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(54) **APPARATUS AND METHOD FOR GENERATING MULTI-CHANNEL SYNTHESIZER CONTROL SIGNAL AND APPARATUS AND METHOD FOR MULTI-CHANNEL SYNTHESIZING**

VORRICHTUNG UND VERFAHREN ZUR ERZEUGUNG VON STEUERSIGNALEN FÜR MEHRKANALSYNTHESIZER SOWIE VORRICHTUNG UND VERFAHREN FÜR MEHRKANALIGES SYNTHESIZEN

PROCEDE ET DISPOSITIF DE PRODUCTION DE SIGNAL DE COMMANDE DE SYNTHETISEUR MULTIVOIES ET DISPOSITIF ET PROCEDE DE SYNTHESE MULTIVOIES

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## Description

### Field of the invention

**[0001]** The present invention relates to multi-channel audio processing and, in particular, to multi-channel encoding and synthesizing using parametric side information.

### Background of the invention and prior art

**[0002]** In recent times, multi-channel audio reproduction techniques are becoming more and more popular. This may be due to the fact that audio compression/encoding techniques such as the well-known MPEG-1 layer 3 (also known as mp3) technique have made it possible to distribute audio contents via the Internet or other transmission channels having a limited bandwidth.

**[0003]** A further reason for this popularity is the increased availability of multi-channel content and the increased penetration of multi-channel playback devices in the home environment.

**[0004]** The mp3 coding technique has become so famous because of the fact that it allows distribution of all the records in a stereo format, i.e., a digital representation of the audio record including a first or left stereo channel and a second or right stereo channel. Furthermore, the mp3 technique created new possibilities for audio distribution given the available storage and transmission bandwidths

**[0005]** Nevertheless, there are basic shortcomings of conventional two-channel sound systems. They result in a limited spatial imaging due to the fact that only two loudspeakers are used. Therefore, surround techniques have been developed. A recommended multi-channel-surround representation includes, in addition to the two stereo channels L and R, an additional center channel C, two surround channels Ls, Rs and optionally a low frequency enhancement channel or sub-woofer channel. This reference sound format is also referred to as three/two-stereo (or 5.1 format), which means three front channels and two surround channels. Generally, five transmission channels are required. In a playback environment, at least five speakers at the respective five different places are needed to get an optimum sweet spot at a certain distance from the five well-placed loudspeakers.

**[0006]** Several techniques are known in the art for reducing the amount of data required for transmission of a multi-channel audio signal. Such techniques are called joint stereo techniques. To this end, reference is made to Fig. 10, which shows a joint stereo device 60. This device can be a device implementing e.g. intensity stereo (IS), parametric stereo (PS) or (a related) binaural cue coding (BCC). Such a device generally receives - as an input - at least two channels (CH1, CH2, ... CHn), and outputs a single carrier channel and parametric data. The parametric data are defined such that, in a decoder, an

approximation of an original channel (CH1, CH2, ... CHn) can be calculated.

**[0007]** Normally, the carrier channel will include sub-band samples, spectral coefficients, time domain samples etc, which provide a comparatively fine representation of the underlying signal, while the parametric data does not include such samples of spectral coefficients but include control parameters for controlling a certain reconstruction algorithm such as weighting by multiplication, time shifting, frequency shifting, phase shifting. The parametric data, therefore, include only a comparatively coarse representation of the signal of the associated channel. Stated in numbers, the amount of data required by a carrier channel encoded using a conventional lossy audio coder will be in the range of 60 - 70 kBit/s, while the amount of data required by parametric side information for one channel will be in the range of 1,5 - 2,5 kBit/s. An example for parametric data are the well-known scale factors, intensity stereo information or binaural cue parameters as will be described below.

**[0008]** Intensity stereo coding is described in AES preprint 3799, "Intensity Stereo Coding", J. Herre, K. H. Brandenburg, D. Lederer, at 96th AES, February 1994, Amsterdam. Generally, the concept of intensity stereo is based on a main axis transform to be applied to the data of both stereophonic audio channels. If most of the data points are concentrated around the first principle axis, a coding gain can be achieved by rotating both signals by a certain angle prior to coding and excluding the second orthogonal component from transmission in the bit stream. The reconstructed signals for the left and right channels consist of differently weighted or scaled versions of the same transmitted signal. Nevertheless, the reconstructed signals differ in their amplitude but are identical regarding their phase information. The energy-time envelopes of both original audio channels, however, are preserved by means of the selective scaling operation, which typically operates in a frequency selective manner. This conforms to the human perception of sound at high frequencies, where the dominant spatial cues are determined by the energy envelopes.

**[0009]** Additionally, in practical implementations, the transmitted signal, i.e. the carrier channel is generated from the sum signal of the left channel and the right channel instead of rotating both components. Furthermore, this processing, i.e., generating intensity stereo parameters for performing the scaling operation, is performed frequency selective, i.e., independently for each scale factor band, i.e., encoder frequency partition. Preferably, both channels are combined to form a combined or "carrier" channel, and, in addition to the combined channel, the intensity stereo information is determined which depend on the energy of the first channel, the energy of the second channel or the energy of the combined channel.

**[0010]** The BCC technique is described in AES convention paper 5574, "Binaural cue coding applied to stereo and multi-channel audio compression", C. Faller, F. Baumgarte, May 2002, Munich. In BCC encoding, a

number of audio input channels are converted to a spectral representation using a DFT based transform with overlapping windows. The resulting uniform spectrum is divided into non-overlapping partitions each having an index. Each partition has a bandwidth proportional to the equivalent rectangular bandwidth (ERB). The inter-channel level differences (ICLD) and the inter-channel time differences (ICTD) are estimated for each partition for each frame  $k$ . The ICLD and ICTD are quantized and coded resulting in a BCC bit stream. The inter-channel level differences and inter-channel time differences are given for each channel relative to a reference channel. Then, the parameters are calculated in accordance with prescribed formulae, which depend on the certain partitions of the signal to be processed.

**[0011]** At a decoder-side, the decoder receives a mono signal and the BCC bit stream. The mono signal is transformed into the frequency domain and input into a spatial synthesis block, which also receives decoded ICLD and ICTD values. In the spatial synthesis block, the BCC parameters (ICLD and ICTD) values are used to perform a weighting operation of the mono signal in order to synthesize the multi-channel signals, which, after a frequency/time conversion, represent a reconstruction of the original multi-channel audio signal.

**[0012]** In case of BCC, the joint stereo module 60 is operative to output the channel side information such that the parametric channel data are quantized and encoded ICLD or ICTD parameters, wherein one of the original channels is used as the reference channel for coding the channel side information.

**[0013]** Typically, in the most simple embodiment, the carrier channel is formed of the sum of the participating original channels.

**[0014]** Naturally, the above techniques only provide a mono representation for a decoder, which can only process the carrier channel, but is not able to process the parametric data for generating one or more approximations of more than one input channel.

**[0015]** The audio coding technique known as binaural cue coding (BCC) is also well described in the United States patent application publications US 2003, 0219130 A1, 2003/0026441 A1 and 2003/0035553 A1. Additional reference is also made to "Binaural Cue Coding. Part II: Schemes and Applications", C. Faller and F. Baumgarte, IEEE Trans. On Audio and Speech Proc., Vol. 11, No. 6, Nov. 2003. The cited United States patent application publications and the two cited technical publications on the BCC technique authored by Faller and Baumgarte are incorporated herein by reference in their entireties.

**[0016]** Significant improvements of binaural cue coding schemes that make parametric schemes applicable to a much wider bit-rate range are known as 'parametric stereo' (PS), such as standardized in MPEG-4 high-efficiency AAC v2. One of the important extensions of parametric stereo is the inclusion of a spatial 'diffuseness' parameter. This percept is captured in the mathematical property of inter-channel correlation or inter-channel co-

herence (ICC). The analysis, perceptual quantization, transmission and synthesis processes of PS parameters are described in detail in "Parametric coding of stereo audio", J. Breebaart, S. van de Par, A. Kohlrausch and E. Schuijers, EURASIP J. Appl. Sign. Proc. 2005:9, 1305-1322. Further reference is made to J. Breebaart, S. van de Par, A. Kohlrausch, E. Schuijers, "High-Quality Parametric Spatial Audio Coding at Low Bitrates", AES 116th Convention, Berlin, Preprint 6072, May 2004, and E. Schuijers, J. Breebaart, H. Purnhagen, J. Engdegard, "Low Complexity Parametric Stereo Coding", AES 116th Convention, Berlin, Preprint 6073, May 2004.

**[0017]** In the following, a typical generic BCC scheme for multi-channel audio coding is elaborated in more detail with reference to Figures 11 to 13. Figure 11 shows such a generic binaural cue coding scheme for coding/transmission of multi-channel audio signals. The multi-channel audio input signal at an input 110 of a BCC encoder 112 is down mixed in a down mix block 114. In the present example, the original multi-channel signal at the input 110 is a 5-channel surround signal having a front left channel, a front right channel, a left surround channel, a right surround channel and a center channel. In a preferred embodiment of the present invention, the down mix block 114 produces a sum signal by a simple addition of these five channels into a mono signal. Other down mixing schemes are known in the art such that, using a multi-channel input signal, a down mix signal having a single channel can be obtained. This single channel is output at a sum signal line 115. A side information obtained by a BCC analysis block 116 is output at a side information line 117. In the BCC analysis block, inter-channel level differences (ICLD), and inter-channel time differences (ICTD) are calculated as has been outlined above. Recently, the BCC analysis block 116 has inherited Parametric Stereo parameters in the form of inter-channel correlation values (ICC values). The sum signal and the side information is transmitted, preferably in a quantized and encoded form, to a BCC decoder 120. The BCC decoder decomposes the transmitted sum signal into a number of subbands and applies scaling, delays and other processing to generate the subbands of the output multi-channel audio signals. This processing is performed such that ICLD, ICTD and ICC parameters (cues) of a reconstructed multi-channel signal at an output 121 are similar to the respective cues for the original multi-channel signal at the input 110 into the BCC encoder 112. To this end, the BCC decoder 120 includes a BCC synthesis block 122 and a side information processing block 123.

**[0018]** In the following, the internal construction of the BCC synthesis block 122 is explained with reference to Fig. 12. The sum signal on line 115 is input into a time/frequency conversion unit or filter bank FB 125. At the output of block 125, there exists a number  $N$  of sub band signals or, in an extreme case, a block of a spectral coefficients, when the audio filter bank 125 performs a 1:1 transform, i.e., a transform which produces  $N$  spectral

coefficients from N time domain samples.

**[0019]** The BCC synthesis block 122 further comprises a delay stage 126, a level modification stage 127, a correlation processing stage 128 and an inverse filter bank stage IFB 129. At the output of stage 129, the reconstructed multi-channel audio signal having for example five channels in case of a 5-channel surround system, can be output to a set of loudspeakers 124 as illustrated in Fig. 11.

**[0020]** As shown in Fig. 12, the input signal  $s(n)$  is converted into the frequency domain or filter bank domain by means of element 125. The signal output by element 125 is multiplied such that several versions of the same signal are obtained as illustrated by multiplication node 130. The number of versions of the original signal is equal to the number of output channels in the output signal. to be reconstructed When, in general, each version of the original signal at node 130 is subjected to a certain delay  $d_1, d_2, \dots, d_i, \dots, d_N$ . The delay parameters are computed by the side information processing block 123 in Fig. 11 and are derived from the inter-channel time differences as determined by the BCC analysis block 116.

**[0021]** The same is true for the multiplication parameters  $a_1, a_2, \dots, a_i, \dots, a_N$ , which are also calculated by the side information processing block 123 based on the inter-channel level differences as calculated by the BCC analysis block 116.

**[0022]** The ICC parameters calculated by the BCC analysis block 116 are used for controlling the functionality of block 128 such that certain correlations between the delayed and level-manipulated signals are obtained at the outputs of block 128. It is to be noted here that the ordering of the stages 126, 127, 128 may be different from the case shown in Fig. 12.

**[0023]** It is to be noted here that, in a frame-wise processing of an audio signal, the BCC analysis is performed frame-wise, i.e. time-varying, and also frequency-wise. This means that, for each spectral band, the BCC parameters are obtained. This means that, in case the audio filter bank 125 decomposes the input signal into for example 32 band pass signals, the BCC analysis block obtains a set of BCC parameters for each of the 32 bands. Naturally the BCC synthesis block 122 from Fig. 11, which is shown in detail in Fig. 12, performs a reconstruction that is also based on the 32 bands in the example.

**[0024]** In the following, reference is made to Fig. 13 showing a setup to determine certain BCC parameters. Normally, ICLD, ICTD and ICC parameters can be defined between pairs of channels. However, it is preferred to determine ICLD and ICTD parameters between a reference channel and each other channel. This is illustrated in Fig. 13A.

