREEBAART ; CHRISTOF FALLER.** Spatial Audio Processing. MPEG Surround and Other Applications. Wiley & Sons, 2007 **[0123]**
- **JEROEN, BREEBAART et al.** Multi-Channel goes Mobile : MPEG Surround Binaural Rendering. *AES 29th International Conference, Seoul, Korea,* 2006 **[0123]**