**[0025]** ICC parameters can be defined in different ways. Most generally, one could estimate ICC parameters in the encoder between all possible channel pairs as indicated in Fig. 13B. In this case, a decoder would synthesize ICC such that it is approximately the same as

in the original multi-channel signal between all possible channel pairs. It was, however, proposed to estimate only ICC parameters between the strongest two channels at each time. This scheme is illustrated in Fig. 13C, where an example is shown, in which at one time instance, an ICC parameter is estimated between channels 1 and 2, and, at another time instance, an ICC parameter is calculated between channels 1 and 5. The decoder then synthesizes the inter-channel correlation between the strongest channels in the decoder and applies some heuristic rule for computing and synthesizing the inter-channel coherence for the remaining channel pairs.

**[0026]** Regarding the calculation of, for example, the multiplication parameters  $a_1, \dots, a_N$  based on transmitted ICLD parameters, reference is made to AES convention paper 5574 cited above. The ICLD parameters represent an energy distribution in an original multi-channel signal. Without loss of generality, it is shown in Fig. 13A that there are four ICLD parameters showing the energy difference between all other channels and the front left channel. In the side information processing block 123, the multiplication parameters  $a_1, \dots, a_N$  are derived from the ICLD parameters such that the total energy of all reconstructed output channels is the same as (or proportional to) the energy of the transmitted sum signal. A simple way for determining these parameters is a 2-stage process, in which, in a first stage, the multiplication factor for the left front channel is set to unity, while multiplication factors for the other channels in Fig. 13A are set to the transmitted ICLD values. Then, in a second stage, the energy of all five channels is calculated and compared to the energy of the transmitted sum signal. Then, all channels are downscaled using a downscaling factor that is equal for all channels, wherein the downscaling factor is selected such that the total energy of all reconstructed output channels is, after downscaling, equal to the total energy of the transmitted sum signal.

**[0027]** Naturally, there are other methods for calculating the multiplication factors, which do not rely on the 2-stage process but which only need a 1-stage process. A 1-stage method is described in AES preprint "The reference model architecture for MPEG spatial audio coding", J. Herre et al., 2005, Barcelona.

**[0028]** Regarding the delay parameters, it is to be noted that the delay parameters ICTD, which are transmitted from a BCC encoder can be used directly, when the delay parameter  $d_1$  for the left front channel is set to zero. No rescaling has to be done here, since a delay does not alter the energy of the signal.

**[0029]** Regarding the inter-channel coherence measure ICC transmitted from the BCC encoder to the BCC decoder, it is to be noted here that a coherence manipulation can be done by modifying the multiplication factors  $a_1, \dots, a_N$  such as by multiplying the weighting factors of all subbands with random numbers with values between  $20\log_{10}(-6)$  and  $20\log_{10}(6)$ . The pseudo-random sequence is preferably chosen such that the variance is approximately constant for all critical bands, and the av-

erage is zero within each critical band. The same sequence is applied to the spectral coefficients for each different frame. Thus, the auditory image width is controlled by modifying the variance of the pseudo-random sequence. A larger variance creates a larger image width. The variance modification can be performed in individual bands that are critical-band wide. This enables the simultaneous existence of multiple objects in an auditory scene, each object having a different image width. A suitable amplitude distribution for the pseudo-random sequence is a uniform distribution on a logarithmic scale as it is outlined in the US patent application publication 2003/0219130 A1. Nevertheless, all BCC synthesis processing is related to a single input channel transmitted as the sum signal from the BCC encoder to the BCC decoder as shown in Fig. 11.

**[0030]** As has been outlined above with respect to Fig. 13, the parametric side information, i.e., the interchannel level differences (ICLD), the interchannel time differences (ICTD) or the interchannel coherence parameter (ICC) can be calculated and transmitted for each of the five channels. This means that one, normally, transmits five sets of interchannel level differences for a five-channel signal. The same is true for the interchannel time differences. With respect to the interchannel coherence parameter, it can also be sufficient to only transmit for example two sets of these parameters.

**[0031]** As has been outlined above with respect to Fig. 12, there is not a single level difference parameter, time difference parameter or coherence parameter for one frame or time portion of a signal. Instead, these parameters are determined for several different frequency bands so that a frequency-dependent parameterisation is obtained. Since it is preferred to use for example 32 frequency channels, i.e., a filter bank having 32 frequency bands for BCC analysis and BCC synthesis, the parameters can occupy quite a lot of data. Although - compared to other multi-channel transmissions - the parametric representation results in a quite low data rate, there is a continuing need for further reduction of the necessary data rate for representing a multi-channel signal such as a signal having two channels (stereo signal) or a signal having more than two channels such as a multi-channel surround signal.

**[0032]** To this end, the encoder-side calculated reconstruction parameters are quantized in accordance with a certain quantization rule. This means that unquantized reconstruction parameters are mapped onto a limited set of quantization levels or quantization indices as it is known in the art and described specifically for parametric coding in detail in "Parametric coding of stereo audio", J. Breebaart, S. van de Par, A. Kohlrausch and E. Schuijers, EURASIP J. Appl. Sign. Proc. 2005:9, 1305-1322. and in C. Faller and F. Baumgarte, "Binaural cue coding applied to audio compression with flexible rendering," AES 113th Convention, Los Angeles, Preprint 5686, October 2002.

**[0033]** Quantization has the effect that all parameter

values, which are smaller than the quantization step size, are quantized to zero, depending on whether the quantizer is of the mid-tread or mid-riser type. By mapping a large set of unquantized values to a small set of quantized values additional data saving are obtained. These data rate savings are further enhanced by entropy-encoding the quantized reconstruction parameters on the encoder-side. Preferred entropy-encoding methods are Huffman methods based on predefined code tables or based on an actual determination of signal statistics and signal-adaptive construction of codebooks. Alternatively, other entropy-encoding tools can be used such as arithmetic encoding.

**[0034]** Generally, one has the rule that the data rate required for the reconstruction parameters decreases with increasing quantizer step size. Differently stated, a coarser quantization results in a lower data rate, and a finer quantization results in a higher data rate.

**[0035]** Since parametric signal representations are normally required for low data rate environments, one tries to quantize the reconstruction parameters as coarse as possible to obtain a signal representation having a certain amount of data in the base channel, and also having a reasonable small amount of data for the side information which include the quantized and entropy-encoded reconstruction parameters.

**[0036]** Prior art methods, therefore, derive the reconstruction parameters to be transmitted directly from the multi-channel signal to be encoded. A coarse quantization as discussed above results in reconstruction parameter distortions, which result in large rounding errors, when the quantized reconstruction parameter is inversely quantized in a decoder and used for multi-channel synthesis. Naturally, the rounding error increases with the quantizer step size, i.e., with the selected "quantizer coarseness". Such rounding errors may result in a quantization level change, i.e., in a change from a first quantization level at a first time instant to a second quantization level at a later time instant, wherein the difference between one quantizer level and another quantizer level is defined by the quite large quantizer step size, which is preferable for a coarse quantization. Unfortunately, such a quantizer level change amounting to the large quantizer step size can be triggered by only a small change in parameter, when the unquantized parameter is in the middle between two quantization levels. It is clear that the occurrence of such quantizer index changes in the side information results in the same strong changes in the signal synthesis stage. When - as an example - the interchannel level difference is considered, it becomes clear that a large change results in a large decrease of loudness of a certain loudspeaker signal and an accompanying large increase of the loudness of a signal for another loudspeaker. This situation, which is only triggered by a single quantization level change for a coarse quantization can be perceived as an immediate relocation of a sound source from a (virtual) first place to a (virtual) second place. Such an immediate relocation

from one time instant to another time instant sounds unnatural, i.e., is perceived as a modulation effect, since sound sources of, in particular, tonal signals do not change their location very fast.

**[0037]** Generally, also transmission errors may result in large changes of quantizer indices, which immediately result in the large changes in the multi-channel output signal, which is even more true for situations, in which a coarse quantizer for data rate reasons has been adopted.

**[0038]** State-of-the-art techniques for the parametric coding of two ("stereo") or more ("multi-channel") audio input channels derive the spatial parameters directly from the input signals. Examples of such parameters are - as outlined above - inter-channel level differences (ICLD) or inter-channel intensity differences (IID), inter-channel time delays (ICTD) or inter-channel phase differences (IPD), and inter-channel correlation/coherence (ICC), each of which are transmitted in a time and frequency-selective fashion, i.e. per frequency band and as a function of time. For the transmission of such parameters to the decoder, a coarse quantization of these parameters is desirable to keep the side information rate at a minimum. As a consequence, considerable rounding errors occur when comparing the transmitted parameter values to their original values. This means that even a soft and gradual change of one parameter in the original signal may lead to an abrupt change in the parameter value used in the decoder if the decision threshold from one quantized parameter value to the next value is exceeded. Since these parameter values are used for the synthesis of the output signal, abrupt changes in parameter values may also cause "jumps" in the output signal which are perceived as annoying for certain types of signals as "switching" or "modulation" artifacts (depending on the temporal granularity and quantization resolution of the parameters).

**[0039]** The US Patent Application Serial No. 10/883,538 describes a process for post processing transmitted parameter values in the context of BCC-type methods in order to avoid artifacts for certain types of signals when representing parameters at low resolution. These discontinuities in the synthesis process lead to artifacts for tonal signals. Therefore, the US Patent Application proposes to use a tonality detector in the decoder, which is used to analyze the transmitted down-mix signal. When the signal is found to be tonal, then a smoothing operation over time is performed on the transmitted parameters. Consequently, this type of processing represents a means for efficient transmission of parameters for tonal signals.

**[0040]** There are, however, classes of input signals other than tonal input signals, which are equally sensitive to a coarse quantization of spatial parameters.

- One example for such cases are point sources that are moving slowly between two positions (e.g. a noise signal panned very slowly to move between Center and Left Front speaker). A coarse quantiza-

tion of level parameters will lead to perceptible "jumps" (discontinuities) in the spatial position and trajectory of the sound source. Since these signals are generally not detected as tonal in the decoder, prior-art smoothing will obviously not help in this case.

- Other examples are rapidly moving point sources that have tonal material, such as fast moving sinusoids. Prior-art smoothing will detect these components as tonal and thus invoke a smoothing operation. However, as the speed of movement is not known to the prior-art smoothing algorithm, the applied smoothing time constant would be generally inappropriate and e.g. reproduce a moving point source with a much too slow speed of movement and a significant lag of reproduced spatial position as compared to the originally intended position.

**[0041]** US patent No. 5,890,125 discloses a method and apparatus for encoding and decoding multiple audio channels at low bit rates using adaptive selection of encoding method for limiting the temporal rate at which signals change, temporal smoothing is applied. Particularly, the rate at which the spectral level measures can change is reduced.

**[0042]** WO 2005/086139 A1 discloses multichannel audio coding, in which multiple channels of audio are combined either to a monophonic composite signal or to multiple channels of audio along with related auxiliary information from which multiple channels of audio are reconstructed. The monophonic composite signal or the multiple channels of audio are input into an upmix matrix. The output of the upmix matrix is input into adjust amplitude blocks, rotate angle blocks and, subsequently, into inverse filterbanks to provide different reconstructed audio channels. When an interpolation flag is employed, an optional frequency interpolator or interpolation function may be employed in order to interpolate an angle control parameter across frequency. Such interpolation may be, for example, a linear interpolation of the bin angles between the centers of each subband. The state of the 1-bit interpolation flag selects, whether or not interpolation across frequency is employed.

**[0043]** It is the object of the present invention to provide an improved audio signal processing concept allowing a low data rate on the one hand and a good subjective quality on the other hand.

**[0044]** This object is achieved by an apparatus of claim 1

**[0045]** or a multi-channel synthesizer of claim 16

**[0046]** or a method of generating a multi-channel synthesizer control signal of claim 15 or a method of generating an output signal from an input signal of claim 23 corresponding computer programs of claim 32 or a multi-channel synthesizer control signal of claim 24.

**[0047]** The present invention is based on the finding that an encoder-side directed smoothing of reconstruc-

tion parameters will result in an improved audio quality of the synthesized multi-channel output signal. This substantial improvement of the audio quality can be obtained by an additional encoder-side processing to determine the smoothing control information, which can, in preferred embodiments of the present invention, transmitted to the decoder, which transmission only requires a limited (small) number of bits.

**[0048]** On the decoder-side, the smoothing control information is used to control the smoothing operation. This encoder-guided parameter smoothing on the decoder-side can be used instead of the decoder-side parameter smoothing, which is based on for example tonality/transient detection, or can be used in combination with the decoder-side parameter smoothing. Which method is applied for a certain time portion and a certain frequency band of the transmitted down-mix signal can also be signaled using the smoothing control information as determined by a signal analyzer on the encoder-side.

**[0049]** To summarize, the present invention is advantageous in that an encoder-side controlled adaptive smoothing of reconstruction parameters is performed within a multi-channel synthesizer, which results in a substantial increase of audio quality on the one hand and which only results in a small amount of additional bits. Due of the fact that the inherent quality deterioration of quantization is mitigated using the additional smoothing control information, the inventive concepts can even be applied without any increase and even with a decrease of transmitted bits, since the bits for the smoothing control information can be saved by applying an even coarser quantization so that less bits are required for encoding the quantized values. Thus, the smoothing control information together with the encoded quantized values can even require the same or less bit rate of quantized values without smoothing control information as outlined in the non-published US-patent application, while keeping the same level or a higher level of subjective audio quality.

**[0050]** Generally, the post processing for quantized reconstruction parameters used in a multi-channel synthesizer is operative to reduce or even eliminate problems associated with coarse quantization on the one hand and quantization level changes on the other hand.

**[0051]** While, in prior art systems, a small parameter change in an encoder may result in a strong parameter change at the decoder, since a requantization in the synthesizer is only admissible for the limited set of quantized values, the inventive device performs a post processing of reconstruction parameters so that the post processed reconstruction parameter for a time portion to be processed of the input signal is not determined by the encoder-adopted quantization raster, but results in a value of the reconstruction parameter, which is different from a value obtainable by the quantization in accordance with the quantization rule.

**[0052]** While, in a linear quantizer case, the prior art method only allows inversely quantized values being in-

teger multiples of the quantizer step size, the inventive post processing allows inversely quantized values to be non-integer multiples of the quantizer step size. This means that the inventive post processing alleviates the quantizer step size limitation, since also post processed reconstruction parameters lying between two adjacent quantizer levels can be obtained by post processing and used by the inventive multi-channel reconstructor, which makes use of the post processed reconstruction parameter.

**[0053]** This post processing can be performed before or after requantization in a multi-channel synthesizer. When the post processing is performed with the quantized parameters, i.e., with the quantizer indices, an inverse quantizer is needed, which can inversely quantize not only to quantizer step multiples, but which can also inversely quantize to inversely quantized values between multiples of the quantizer step size.

**[0054]** In case the post processing is performed using inversely quantized reconstruction parameters, a straight-forward inverse quantizer can be used, and an interpolation/filtering/smoothing is performed with the inversely quantized values.

**[0055]** In case of a non-linear quantization rule, such as a logarithmic quantization rule, a post processing of the quantized reconstruction parameters before requantization is preferred, since the logarithmic quantization is similar to the human ear's perception of sound, which is more accurate for low-level sound and less accurate for high-level sound, i.e., makes a kind of a logarithmic compression.

**[0056]** It is to be noted here that the inventive merits are not only obtained by modifying the reconstruction parameter itself that is included in the bit stream as the quantized parameter. The advantages can also be obtained by deriving a post processed quantity from the reconstruction parameter. This is especially useful, when the reconstruction parameter is a difference parameter and a manipulation such as smoothing is performed on an absolute parameter derived from the difference parameter.

**[0057]** In a preferred embodiment of the present invention, the post processing for the reconstruction parameters is controlled by means of a signal analyser, which analyses the signal portion associated with a reconstruction parameter to find out, which signal characteristic is present. In a preferred embodiment, the decoder controlled post processing is activated only for tonal portions of the signal (with respect to frequency and/or time) or when the tonal portions are generated by a point source only for slowly moving point sources, while the post processing is deactivated for non-tonal portions, i.e., transient portions of the input signal or rapidly moving point sources having tonal material. This makes sure that the full dynamic of reconstruction parameter changes is transmitted for transient sections of the audio signal, while this is not the case for tonal portions of the signal.

**[0058]** Preferably, the post processor performs a mod-

ification in the form of a smoothing of the reconstruction parameters, where this makes sense from a psycho-acoustic point of view, without affecting important spatial detection cues, which are of special importance for non-tonal, i.e., transient signal portions.

**[0059]** The present invention results in a low data rate, since an encoder-side quantization of reconstruction parameters can be a coarse quantization, since the system designer does not have to fear significant changes in the decoder because of a change from a reconstruction parameter from one inversely quantized level to another inversely quantized level, which change is reduced by the inventive processing by mapping to a value between two requantization levels.

**[0060]** Another advantage of the present invention is that the quality of the system is improved, since audible artefacts caused by a change from one requantization level to the next allowed requantization level are reduced by the inventive post processing, which is operative to map to a value between two allowed requantization levels.

**[0061]** Naturally, the inventive post processing of quantized reconstruction parameters represents a further information loss, in addition to the information loss obtained by parameterisation in the encoder and subsequent quantization of the reconstruction parameter. This, however, is not a problem, since the inventive post processor preferably uses the actual or preceding quantized reconstruction parameters for determining a post processed reconstruction parameter to be used for reconstruction of the actual time portion of the input signal, i.e., the base channel. It has been shown that this results in an improved subjective quality, since encoder-induced errors can be compensated to a certain degree. Even when encoder-side induced errors are not compensated by the post processing of the reconstruction parameters, strong changes of the spatial perception in the reconstructed multi-channel audio signal are reduced, preferably only for tonal signal portions, so that the subjective listening quality is improved in any case, irrespective of the fact, whether this results in a further information loss or not.

#### Brief description of the drawings

**[0062]** Preferred embodiments of the present invention are subsequently described by referring to the enclosed drawings, in which:

Fig. 1a is a schematic diagram of an encoder-side device and the corresponding decoder-side device in accordance with the first embodiment of the present invention;

Fig. 1b is a schematic diagram of an encoder-side device and the corresponding decoder-side device in accordance with a further preferred embodiment of the present invention;

Fig. 1c is a schematic block diagram of a preferred control signal generator;

5 Fig. 2a is a schematic representation for determining the spatial position of a sound source;

Fig. 2b is a flow chart of a preferred embodiment for calculating a smoothing time constant as an example for smoothing information;

10 Fig. 3a is an alternative embodiment for calculating quantized inter-channel intensity differences and corresponding smoothing parameters;

15 Fig. 3b is an exemplary diagram illustrating the difference between a measured IID parameter per frame and a quantized IID parameter per frame and a processed quantized IID parameter per frame for various time constants;

20 Fig. 3c is a flow chart of a preferred embodiment of the concept as applied in Fig. 3a;

25 Fig. 4a is a schematic representation illustrating a decoder-side directed system;

Fig. 4b is a schematic diagram of a post processor/signal analyzer combination to be used in the inventive multi-channel synthesizer of Fig. 1b;

30 Fig. 4c is a schematic representation of time portions of the input signal and associated quantized reconstruction parameters for past signal portions, actual signal portions to be processed and future signal portions;

35 Fig. 5 is an embodiment of the encoder guided parameter smoothing device from Fig. 1;

40 Fig. 6a is another embodiment of the encoder guided parameter smoothing device shown in Fig. 1;

Fig. 6b is another preferred embodiment of the encoder guided parameter smoothing device;

45 Fig. 7a is another embodiment of the encoder guided parameter smoothing device shown in Fig. 1;

Fig. 7b is a schematic indication of the parameters to be post processed in accordance with the invention showing that also a quantity derived from the reconstruction parameter can be smoothed;

55 Fig. 8 is a schematic representation of a quantizer/inverse quantizer performing a straightforward mapping or an enhanced mapping;



- Fig. 9a is an exemplary time course of quantized reconstruction parameters associated with subsequent input signal portions;
- Fig. 9b is a time course of post processed reconstruction parameters, which have been post-processed by the post processor implementing a smoothing (low-pass) function;
- Fig. 10 illustrates a prior art joint stereo encoder;
- Fig. 11 is a block diagram representation of a prior art BCC encoder/decoder chain;
- Fig. 12 is a block diagram of a prior art implementation of a BCC synthesis block of Fig. 11;
- Fig. 13 is a representation of a well-known scheme for determining ICLD, ICTD and ICC parameters;
- Fig. 14 a transmitter and a receiver of a transmission system; and
- Fig. 15 an audio recorder having an inventive encoder and an audio player having a decoder.

**[0063]** Figs. 1a and 1b show block diagrams of inventive multi-channel encoder/synthesizer scenarios. As will be shown later with respect to Fig. 4c, a signal arriving on the decoder-side has at least one input channel and a sequence of quantized reconstruction parameters, the quantized reconstruction parameters being quantized in accordance with a quantization rule. Each reconstruction parameter is associated with a time portion of the input channel so that a sequence of time portions is associated with a sequence of quantized reconstruction parameters. Additionally, the output signal, which is generated by a multi-channel synthesizer as shown in Figs. 1a and 1b has a number of synthesized output channels, which is in any case greater than the number of input channels in the input signal. When the number of input channels is 1, i.e. when there is a single input channel, the number of output channels will be 2 or more. When, however, the number of input channels is 2 or 3, the number of output channels will be at least 3 or at least 4 respectively.

**[0064]** In the BCC case, the number of input channels will be 1 or generally not more than 2, while the number of output channels will be 5 (left-surround, left, center, right, right surround) or 6 (5 surround channels plus 1 sub-woofer channel) or even more in case of a 7.1 or 9.1 multi-channel format. Generally stated, the number of output sources will be higher than the number of input sources.

**[0065]** Fig. 1a illustrates, on the left side, an apparatus 1 for generating a multi-channel synthesizer control signal. Box 1 titled "Smoothing Parameter Extraction" comprises a signal analyzer, a smoothing information calcu-

lator and a data generator. As shown in Fig. 1c, the signal analyzer 1a receives, as an input, the original multi-channel signal. The signal analyzer analyses the multi-channel input signal to obtain an analysis result. This analysis result is forwarded to the smoothing information calculator for determining smoothing control information in response to the signal analyzer, i.e. the signal analysis result. In particular, the smoothing information calculator 1b is operative to determine the smoothing information such that, in response to the smoothing control information, a decoder-side parameter post processor generates a smoothed parameter or a smoothed quantity derived from the parameter for a time portion of the input signal to be processed, so that a value of the smoothed reconstruction parameter or the smoothed quantity is different from a value obtainable using requantization in accordance with a quantization rule.

**[0066]** Furthermore, the smoothing parameter extraction device 1 in Fig. 1a includes a data generator for outputting a control signal representing the smoothing control information as the decoder control signal.

**[0067]** In particular, the control signal representing the smoothing control information can be a smoothing mask, a smoothing time constant, or any other value controlling a decoder-side smoothing operation so that a reconstructed multi-channel output signal, which is based on smoothed values has an improved quality compared to reconstructed multi-channel output signals, which is based on non-smoothed values.

**[0068]** The smoothing mask includes the signaling information consisting e.g. of flags that indicate the "on/off" state of each frequency used for smoothing. Thus, the smoothing mask can be seen as a vector associated to one frame having a bit for each band, wherein this bit controls, whether the encoder-guided smoothing is active for this band or not.

**[0069]** A spatial audio encoder as shown in Fig. 1a preferably includes a down-mixer 3 and a subsequent audio encoder 4. Furthermore, the spatial audio encoder includes a spatial parameter extraction device 2, which outputs quantized spatial cues such as inter-channel level differences (ICLD), inter-channel time differences (ICTDs), inter-channel coherence values (ICC), inter-channel phase differences (IPD), inter-channel intensity differences (IIDs), etc. In this context, it is to be outlined that inter-channel level differences are substantially the same as inter-channel intensity differences.

**[0070]** The down-mixer 3 may be constructed as outlined for item 114 in Fig. 11. Furthermore, the spatial parameter extraction device 2 may be implemented as outlined for item 116 in Fig. 11. Nevertheless, alternative embodiments for the down-mixer 3 as well as the spatial parameter extractor 2 can be used in the context of the present invention.

**[0071]** Furthermore, the audio encoder 4 is not necessarily required. This device, however, is used, when the data rate of the down-mix signal at the output of element 3 is too high for a transmission of the down-mix signal

via the transmission/storage means.

**[0072]** A spatial audio decoder includes an encoder-guided parameter smoothing device 9a, which is coupled to multi-channel up-mixer 12. The input signal for the multi-channel up-mixer 12 is normally the output signal of an audio decoder 8 for decoding the transmitted/stored down-mix signal.

**[0073]** Preferably, the inventive multi-channel synthesizer for generating an output signal from an input signal, the input signal having at least one input channel and a sequence of quantized reconstruction parameters, the quantized reconstruction parameters being quantized in accordance with a quantization rule, and being associated with subsequent time portions of the input signal, the output signal having a number of synthesized output channels, and the number of synthesized output channels being greater than one or greater than a number of input channels, comprises a control signal provider for providing a control signal having the smoothing control information. This control signal provider can be a data stream demultiplexer, when the control information is multiplexed with the parameter information. When, however, the smoothing control information is transmitted from device 1 to device 9a in Fig. 1a via a separate channel, which is separated from the parameter channel 14a or the down-mix signal channel, which is connected to the input-side of the audio decoder 8, then the control signal provider is simply an input of device 9a receiving the control signal generated by the smoothing parameter extraction device 1 in Fig. 1a.

**[0074]** Furthermore, the inventive multi-channel synthesizer comprises a post processor 9a, which is also termed an "encoder-guided parameter smoothing device". The post processor is for determining a post processed reconstruction parameter or a post processed quantity derived from the reconstruction parameter for a time portion of the input signal to be processed, wherein the post processor is operative to determine the post processed reconstruction parameter or the post processed quantity such that a value of the post processed reconstruction parameter or the post processed quantity is different from a value obtainable using requantization in accordance with the quantization rule. The post processed reconstruction parameter or the post processed quantity is forwarded from device 9a to the multi-channel up mixer 12 so that the multi-channel up mixer or multi-channel reconstructor 12 can perform a reconstruction operation for reconstructing a time portion of the number of synthesized output channels using the time portion of the input channel and the post processed reconstruction parameter or the post processed value.

**[0075]** Subsequently, reference is made to the preferred embodiment of the present invention illustrated in Fig. 1b, which combines the encoder-guided parameter smoothing and the decoder-guided parameter smoothing as defined in the non-prepublished US-patent application No. 10/883,538. In this embodiment, the smoothing parameter extraction device 1, which is shown in de-

tail in Fig. 1c additionally generates an encoder/decoder control flag 5a, which is transmitted to a combined/switch results block 9b.

**[0076]** The Fig. 1b multi-channel synthesizer or spatial audio decoder includes a reconstruction parameter post processor 10, which is the decoder-guided parameter-smoothing device, and the multi-channel reconstructor 12. The decoder-guided parameter-smoothing device 10 is operative to receive quantized and preferably encoded reconstruction parameters for subsequent time portions of the input signal. The reconstruction parameter post processor 10 is operative to determine the post-processed reconstruction parameter at an output thereof for a time portion to be processed of the input signal. The reconstruction parameter post processor operates in accordance with a post-processing rule, which is in certain preferred embodiments a low-pass filtering rule, a smoothing rule, or another similar operation. In particular, the post processor is operative to determine the post processed reconstruction parameter such that a value of the post-processed reconstruction parameter is different from a value obtainable by requantization of any quantized reconstruction parameter in accordance with the quantization rule.

**[0077]** The multi-channel reconstructor 12 is used for reconstructing a time portion of each of the number of synthesis output channels using the time portions of the processed input channel and the post processed reconstruction parameter.

**[0078]** In preferred embodiments of the present invention, the quantized reconstruction parameters are quantized BCC parameters such as inter-channel level differences, inter-channel time differences or inter-channel coherence parameters or inter-channel phase differences or inter-channel intensity differences. Naturally, all other reconstruction parameters such as stereo parameters for intensity stereo or parameters for parametric stereo can be processed in accordance with the present invention as well.

**[0079]** The encoder/decoder control flag transmitted via line 5a is operative to control the switch or combine device 9b to forward either decoder-guided smoothing values or encoder-guided smoothing values to the multi-channel up mixer 12.

**[0080]** In the following, reference will be made to Fig. 4c, which shows an example for a bit stream. The bit stream includes several frames 20a, 20b, 20c,... Each frame includes a time portion of the input signal indicated by the upper rectangle of a frame in Fig. 4c. Additionally, each frame includes a set of quantized reconstruction parameters which are associated with the time portion, and which are illustrated in Fig. 4c by the lower rectangle of each frame 20a, 20b, 20c. Exemplarily, frame 20b is considered as the input signal portion to be processed, wherein this frame has preceding input signal portions, i.e., which form the "past" of the input signal portion to be processed. Additionally, there are following input signal portions, which form the "future" of the input signal

portion to be processed (the input portion to be processed is also termed as the "actual" input signal portion), while input signal portions in the "past" are termed as former input signal portions, while signal portions in the future are termed as later input signal portions.

**[0081]** The inventive method successfully handles problematic situations with slowly moving point sources preferably having noise-like properties or rapidly moving point sources having tonal material such as fast moving sinusoids by allowing a more explicit encoder control of the smoothing operation carried out in the decoder.

**[0082]** As outlined before, the preferred way of performing a post-processing operation within the encoder-guided parameter smoothing device 9a or the decoder-guided parameter smoothing device 10 is a smoothing operation carried out in a frequency-band oriented way.

**[0083]** Furthermore, in order to actively control the post processing in the decoder performed by the encoder-guided parameter smoothing device 9a, the encoder conveys signaling information preferably as part of the side information to the synthesizer/decoder. The multi-channel synthesizer control signal can, however, also be transmitted separately to the decoder without being part of side information of parametric information or down-mix signal information.

**[0084]** In a preferred embodiment, this signaling information consists of flags that indicate the "on/off" state of each frequency band used for smoothing. In order to allow an efficient transmission of this information, a preferred embodiment can also use a set of "short cuts" to signal certain frequently used configurations with very few bits.

**[0085]** To this end, the smoothing information calculator 1b in Fig. 1c determines that no smoothing is to be carried out in any of the frequency bands. This is signaled via an "all-off" short cut signal generated by the data generator 1c. In particular, a control signal representing the "all-off" short cut signal can be a certain bit pattern or a certain flag.

**[0086]** Furthermore, the smoothing information calculator 1b may determine that in all frequency bands, an encoder-guided smoothing operation is to be performed. To this end, the data generator 1c generates an "all-on" short cut signal, which signals that smoothing is applied in all frequency bands. This signal can be a certain bit pattern or a flag.

**[0087]** Furthermore, when the signal analyzer 1a determines that the signal did not very much change from one time portion to the next time portion, i.e. from a current time portion to a future time portion, the smoothing information calculator 1b may determine that no change in the encoder-guided parameter smoothing operation has to be performed. Then, the data generator 1c will generate a "repeat last mask" short cut signal, which will signal to the decoder/synthesizer that the same band-wise on/off status shall be used for smoothing as it was employed for the processing of the previous frame.

**[0088]** In a preferred embodiment, the signal analyzer

1a is operative to estimate the speed of movement so that the impact of the decoder smoothing is adapted to the speed of a spatial movement of a point source. As a result of this process, a suitable smoothing time constant is determined by the smoothing information calculator 1b and signaled to the decoder by dedicated side information via data generator 1c. In a preferred embodiment, the data generator 1c generates and transmits an index value to a decoder, which allows the decoder to select between different pre-defined smoothing time constants (such as 125 ms, 250 ms, 500 ms,...). In a further preferred embodiment, only one time constant is transmitted for all frequency bands. This reduces the amount of signaling information for smoothing time constants and is sufficient for the frequently occurring case of one dominant moving point source in the spectrum. An exemplary process of determining a suitable smoothing time constant is described in connection with Figs. 2a and 2b.

**[0089]** The explicit control of the decoder smoothing process requires a transmission of some additional side information compared to a decoder-guided smoothing method. Since this control may only be necessary for a certain fraction of all input signals with specific properties, both approaches are preferably combined into a single method, which is also called the "hybrid method". This can be done by transmitting signaling information such as one bit determining whether smoothing is to be carried out based on a tonality/transient estimation in the decoder as performed by device 16 in Fig. 1b or under explicit encoder control. In the latter case, the side information 5a of Fig. 1b is transmitted to the decoder.

**[0090]** Subsequently, preferred embodiments for identifying slowly moving point sources and estimating appropriate time constants to be signaled to a decoder are discussed. Preferably, all estimations are carried out in the encoder and can, thus, access non-quantized versions of signal parameters, which are, of course, not available in the decoder because of the fact that device 2 in Fig. 1a and Fig. 1b transmits quantized spatial cues for data compression reasons.

**[0091]** Subsequently, reference is made to Figs. 2a and 2b for showing a preferred embodiment for identification of slowly moving point sources. The spatial position of a sound event within a certain frequency band and time frame is identified as shown in connection with Fig. 2a. In particular, for each audio output channel, a unit-length vector  $e_x$  indicates the relative positioning of the corresponding loud speaker in a regular listening set-up. In the example shown in Fig. 2a, the common 5-channel listening set-up is used with speakers L, C, R, Ls, and Rs and the corresponding unit-length vectors  $e_L$ ,  $e_C$ ,  $e_R$ ,  $e_{Ls}$ , and  $e_{Rs}$ .

**[0092]** The spatial position of the sound event within a certain frequency band and time frame is calculated as the energy-weighted average of these vectors as outlined in the equation of Fig. 2a. As becomes clear from Fig. 2a, each unit-length vector has a certain x-coordinate and a certain y-coordinate. By multiplying each coordi-

nate of the unit-length vector with the corresponding energy and by summing-up the x-coordinate terms and the y-coordinate terms, a spatial position for a certain frequency band and a certain time frame at a certain position  $x, y$  is obtained.

**[0093]** As outlined in step 40 of Fig. 2b, this determination is performed for two subsequent time instants.

**[0094]** Then, in step 41, it is determined, whether the source having the spatial positions  $p_1, p_2$  is slowly moving. When the distance between subsequent spatial positions is below a predetermined threshold, then the source is determined to be a slowly moving source. When, however, it is determined that the displacement is above a certain maximum displacement threshold, then it is determined that the source is not slowly moving, and the process in Fig. 2b is stopped.

**[0095]** Values  $L, C, R, L_s$ , and  $R_s$  in Fig. 2a denote energies of the corresponding channels, respectively. Alternatively, the energies measured in dB may also be employed for determining a spatial position  $p$ .

**[0096]** In step 42, it is determined, whether the source is a point or a near point source. Preferably, point sources are detected, when the relevant ICC parameters exceed a certain minimum threshold such as 0.85. When it is determined that the ICC parameter is below the predetermined threshold, then the source is not a point source and the process in Fig. 2a is stopped. When, however, it is determined that the source is a point source or a near point source, the process in Fig. 2b advances to step 43. In this step, preferably the inter-channel level difference parameters of the parametric multi-channel scheme are determined within a certain observation interval, resulting in a number of measurements. The observation interval may consist of a number of coding frames or a set of observations taking place at a higher time resolution than defined by the sequence of frames.

**[0097]** In a step 44, the slope of an ICLD curve for subsequent time instances is calculated. Then, in step 45, a smoothing time constant is chosen, which is inversely proportional to the slope of the curve.

**[0098]** Then, in step 45, a smoothing time constant as an example of a smoothing information is output and used in a decoder-side smoothing device, which, as it becomes clear from Figs. 4a and 4b may be a smoothing filter. The smoothing time constant determined in step 45 is, therefore, used to set filter parameters of a digital filter used for smoothing in block 9a.

**[0099]** Regarding Fig. 1b, it is emphasized that the encoder-guided parameter smoothing 9a and decoder-guided parameter smoothing 10 can also be implemented using a single device such as shown in Fig. 4b, 5, or 6a, since the smoothing control information on the one hand and the decoder-determined information output by the control parameter extraction device 16 on the other hand both act on a smoothing filter and the activation of the smoothing filter in a preferred embodiment of the present invention.

**[0100]** When only one common smoothing time con-

stant is signaled for all frequency bands, the individual results for each band can be combined into an overall result e.g. by averaging or energy-weighted averaging. In this case, the decoder applies the same (energy-weighted) averaged smoothing time constant to each band so that only a single smoothing time constant for the whole spectrum needs to be transmitted. When bands are found with a significant deviation from the combined time constant, smoothing may be disabled for these bands using the corresponding "on/off" flags.

**[0101]** Subsequently, reference is made to Figs. 3a, 3b, and 3c to illustrate an alternative embodiment, which is based on an analysis-by-synthesis approach for encoder-guided smoothing control. The basic idea consists of a comparison of a certain reconstruction parameter (preferably the IID/ICLD parameter) resulting from quantization and parameter smoothing to the corresponding non-quantized (i.e. measured) (IID/ICLD) parameter. This process is summarized in the schematic preferred embodiment illustrated in Fig. 3a. Two different multi-channel input channels such as  $L$  on the one hand and  $R$  on the other hand are input in respective analysis filter banks. The filter bank outputs are segmented and windowed to obtain a suitable time/frequency representation.

**[0102]** Thus, Fig. 3a includes an analysis filter bank device having two separate analysis filter banks 70a, 70b. Naturally, a single analysis filter bank and a storage can be used twice to analyze both channels. Then, in the segmentation and windowing device 72, the time segmentation is performed. Then, an ICLD/IID estimation per frame is performed in device 73. The parameter for each frame is subsequently sent to a quantizer 74. Thus, a quantized parameter at the output of device 74 is obtained. The quantized parameter is subsequently processed by a set of different time constants in device 75. Preferably, essentially all time constants that are available to the decoder are used by device 75. Finally, a comparison and selection unit 76 compares the quantized and smoothed IID parameters to the original (unprocessed) IID estimates. Unit 76 outputs the quantized IID parameter and the smoothing time constant that resulted in a best fit between processed and originally measured IID values.

**[0103]** Subsequently, reference is made to the flow chart in Fig. 3c, which corresponds to the device in Fig. 3a. As outlined in step 46, IID parameters for several frames are generated. Then, in step 47, these IID parameters are quantized. In step 48, the quantized IID parameters are smoothed using different time constants. Then, in step 49, an error between a smoothed sequence and an originally generated sequence is calculated for each time constant used in step 49. Finally, in step 50, the quantized sequence is selected together with the smoothing time constant, which resulted in the smallest error. Then, step 50 outputs the sequence of quantized values together with the best time constant.

**[0104]** In a more elaborate embodiment, which is pre-

ferred for advanced devices, this process can also be performed for a set of quantized IID/ICLD parameters selected from the repertoire of possible IID values from the quantizer. In that case, the comparison and selection procedure would comprise a comparison of processed IID and unprocessed IID parameters for various combinations of transmitted (quantized) IID parameters and smoothing time constants. Thus, as outlined by the square brackets in step 47, in contrast to the first embodiment, the second embodiment uses different quantization rules or the same quantization rules but different quantization step sizes to quantize the IID parameters. Then, in step 51, an error is calculated for each quantization way and each time constant. Thus, the number of candidates to be decided in step 52 compared to step 50 of Fig. 3c is, in the more elaborate embodiment, higher by a factor being equal to the number of different quantization ways compared to the first embodiment.

**[0105]** Then, in step 52, a two-dimensional optimization for (1) error and (2) bit rate is performed to search for a sequence of quantized values and a matching time constant. Finally, in step 53, the sequence of quantized values is entropy-encoded using a Huffman code or an arithmetic code. Step 53 finally results in a bit sequence to be transmitted to a decoder or multi-channel synthesizer.

**[0106]** Fig. 3b illustrates the effect of post processing by smoothing. Item 77 illustrates a quantized IID parameter for frame n. Item 78 illustrates a quantized IID parameter for a frame having a frame index n+1. The quantized IID parameter 78 has been derived by a quantization from the measured IID parameter per frame indicated by reference number 79. Smoothing of this parameter sequence of quantized parameter 77 and 78 with different time constants results in smaller post-processed parameter values at 80a and 80b. The time constant for smoothing the parameter sequence 77, 78, which resulted in the post-processed (smoothed) parameter 80a was smaller than the smoothing time constant, which resulted in a post-processed parameter 80b. As known in the art, the smoothing time constant is inverse to the cut-off frequency of a corresponding low-pass filter.

**[0107]** The embodiment illustrated in connection with steps 51 to 53 in Fig. 3c is preferable, since one can perform a two-dimensional optimization for error and bit rate, since different quantization rules may result in different numbers of bits for representing the quantized values. Furthermore, this embodiment is based on the finding that the actual value of the post-processed reconstruction parameter depends on the quantized reconstruction parameter as well as the way of processing.

**[0108]** For example, a large difference in (quantized) IID from frame to frame, in combination with a large smoothing time constant effectively results in only a small net effect of the processed IID. The same net effect may be constructed by a small difference in IID parameters, compared with a smaller time constant. This additional degree of freedom enables the encoder to optimize both

the reconstructed IID as well as the resulting bit rate simultaneously (given the fact that transmission of a certain IID value can be more expensive than transmission of a certain alternative IID parameter).

**[0109]** As outlined above, the effect on IID trajectories on the smoothing is outlined in Fig. 3b, which shows an IID trajectory for various values of smoothing time constants, where the star indicates a measured IID per frame, and where the triangle indicates a possible value of an IID quantizer. Given a limited accuracy of the IID quantizer, the IID value indicated by the star on frame n+1 is not available. The closest IID value is indicated by the triangle. The lines in the figure show the IID trajectory between the frames that would result from various smoothing constants. The selection algorithm will choose the smoothing time constant that results in an IID trajectory that ends closest to the measured IID parameter for frame n+1.

**[0110]** The examples above are all related to IID parameters. In principle, all described methods can also be applied to IPD, ITD, or ICC parameters.

**[0111]** The present invention, therefore, relates to an encoder-side processing and a decoder-side processing, which form a system using a smoothing enable/disable mask and a time constant signaled via a smoothing control signal. Furthermore, a band-wise signaling per frequency band is performed, wherein, furthermore, short cuts are preferred, which may include an all bands on, an all bands off or a repeat previous status short cut. Furthermore, it is preferred to use one common smoothing time constant for all bands. Furthermore, in addition or alternatively, a signal for automatic tonality-based smoothing versus explicit encoder control can be transmitted to implement a hybrid method.

**[0112]** Subsequently, reference is made to the decoder-side implementation, which works in connection with the encoder-guided parameter smoothing.

**[0113]** Fig. 4a shows an encoder-side 21 and a decoder-side 22. In the encoder, N original input channels are input into a down mixer stage 23. The down mixer stage is operative to reduce the number of channels to e.g. a single mono-channel or, possibly, to two stereo channels. The down mixed signal representation at the output of down mixer 23 is, then, input into a source encoder 24, the source encoder being implemented for example as an mp3 encoder or as an AAC encoder producing an output bit stream. The encoder-side 21 further comprises a parameter extractor 25, which, in accordance with the present invention, performs the BCC analysis (block 116 in Fig. 11) and outputs the quantized and preferably Huffman-encoded interchannel level differences (ICLD). The bit stream at the output of the source encoder 24 as well as the quantized reconstruction parameters output by parameter extractor 25 can be transmitted to a decoder 22 or can be stored for later transmission to a decoder, etc.

**[0114]** The decoder 22 includes a source decoder 26, which is operative to reconstruct a signal from the re-

ceived bit stream (originating from the source encoder 24). To this end, the source decoder 26 supplies, at its output, subsequent time portions of the input signal to an up-mixer 12, which performs the same functionality as the multi-channel reconstructor 12 in Fig. 1. Preferably, this functionality is a BCC synthesis as implemented by block 122 in Fig. 11.

**[0115]** Contrary to Fig. 11, the inventive multi-channel synthesizer further comprises the post processor 10 (Fig. 4a), which is termed as "interchannel level difference (ICLD) smoother", which is controlled by the input signal analyser 16, which preferably performs a tonality analysis of the input signal.

**[0116]** It can be seen from Fig. 4a that there are reconstruction parameters such as the interchannel level differences (ICLDs), which are input into the ICLD smoother, while there is an additional connection between the parameter extractor 25 and the up-mixer 12. Via this bypass connection, other parameters for reconstruction, which do not have to be post processed, can be supplied from the parameter extractor 25 to the up-mixer 12.

**[0117]** Fig. 4b shows a preferred embodiment of the signal-adaptive reconstruction parameter processing formed by the signal analyser 16 and the ICLD smoother 10.

**[0118]** The signal analyser 16 is formed from a tonality determination unit 16a and a subsequent thresholding device 16b. Additionally, the reconstruction parameter post processor 10 from Fig. 4a includes a smoothing filter 10a and a post processor switch 10b. The post processor switch 10b is operative to be controlled by the thresholding device 16b so that the switch is actuated, when the thresholding device 16b determines that a certain signal characteristic of the input signal such as the tonality characteristic is in a predetermined relation to a certain specified threshold. In the present case, the situation is such that the switch is actuated to be in the upper position (as shown in Fig. 4b), when the tonality of a signal portion of the input signal, and, in particular, a certain frequency band of a certain time portion of the input signal has a tonality above a tonality threshold. In this case, the switch 10b is actuated to connect the output of the smoothing filter 10a to the input of the multi-channel reconstructor 12 so that post processed, but not yet inversely quantized interchannel differences are supplied to the decoder/multi-channel reconstructor/up-mixer 12.

**[0119]** When, however, the tonality determination means in a decoder-controlled implementation determines that a certain frequency band of a actual time portion of the input signal, i.e., a certain frequency band of an input signal portion to be processed has a tonality lower than the specified threshold, i.e., is transient, the switch is actuated such that the smoothing filter 10a is by-passed.

**[0120]** In the latter case, the signal-adaptive post processing by the smoothing filter 10a makes sure that the reconstruction parameter changes for transient signals pass the post processing stage unmodified and re-

sult in fast changes in the reconstructed output signal with respect to the spatial image, which corresponds to real situations with a high degree of probability for transient signals.

**[0121]** It is to be noted here that the Fig. 4b embodiment, i.e., activating post processing on the one hand and fully deactivating post processing on the other hand, i.e., a binary decision for post processing or not is only a preferred embodiment because of its simple and efficient structure. Nevertheless, it has to be noted that, in particular with respect to tonality, this signal characteristic is not only a qualitative parameter but also a quantitative parameter, which can be normally between 0 and 1. In accordance with the quantitatively determined parameter, the smoothing degree of a smoothing filter or, for example, the cut-off frequency of a low pass filter can be set so that, for heavily tonal signals, a strong smoothing is activated, while for signals which are not so tonal, the smoothing with a lower smoothing degree is initiated.

**[0122]** Naturally, one could also detect transient portions and exaggerate the changes in the parameters to values between predefined quantized values or quantization indices so that, for strong transient signals, the post processing for the reconstruction parameters results in an even more exaggerated change of the spatial image of a multi-channel signal. In this case, a quantization step size of 1 as instructed by subsequent reconstruction parameters for subsequent time portions can be enhanced to for example 1.5, 1.4, 1.3 etc, which results in an even more dramatically changing spatial image of the reconstructed multi-channel signal.

**[0123]** It is to be noted here that a tonal signal characteristic, a transient signal characteristic or other signal characteristics are only examples for signal characteristics, based on which a signal analysis can be performed to control a reconstruction parameter post processor. In response to this control, the reconstruction parameter post processor determines a post processed reconstruction parameter having a value which is different from any values for quantization indices on the one hand or re-quantization values on the other hand as determined by a predetermined quantization rule.

**[0124]** It is to be noted here that post processing of reconstruction parameters dependent on a signal characteristic, i.e., a signal-adaptive parameter post processing is only optional. A signal-independent post processing also provides advantages for many signals. A certain post processing function could, for example, be selected by the user so that the user gets enhanced changes (in case of an exaggeration function) or damped changes (in case of a smoothing function). Alternatively, a post processing independent of any user selection and independent of signal characteristics can also provide certain advantages with respect to error resilience. It becomes clear that, especially in case of a large quantizer step size, a transmission error in a quantizer index may result in audible artefacts. To this end, one would perform a forward error correction or another similar operation,

when the signal has to be transmitted over error-prone channels. In accordance with the present invention, the post processing can obviate the need for any bit-inefficient error correction codes, since the post processing of the reconstruction parameters based on reconstruction parameters in the past will result in a detection of erroneous transmitted quantized reconstruction parameters and will result in suitable counter measures against such errors. Additionally, when the post processing function is a smoothing function, quantized reconstruction parameters strongly differing from former or later reconstruction parameters will automatically be manipulated as will be outlined later.

**[0125]** Fig. 5 shows a preferred embodiment of the reconstruction parameter post processor 10 from Fig. 4a. In particular, the situation is considered, in which the quantized reconstruction parameters are encoded. Here, the encoded quantized reconstruction parameters enter an entropy decoder 10c, which outputs the sequence of decoded quantized reconstruction parameters. The reconstruction parameters at the output of the entropy decoder are quantized, which means that they do not have a certain "useful" value but which means that they indicate certain quantizer indices or quantizer levels of a certain quantization rule implemented by a subsequent inverse quantizer. The manipulator 10d can be, for example, a digital filter such as an IIR (preferably) or a FIR filter having any filter characteristic determined by the required post processing function. A smoothing or low pass filtering post-processing function is preferred. At the output of the manipulator 10d, a sequence of manipulated quantized reconstruction parameters is obtained, which are not only integer numbers but which are any real numbers lying within the range determined by the quantization rule. Such a manipulated quantized reconstruction parameter could have values of 1.1, 0.1, 0.5, ..., compared to values 1, 0, 1 before stage 10d. The sequence of values at the output of block 10d are then input into an enhanced inverse quantizer 10e to obtain post-processed reconstruction parameters, which can be used for multi-channel reconstruction (e. g. BCC synthesis) in block 12 of Figs. 1a and 1b.

**[0126]** It has to be noted that the enhanced quantizer 10e (Fig. 5) is different from a normal inverse quantizer since a normal inverse quantizer only maps each quantization input from a limited number of quantization indices into a specified inversely quantized output value. Normal inverse quantizers cannot map non-integer quantizer indices. The enhanced inverse quantizer 10e is therefore implemented to preferably use the same quantization rule such as a linear or logarithmic quantization law, but it can accept non-integer inputs to provide output values which are different from values obtainable by only using integer inputs.

**[0127]** With respect to the present invention, it basically makes no difference, whether the manipulation is performed before requantization (see Fig. 5) or after requantization (see Fig. 6a, Fig. 6b). In the latter case, the in-

verse quantizer only has to be a normal straightforward inverse quantizer, which is different from the enhanced inverse quantizer 10e of Fig. 5 as has been outlined above. Naturally, the selection between Fig. 5 and Fig. 6a will be a matter of choice depending on the certain implementation. For the present implementation, the Fig. 5 embodiment is preferred, since it is more compatible with existing BCC algorithms. Nevertheless, this may be different for other applications.

**[0128]** Fig. 6b shows an embodiment in which the enhanced inverse quantizer 10e in Fig. 6a is replaced by a straightforward inverse quantizer and a mapper 10g for mapping in accordance with a linear or preferably non-linear curve. This mapper can be implemented in hardware or in software such as a circuit for performing a mathematical operation or as a look up table. Data manipulation using e.g. the smoother 10g can be performed before the mapper 10g or after the mapper 10g or at both places in combination. This embodiment is preferred, when the post processing is performed in the inverse quantizer domain, since all elements 10f, 10h, 10g can be implemented using straightforward components such as circuits of software routines.

**[0129]** Generally, the post processor 10 is implemented as a post processor as indicated in Fig. 7a, which receives all or a selection of actual quantized reconstruction parameters, future reconstruction parameters or past quantized reconstruction parameters. In the case, in which the post processor only receives at least one past reconstruction parameter and the actual reconstruction parameter, the post processor will act as a low pass filter. When the post processor 10, however, receives a future but delayed quantized reconstruction parameter, which is possible in real-time applications using a certain delay, the post processor can perform an interpolation between the future and the present or a past quantized reconstruction parameter to for example smooth a time-course of a reconstruction parameter, for example for a certain frequency band.

**[0130]** Fig. 7b shows an example implementation, in which the post processed value is not derived from the inversely quantized reconstruction parameter but from a value derived from the inversely quantized reconstruction parameter. The processing for deriving is performed by the means 700 for deriving which, in this case, can receive the quantized reconstruction parameter via line 702 or can receive an inversely quantized parameter via line 704. One could for example receive as a quantized parameter an amplitude value, which is used by the means for deriving for calculating an energy value. Then, it is this energy value which is subjected to the post processing (e.g. smoothing) operation. The quantized parameter is forwarded to block 706 via line 708. Thus, postprocessing can be performed using the quantized parameter directly as shown by line 710, or using the inversely quantized parameter as shown by line 712, or using the value derived from the inversely quantized parameter as shown by line 714.

**[0131]** As has been outlined above, the data manipulation to overcome artefacts due to quantization step sizes in a coarse quantization environment can also be performed on a quantity derived from the reconstruction parameter attached to the base channel in the parametrically encoded multi channel signal. When for example the quantized reconstruction parameter is a difference parameter (ICLD), this parameter can be inversely quantized without any modification. Then an absolute level value for an output channel can be derived and the inventive data manipulation is performed on the absolute value. This procedure also results in the inventive artefact reduction, as long as a data manipulation in the processing path between the quantized reconstruction parameter and the actual reconstruction is performed so that a value of the post processed reconstruction parameter or the post processed quantity is different from a value obtainable using requantization in accordance with the quantization rule, i.e. without manipulation to overcome the "step size limitation".

**[0132]** Many mapping functions for deriving the eventually manipulated quantity from the quantized reconstruction parameter are devisable and used in the art, wherein these mapping functions include functions for uniquely mapping an input value to an output value in accordance with a mapping rule to obtain a non post processed quantity, which is then post processed to obtain the postprocessed quantity used in the multi channel reconstruction (synthesis) algorithm.

**[0133]** In the following, reference is made to Fig. 8 to illustrate differences between an enhanced inverse quantizer 10e of Fig. 5 and a straightforward inverse quantizer 10f in Fig. 6a. To this end, the illustration in Fig. 8 shows, as a horizontal axis, an input value axis for non-quantized values. The vertical axis illustrates the quantizer levels or quantizer indices, which are preferably integers having a value of 0, 1, 2, 3. It has to be noted here that the quantizer in Fig. 8 will not result in any values between 0 and 1 or 1 and 2. Mapping to these quantizer levels is controlled by the stair-shaped function so that values between -10 and 10 for example are mapped to 0, while values between 10 and 20 are quantized to 1, etc.

**[0134]** A possible inverse quantizer function is to map a quantizer level of 0 to an inversely quantized value of 0. A quantizer level of 1 would be mapped to an inversely quantized value of 10. Analogously, a quantizer level of 2 would be mapped to an inversely quantized value of 20 for example. Requantization is, therefore, controlled by an inverse quantizer function indicated by reference number 31. It is to be noted that, for a straightforward inverse quantizer, only the crossing points of line 30 and line 31 are possible. This means that, for a straightforward inverse quantizer having an inverse quantizer rule of Fig. 8 only values of 0, 10, 20, 30 can be obtained by requantization.

**[0135]** This is different in the enhanced inverse quantizer 10e, since the enhanced inverse quantizer receives, as an input, values between 0 and 1 or 1 and 2 such as

value 0.5. The advanced requantization of value 0.5 obtained by the manipulator 10d will result in an inversely quantized output value of 5, i.e., in a post processed reconstruction parameter which has a value which is different from a value obtainable by requantization in accordance with the quantization rule. While the normal quantization rule only allows values of 0 or 10, the preferred inverse quantizer working in accordance with the preferred quantizer function 31 results in a different value, i.e., the value of 5 as indicated in Fig. 8.

**[0136]** While the straight-forward inverse quantizer maps integer quantizer levels to quantized levels only, the enhanced inverse quantizer receives non-integer quantizer "levels" to map these values to "inversely quantized values" between the values determined by the inverse quantizer rule.

**[0137]** Fig. 9 shows the impact of the preferred post processing for the Fig. 5 embodiment. Fig. 9a shows a sequence of quantized reconstruction parameters varying between 0 and 3. Fig. 9b shows a sequence of post processed reconstruction parameters, which are also termed as "modified quantizer indices", when the waveform in Fig. 9a is input into a low pass (smoothing) filter. It is to be noted here that the increases/decreases at time instance 1, 4, 6, 8, 9, and 10 are reduced in the Fig. 9b embodiment. It is to be noted with emphasis that the peak between time instant 8 and time instant 9, which might be an artefact is damped by a whole quantization step. The damping of such extreme values can, however, be controlled by a degree of post processing in accordance with a quantitative tonality value as has been outlined above.

**[0138]** The present invention is advantageous in that the inventive post processing smoothes fluctuations or smoothes short extreme values. The situation especially arises in a case, in which signal portions from several input channels having a similar energy are super-positioned in a frequency band of a signal, i.e., the base channel or input signal channel. This frequency band is then, per time portion and depending on the instant situation mixed to the respective output channels in a highly fluctuating manner. From the psycho-acoustic point of view, it would, however, be better to smooth these fluctuations, since these fluctuations do not contribute substantially to a detection of a location of a source but affect the subjective listening impression in a negative manner.

**[0139]** In accordance with a preferred embodiment of the present invention, such audible artefacts are reduced or even eliminated without incurring any quality losses at a different place in the system or without requiring a higher resolution/quantization (and, thus, a higher data rate) of the transmitted reconstruction parameters. The present invention reaches this object by performing a signal-adaptive modification (smoothing) of the parameters without substantially influencing important spatial localization detection cues.

**[0140]** The sudden occurring changes in the characteristic of the reconstructed output signal result in audible



artefacts in particular for audio signals having a highly constant stationary characteristic. This is the case with tonal signals. Therefore, it is important to provide a "smoother" transition between quantized reconstruction parameters for such signals. This can be obtained for example by smoothing, interpolation, etc.

**[0141]** Additionally, such a parameter value modification can introduce audible distortions for other audio signal types. This is the case for signals, which include fast fluctuations in their characteristic. Such a characteristic can be found in the transient part or attack of a percussive instrument. In this case, the embodiment provides for a deactivation of parameter smoothing.

**[0142]** This is obtained by post processing the transmitted quantized reconstruction parameters in a signal-adaptive way.

**[0143]** The adaptivity can be linear or non-linear. When the adaptivity is non-linear, a thresholding procedure as described in Fig. 3c is performed.

**[0144]** Another criterion for controlling the adaptivity is a determination of the stationarity of a signal characteristic. A certain form for determining the stationarity of a signal characteristic is the evaluation of the signal envelope or, in particular, the tonality of the signal. It is to be noted here that the tonality can be determined for the whole frequency range or, preferably, individually for different frequency bands of an audio signal.

**[0145]** This embodiment results in a reduction or even elimination of artefacts, which were, up to now, unavoidable, without incurring an increase of the required data rate for transmitting the parameter values.

**[0146]** As has been outlined above with respect to Figs. 4a and 4b, the preferred embodiment of the present invention in the decoder control mode performs a smoothing of interchannel level differences, when the signal portion under consideration has a tonal characteristic. Interchannel level differences, which are calculated in an encoder and quantized in an encoder are sent to a decoder for experiencing a signal-adaptive smoothing operation. The adaptive component is a tonality determination in connection with a threshold determination, which switches on the filtering of interchannel level differences for tonal spectral components, and which switches off such post processing for noise-like and transient spectral components. In this embodiment, no additional side information of an encoder are required for performing adaptive smoothing algorithms.

**[0147]** It is to be noted here that the inventive post processing can also be used for other concepts of parametric encoding of multi-channel signals such as for parametric stereo, MP3 surround, and similar methods.

**[0148]** The inventive methods or devices or computer programs can be implemented or included in several devices. Fig. 14 shows a transmission system having a transmitter including an inventive encoder and having a receiver including an inventive decoder. The transmission channel can be a wireless or wired channel. Furthermore, as shown in Fig. 15, the encoder can be in-

cluded in an audio recorder or the decoder can be included in an audio player. Audio records from the audio recorder can be distributed to the audio player via the Internet or via a storage medium distributed using mail or courier resources or other possibilities for distributing storage media such as memory cards, CDs or DVDs.

**[0149]** Depending on certain implementation requirements of the inventive methods, the inventive methods can be implemented in hardware or in software. The implementation can be performed using a digital storage medium, in particular a disk or a CD having electronically readable control signals stored thereon, which can cooperate with a programmable computer system such that the inventive methods are performed. Generally, the present invention is, therefore, a computer program product with a program code stored on a machine-readable carrier, the program code being configured for performing at least one of the inventive methods, when the computer program products runs on a computer. In other words, the inventive methods are, therefore, a computer program having a program code for performing the inventive methods, when the computer program runs on a computer.

**[0150]** While the foregoing has been particularly shown and described with reference to particular embodiments thereof, it will be understood by those skilled in the art that various other changes in the form and details may be made. It is to be understood that various changes may be made in adapting to different embodiments without departing from the broader concepts disclosed herein and comprehended by the claims that follow.

## Claims

1. Apparatus for generating an audio multi-channel synthesizer control signal, comprising:

a signal analyzer for analyzing an audio multi-channel input signal;

a smoothing information calculator for determining time smoothing control information in response to the signal analyzer, the smoothing information calculator being operative to determine the time smoothing control information such that, in response to the time smoothing control information, a separate synthesizer-side post-processor of an audio multi-channel synthesizer in accordance with claim 16 generates a post-processed reconstruction parameter or a post-processed quantity derived from the reconstruction parameter for a time portion of an input signal to be processed; and

a data generator for generating a control signal representing the time smoothing control information as the audio multi-channel synthesizer control signal.

2. Apparatus in accordance with claim 1, in which the signal analyzer is operative to analyze a change of an audio multi-channel signal characteristic from a first time portion of the audio multi-channel input signal to a later second time portion of the audio multi-channel input signal, and  
 5 in which the smoothing information calculator is operative to determine a smoothing time constant information based on the analyzed change.
3. Apparatus in accordance with claim 1, in which the signal analyzer is operative to perform band-wise analysis of the audio multi-channel input signal, and  
 10 in which the smoothing information calculator is operative to determine a band-wise time smoothing control information.
4. Apparatus in accordance with claim 3, in which the data generator is operative to output a time smoothing control mask having a bit for each frequency band, the bit for each frequency band indicating  
 20 whether the decoder-side post-processor is to perform smoothing over time or not.
5. Apparatus in accordance with claim 3, in which the data generator is operative to generate an all-off short cut signal indicating that no smoothing over time is to be carried out, or  
 25 to generate an all-on short cut signal indicating that smoothing over time is to be carried out in each frequency band, or  
 30 to generate a repeat last mask signal indicating that a band-wise status is to be used for a current time portion, which has already been used by the separate synthesizer-side post-processor for a preceding time portion.
6. Apparatus in accordance with claim 1, in which the data generator is operative to generate a synthesizer activation signal indicating, whether the separate synthesizer-side post-processor is to work using information transmitted in a data stream or using information derived from a separate synthesizer-side signal analysis.  
 40
7. Apparatus in accordance with claim 2, in which the data generator is operative to generate, as the time smoothing control information, a signal indicating a certain smoothing time constant value from a set of values known to the separate synthesizer-side post-processor.  
 50
8. Apparatus in accordance with claim 2, in which the signal analyzer is operative to determine, whether a point source exists, based on an inter-channel coherence parameter for an audio multi-channel input signal time portion, and  
 55 in which the smoothing information calculator or the data generator are only active, when the signal analyzer has determined that a point source exists.
9. Apparatus in accordance with claim 1, in which the smoothing information calculator is operative to calculate a change in a position of a point source for subsequent audio multi-channel input signal time portions, and  
 10 in which the data generator is operative to output a control signal indicating that the change in position is below a predetermined threshold so that smoothing over time is to be applied by the separate synthesizer-side post-processor.
10. Apparatus in accordance with claim 2, in which the signal analyzer is operative to generate an inter-channel level difference or inter-channel intensity difference for several time instants, and  
 20 in which the smoothing information calculator is operative to calculate a smoothing time constant, which is inversely proportional to a slope of a curve of the inter-channel level difference or inter-channel intensity difference parameters.
11. Apparatus in accordance with claim 2, in which the smoothing information calculator is operative to calculate a single smoothing time constant for a group of several frequency bands, and  
 25 in which the data generator is operative to indicate information for one or more bands in the group of several frequency bands, in which the separate synthesizer-side post-processor is to be deactivated.
12. Apparatus in accordance with claim 1, in which the smoothing information calculator is operative to perform an analysis by synthesis processing.  
 35
13. Apparatus in accordance with claim 12, in which the smoothing information calculator is operative to calculate several time constants,  
 40 to simulate a synthesizer-side post-processing using the several time constants,  
 to select a time constant, which results in values for subsequent frames, which shows the smallest deviation from non-quantized corresponding values.  
 45
14. Apparatus in accordance with claim 12, in which different test pairs are generated, in which a test pair has a smoothing time constant and a certain quantization rule, and  
 50 in which the smoothing information calculator is operative to select quantized values using a quantization rule and the smoothing time constant from the pair, which results in a smallest deviation between post-processed values and non-quantized corresponding values.
15. Method of generating at an audio encoder an audio

multi-channel synthesizer control signal, comprising:

analyzing an audio multi-channel input signal;  
 determining time smoothing control information  
 in response to the signal analyzing step, such  
 that, in response to the time smoothing control  
 information, a at a separate audio multi-channel  
 synthesizer for post-processing step of a method  
 generating an audio output signal from an  
 audio input signal generates a post-processed  
 reconstruction parameter or a post-processed  
 quantity derived from the reconstruction param-  
 eter for a time portion of an input signal to be  
 processed; and  
 generating a control signal representing the time  
 smoothing control information as the audio multi-  
 channel synthesizer control signal.

- 16.** Audio multi-channel synthesizer for generating an  
 audio output signal from an audio input signal, the  
 input signal having at least one audio input channel  
 and a sequence of quantized reconstruction param-  
 eters, the quantized reconstruction parameters be-  
 ing quantized in accordance with a quantization rule,  
 and being associated with subsequent time portions  
 of the audio input signal, the audio output signal hav-  
 ing a number of synthesized audio output channels,  
 and the number of synthesized audio output chan-  
 nels being greater than the number of audio input  
 channels, the audio input channel having associated  
 therewith an audio multi-channel synthesizer control  
 signal representing time smoothing control informa-  
 tion, comprising:

a control signal provider for providing the control  
 signal having the time smoothing control informa-  
 tion;  
 a post-processor for determining, in response  
 to the control signal, the post-processed recon-  
 struction parameter or the post-processed  
 quantity derived from the reconstruction param-  
 eter for a time portion of the input signal to be  
 processed by a smoothing operation over time,  
 wherein the post-processor is operative to de-  
 termine the post-processed reconstruction param-  
 eter or the post-processed quantity such  
 that the value of the post-processed reconstruc-  
 tion parameter or the post-processed quantity  
 is different from a value obtainable using re-  
 quantization in accordance with the quantization  
 rule; and  
 a multi-channel reconstructor for reconstructing  
 a time portion of the number of synthesized au-  
 dio output channels using the time portion of the  
 audio input channel and the post-processed re-  
 construction parameter or the post-processed  
 value.

**17.** Audio multi-channel synthesizer in accordance with  
 claim 16, in which the time smoothing control infor-  
 mation indicates a smoothing time constant, and  
 in which the post-processor is operative to perform  
 a low-pass filtering, wherein a filter characteristic is  
 set in response to the smoothing time constant.

**18.** Audio multi-channel synthesizer in accordance with  
 claim 16, in which the control signal includes time  
 smoothing control information for each band of a plu-  
 rality of bands of the at least one audio input channel,  
 and  
 in which the post-processor is operative to perform  
 post-processing in a band-wise manner in response  
 to the control signal.

**19.** Audio multi-channel synthesizer in accordance with  
 claim 16, in which the control signal includes a time  
 smoothing control mask having a bit for each fre-  
 quency band, the bit for each frequency band indi-  
 cating, whether the post-processor is to perform  
 smoothing over time or not, and  
 in which the post-processor is operative to perform  
 smoothing over time in response to the time smooth-  
 ing control mask, only when a bit for the frequency  
 band in the time smoothing control mask has a pre-  
 determined value.

**20.** Audio multi-channel synthesizer in accordance with  
 claim 16, in which the control signal includes an all-  
 off short cut signal, an all-on short cut signal or a  
 repeat last mask short cut signal, and  
 in which the post-processor is operative to perform  
 a smoothing operation over time, in response to the  
 all-off short cut signal, the all-on short cut signal or  
 the repeat last mask short cut signal.

**21.** Audio multi-channel synthesizer in accordance with  
 claim 16, in which the data signal includes a decoder  
 activation signal indicating, whether the post-proc-  
 essor is to work using information transmitted in the  
 data signal or using information derived from a de-  
 coder-side signal analysis, and  
 in which the post-processor is operative to work us-  
 ing the time smoothing control information or based  
 on a decoder-side signal analysis in response to the  
 control signal.

**22.** Audio multi-channel synthesizer in accordance with  
 claim 21, further comprising an input signal analyzer  
 for analyzing the audio input signal to determine a  
 signal characteristic of the time portion of the audio  
 input signal to be processed,  
 wherein the post-processor is operative to determine  
 the post-processed reconstruction parameter de-  
 pending on the signal characteristic,  
 wherein the signal characteristic is a tonality char-  
 acteristic or a transient characteristic of the portion

of the audio input signal to be processed.

23. Method of generating an audio output signal from an audio input signal, the audio input signal having at least one audio input channel and a sequence of quantized reconstruction parameters, the quantized reconstruction parameters being quantized in accordance with a quantization rule, and being associated with subsequent time portions of the audio input signal, the audio output signal having a number of synthesized audio output channels, and the number of synthesized audio output channels being greater than the number of audio input channels, the audio input signal having associated therewith an audio multi-channel synthesizer control signal representing time smoothing control information, comprising:

providing the control signal having the time smoothing control information;  
determining, in response to the control signal, the post-processed reconstruction parameter or the post-processed quantity derived from the reconstruction parameter for a time portion of the input signal to be processed by a smoothing operation over time; and  
reconstructing a time portion of the number of synthesized audio output channels using the time portion of the audio input channel and the post-processed reconstruction parameter or the post-processed value.

24. Audio multi-channel synthesizer control signal having time smoothing control information depending on an audio multi-channel input signal, the audio multi-channel synthesizer control signal being such that when input into an audio multi-channel synthesizer in accordance with claim 16, the post-processor of the audio multi-channel synthesizer generates, in response to the time smoothing control information, a post-processed reconstruction parameter or a post-processed quantity derived from the reconstruction parameter for a time portion of the audio input signal to be processed by a smoothing operation over time, which is different from a value obtainable using re-quantization in accordance with a quantization rule.
25. Audio multi-channel synthesizer control signal in accordance with claim 24, which is stored on a machine readable storage medium.
26. Transmitter or audio recorder having an apparatus for generating an audio multi-channel synthesizer control signal in accordance with claim 1.
27. Receiver or audio player having an audio multi-channel synthesizer in accordance with claim 16.

28. Transmission system having a transmitter and a receiver, the transmitter having an apparatus for generating an audio multi-channel synthesizer control signal in accordance with claim 1, and the receiver having an audio multi-channel synthesizer in accordance with claim 16.
29. Method of transmitting or audio recording, the method having a method of generating an audio multi-channel synthesizer control signal in accordance with claim 15.
30. Method of receiving or audio playing, the method including a method of generating an audio output signal from an audio input signal in accordance with claim 23.
31. Method of receiving and transmitting, the method including a transmitting method having a method of generating an audio multi-channel synthesizer control signal, in accordance with claim 15, and including a receiving method having a method of generating an audio output signal from an audio input signal in accordance with claim 23.
32. Computer program for performing, when running on a computer, a method in accordance with any one of method claims 15, 23, 29, 30 or 31.

#### Patentansprüche

1. Vorrichtung zum Erzeugen eines Audio-Multikanal-Synthesizer-Steuersignals, die folgende Merkmale aufweist:
- einen Signalanalysator zum Analysieren eines Audio-Multikanal-Eingangssignals;  
einen Glättungsinformationsrechner zum Bestimmen von Zeitglättungssteuerinformationen ansprechend auf den Signalanalysator, wobei der Glättungsinformationsrechner wirksam ist, um die Zeitglättungssteuerinformationen derart zu bestimmen, dass ansprechend auf die Zeitglättungssteuerinformationen ein separater Synthesizerseitiger Postprozessor eines Audio-Multikanal-Synthesizers gemäß Anspruch 16 einen nachverarbeiteten Rekonstruktionsparameter oder eine nachverarbeitete Größe erzeugt, hergeleitet aus dem Rekonstruktionsparameter für einen Zeitabschnitt eines Eingangssignals, das verarbeitet werden soll; und  
einen Datengenerator zum Erzeugen eines Steuersignals, das die Zeitglättungssteuerinformationen darstellt, als das Audio-Multikanal-Synthesizer-Steuersignal.

2. Vorrichtung gemäß Anspruch 1, bei der der Signalanalysator wirksam ist, um eine Änderung einer Audio-Multikanal-Signalcharakteristik von einem ersten Zeitabschnitt des Audio-Multikanal-Eingangssignals in einen späteren zweiten Zeitabschnitt des Audio-Multikanal-Eingangssignals zu ändern, und bei der der Glättungsinformationsrechner wirksam ist, um Glättungszeitkonstanten-Informationen basierend auf der analysierten Änderung zu bestimmen. 5
3. Vorrichtung gemäß Anspruch 1, bei der Signalanalysator wirksam ist, eine bandweise Analyse des Audio-Multikanal-Eingangssignals auszuführen, und bei der der Glättungsinformationsrechner wirksam ist, bandweise Zeitglättungssteuerinformationen zu bestimmen. 10
4. Vorrichtung gemäß Anspruch 3, bei der der Datengenerator wirksam ist, eine Zeitglättungssteuermaske auszugeben, mit einem Bit für jedes Frequenzband, wobei das Bit für jedes Frequenzband anzeigt, ob der decodierseitige Postprozessor eine Glättung über der Zeit ausführen soll oder nicht. 15
5. Vorrichtung gemäß Anspruch 3, bei der der Datengenerator wirksam ist, ein Alle-Aus-Abkürzungssignal zu erzeugen, das anzeigt, dass keine Glättung über der Zeit ausgeführt werden soll, oder ein Alle-Ein-Abkürzungssignal zu erzeugen, das anzeigt, dass eine Glättung über der Zeit in jedem Frequenzband ausgeführt werden soll, oder um ein Letzte-Maske-Wiederholen-Signal zu erzeugen, das anzeigt, dass ein Bandweise-Status verwendet werden soll für einen aktuellen Zeitabschnitt, der bereits durch den separaten Synthesizer-seitigen Postprozessor für einen vorangehenden Zeitabschnitt verwendet wurde. 20
6. Vorrichtung gemäß Anspruch 1, bei der der Datengenerator wirksam ist, um ein Synthesizer-Aktivierungssignal zu erzeugen, das anzeigt, ob der separate Synthesizer-seitige Postprozessor unter Verwendung von Informationen arbeiten soll, die in einem Datenstrom übertragen werden, oder unter Verwendung von Informationen, die aus einer separaten Synthesizer-seitigen Signalanalyse hergeleitet sind. 25
7. Vorrichtung gemäß Anspruch 2, bei der der Datengenerator wirksam ist, um als Zeitglättungssteuerinformationen ein Signal zu erzeugen, das einen bestimmten Glättungszeitkonstantenwert aus einem Satz von Werten anzeigt, die dem separaten Synthesizer-seitigen Postprozessor bekannt sind. 30
8. Vorrichtung gemäß Anspruch 2, bei der der Signalanalysator wirksam ist, um zu bestimmen, ob eine Punktquelle existiert, basierend auf einem Zwischenkanal-Kohärenzparameter für einen Audio-Multikanal-Eingangssignal-Zeitabschnitt, und bei der der Glättungsinformationsrechner oder der Datengenerator nur aktiv sind, wenn der Signalanalysator bestimmt hat, dass eine Punktquelle existiert. 35
9. Vorrichtung gemäß Anspruch 1, bei der der Glättungsinformationsrechner wirksam ist, um eine Änderung bei einer Position einer Punktquelle für nachfolgende Audio-Multikanal-Eingangssignal-Zeitabschnitte zu berechnen, und bei der der Datengenerator wirksam ist, ein Steuerungssignal auszugeben, das anzeigt, dass die Änderung der Position unter einer vorbestimmten Schwelle ist, so dass eine Glättung über der Zeit durch den separaten, Synthesizer-seitigen Postprozessor angewendet werden soll. 40
10. Vorrichtung gemäß Anspruch 2, bei der der Signalanalysator wirksam ist, eine Zwischenkanalpegeldifferenz oder Zwischenkanalintensitätsdifferenz für mehrere Zeitpunkte zu erzeugen, und bei der der Glättungsinformationsrechner wirksam ist, eine Glättungszeitkonstante zu berechnen, die umgekehrt proportional zu einer Steigung einer Kurve der Zwischenkanalpegeldifferenz- oder Zwischenkanalintensitätsdifferenz-Parameter ist. 45
11. Vorrichtung gemäß Anspruch 2, bei der der Glättungsinformationsrechner wirksam ist, eine einzelne Glättungszeitkonstante für eine Gruppe aus verschiedenen Frequenzbändern zu berechnen, und bei der der Datengenerator wirksam ist, Informationen für ein oder mehrere Bänder in der Gruppe aus verschiedenen Frequenzbändern anzuzeigen, in der der separate, Synthesizer-seitige Postprozessor deaktiviert sein soll. 50
12. Vorrichtung gemäß Anspruch 1, bei der der Glättungsinformationsrechner wirksam ist, eine Analyse durch Syntheseverarbeitung auszuführen. 55
13. Vorrichtung gemäß Anspruch 12, bei der der Glättungsinformationsrechner wirksam ist, um verschiedene Zeitkonstanten zu berechnen, um eine Synthesizer-seitige Nachverarbeitung unter Verwendung der verschiedenen Zeitkonstanten zu simulieren, eine Zeitkonstante auszuwählen, die zu Werten für aufeinanderfolgende Rahmen führt, die die kleinste Abweichung von nichtquantisierten, entsprechenden Werten zeigt.
14. Vorrichtung gemäß Anspruch 12, bei der unterschiedliche Testpaare erzeugt werden, wobei ein Testpaar eine Glättungszeitkonstante und eine bestimmte Quantisierungsregel aufweist, und bei der der Glättungsinformationsrechner wirksam

ist, quantisierte Werte unter Verwendung einer Quantisierungsregel und der Glättungszeitkonstante aus dem Paar auszuwählen, was zu einer kleinsten Abweichung zwischen nachverarbeiteten Werten und nichtquantisierten entsprechenden Werten führt.

15. Verfahren zum Erzeugen, an einem Audiocodierer, eines Audio-Multikanal-Synthesizer-Steuersignals, das folgende Schritte aufweist:

Analysieren eines Audio-Multikanal-Eingangssignals;  
Bestimmen von Zeitglättungssteuerinformationen ansprechend auf den Signalanalyseschritt, derart, dass ansprechend auf die Zeitglättungssteuerinformationen ein Nachverarbeitungsschritt eines Verfahrens an einem separaten Audio-Multikanal-Synthesizer zum Erzeugen eines Audioausgangssignals aus einem Audioeingangssignal einen nachverarbeiteten Rekonstruktionsparameter oder eine nachverarbeitete Größe erzeugt, hergeleitet aus dem Rekonstruktionsparameter für einen Zeitabschnitt eines Eingangssignals, das verarbeitet werden soll; und  
Erzeugen eines Steuersignals, das die Zeitglättungssteuerinformationen darstellt, als das Audio-Multikanal-Synthesizer-Steuersignal.

16. Audio-Multikanal-Synthesizer zum Erzeugen eines Audioausgangssignals aus einem Audioeingangssignal, wobei das Eingangssignal zumindest einen Audioeingangskanal und eine Sequenz aus quantisierten Rekonstruktionsparametern aufweist, wobei die quantisierten Rekonstruktionsparameter gemäß einer Quantisierungsregel quantisiert sind, und nachfolgenden Zeitabschnitten des Audioeingangssignals zugeordnet sind, wobei das Audioausgangssignal eine Anzahl von synthetisierten Audioausgangskanälen aufweist, und die Anzahl der synthetisierten Audioausgangskanäle größer ist als die Anzahl der Audioeingangskanäle, wobei der Audioeingangskanal zugeordnet zu demselben ein Audio-Multikanal-Synthesizer-Steuersignal aufweist, das Zeitglättungssteuerinformationen darstellt, der folgende Merkmale aufweist:

eine Steuersignalebereitstellungseinrichtung zum Bereitstellen des Steuersignals mit den Zeitglättungssteuerinformationen;  
einen Postprozessor zum Bestimmen, ansprechend auf das Steuersignal, des nachverarbeiteten Rekonstruktionsparameters oder der nachverarbeiteten Größe, hergeleitet aus dem Rekonstruktionsparameter für einen Zeitabschnitt des Eingangssignals, das durch eine Glättungsoperation über der Zeit verarbeitet

werden soll, wobei der Postprozessor wirksam ist, den nachverarbeiteten Rekonstruktionsparameter oder die nachverarbeitete Größe derart zu bestimmen, dass der Wert des nachverarbeiteten Rekonstruktionsparameters oder der nachverarbeiteten Größe unterschiedlich von einem Wert ist, der unter Verwendung einer Neuquantisierung gemäß der Quantisierungsregel erhalten werden kann; und  
einen Multikanalrekonstruierer zum Rekonstruieren eines Zeitabschnitts der Anzahl von synthetisierten Audioausgangskanälen unter Verwendung des Zeitabschnitts des Audioeingangskanals und des nachverarbeiteten Rekonstruktionsparameters oder des nachverarbeiteten Werts.

17. Audio-Multikanal-Synthesizer gemäß Anspruch 16, bei dem die Zeitglättungssteuerinformationen eine Glättungszeitkonstante anzeigen, und bei dem der Postprozessor wirksam ist, ein Tiefpassfiltern auszuführen, wobei eine Filtercharakteristik ansprechend auf die Glättungszeitkonstante eingestellt ist.

18. Audio-Multikanal-Synthesizer gemäß Anspruch 16, bei dem das Steuersignal Zeitglättungssteuerinformationen für jedes Band einer Mehrzahl von Bändern des zumindest einen Audioeingangskanals umfasst, und bei dem der Postprozessor wirksam ist, eine Nachverarbeitung bandweise ansprechend auf das Steuersignal auszuführen.

19. Audio-Multikanal-Synthesizer gemäß Anspruch 16, bei dem das Steuersignal eine Zeitglättungssteuermaske mit einem Bit für jedes Frequenzband umfasst, wobei das Bit für jedes Frequenzband anzeigt, ob der Postprozessor eine Glättung über der Zeit ausführen soll oder nicht, und bei dem der Postprozessor wirksam ist, eine Glättung über der Zeit ansprechend auf die Zeitglättungssteuermaske auszuführen, nur wenn ein Bit für das Frequenzband in der Zeitglättungssteuermaske einen vorbestimmten Wert aufweist.

20. Audio-Multikanal-Synthesizer gemäß Anspruch 16, bei dem das Steuersignal ein Alle-Aus-Abkürzungssignal, ein Alle-Ein-Abkürzungssignal oder ein Letzte-Maske-Wiederholen-Abkürzungssignal umfasst, und bei dem der Postprozessor wirksam ist, eine Glättungsoperation über der Zeit auszuführen, ansprechend auf das Alle-Aus-Abkürzungssignal, das Alle-Ein-Abkürzungssignal oder das Letzte-Maske-Wiederholen-Abkürzungssignal.

21. Audio-Multikanal-Synthesizer gemäß Anspruch 16,

bei dem das Datensignal ein Decodiereraktivierungssignal umfasst, das anzeigt, ob der Postprozessor unter Verwendung von Informationen arbeiten soll, die in dem Datensignal übertragen werden, oder unter Verwendung von Informationen, die aus einer Decodierer-seitigen Signalanalyse hergeleitet werden, und

bei dem der Postprozessor wirksam ist, unter Verwendung der Zeitglättungssteuerinformationen zu arbeiten oder basierend auf einer Decodierer-seitigen Signalanalyse ansprechend auf das Steuersignal.

- 22.** Audio-Multikanal-Synthesizer gemäß Anspruch 21, der ferner einen Eingangssignalanalysator aufweist zum Analysieren des Audioeingangssignals, um eine Signalcharakteristik des Zeitabschnitts des Audioeingangssignals zu bestimmen, das verarbeitet werden soll,

wobei der Postprozessor wirksam ist, den nachverarbeiteten Rekonstruktionsparameter abhängig von der Signalcharakteristik zu bestimmen, wobei die Signalcharakteristik eine Tonalitätscharakteristik oder eine Transientencharakteristik des Abschnitts des Audioeingangssignals ist, der verarbeitet werden soll.

- 23.** Verfahren zum Erzeugen eines Audioausgangssignals aus einem Audioeingangssignal, wobei das Audioeingangssignal zumindest einen Audioeingangskanal und eine Sequenz aus quantisierten Rekonstruktionsparametern aufweist, wobei die quantisierten Rekonstruktionsparameter gemäß einer Quantisierungsregel quantisiert sind, und nachfolgenden Zeitabschnitten des Audioeingangssignals zugeordnet sind, wobei das Audioausgangssignal eine Anzahl von synthetisierten Audioausgangskanälen aufweist, und die Anzahl der synthetisierten Audioausgangskanäle größer ist als die Anzahl der Audioeingangskanäle, wobei das Audioeingangssignal zugeordnet zu demselben ein Audio-Multikanal-Synthesizer-Steuersignal aufweist, das Zeitglättungssteuerinformationen darstellt, das folgende Schritte aufweist:

Bereitstellen des Steuersignals mit den Zeitglättungssteuerinformationen;

Bestimmen, ansprechend auf das Steuersignal, des nachverarbeiteten Rekonstruktionsparameters oder der nachverarbeiteten Größe, hergeleitet aus dem Rekonstruktionsparameter, für einen Zeitabschnitt des Eingangssignals, der durch eine Glättungsoperation über der Zeit verarbeitet werden soll; und

Rekonstruieren eines Zeitabschnitts der Anzahl von synthetisierten Audioausgangskanälen unter Verwendung des Zeitabschnitts des Audioeingangskanals und des nachverarbeiteten Re-

konstruktionsparameters oder nachverarbeiteten Werts.

- 24.** Audio-Multikanal-Synthesizer-Steuersignal mit Zeitglättungssteuerinformationen, abhängig von einem Audio-Multikanal-Eingangssignal, wobei das Audio-Multikanal-Synthesizer-Steuersignal derart ist, dass, wenn es in einen Audio-Multikanal-Synthesizer gemäß Anspruch 16 eingegeben wird, der Postprozessor des Audio-Multikanal-Synthesizers ansprechend auf die Zeitglättungssteuerinformationen einen nachverarbeiteten Rekonstruktionsparameter oder eine nachverarbeitete Größe erzeugt, hergeleitet aus dem Rekonstruktionsparameter, für einen Zeitabschnitt des Audioeingangssignals, das durch eine Glättungsoperation über der Zeit verarbeitet werden soll, die sich von einem Wert unterscheidet, die unter Verwendung einer Neuquantisierung gemäß einer Quantisierungsregel erhalten werden kann.

- 25.** Audio-Multikanal-Synthesizer-Steuersignal gemäß Anspruch 24, das auf einem maschinenlesbaren Speicherungsmedium gespeichert ist.

- 26.** Sender oder Audioaufzeichneinrichtung mit einer Vorrichtung zum Erzeugen eines Audio-Multikanal-Synthesizer-Steuersignals gemäß Anspruch 1.

- 27.** Empfänger oder Audioabspielgerät mit einem Audio-Multikanal-Synthesizer gemäß Anspruch 16.

- 28.** Sendesystem mit einem Sender und einem Empfänger, wobei der Sender eine Vorrichtung zum Erzeugen eines Audio-Multikanal-Synthesizer-Steuersignals gemäß Anspruch 1 aufweist, und der Empfänger einen Audio-Multikanal-Synthesizer gemäß Anspruch 16 aufweist.

- 29.** Verfahren zum Senden oder Audioaufzeichnen, wobei das Verfahren ein Verfahren zum Erzeugen eines Audio-Multikanal-Synthesizer-Steuersignals gemäß Anspruch 15 aufweist.

- 30.** Verfahren zum Empfangen oder Audioabspielen, wobei das Verfahren ein Verfahren zum Erzeugen eines Audioausgangssignals aus einem Audioeingangssignal gemäß Anspruch 23 umfasst.

- 31.** Verfahren zum Empfangen und Senden, wobei das Verfahren ein Sendeverfahren umfasst, das ein Verfahren aufweist zum Erzeugen eines Audio-Multikanal-Synthesizer-Steuersignals gemäß Anspruch 15, und ein Empfangsverfahren umfasst, das ein Verfahren aufweist zum Erzeugen eines Audioausgangssignals aus einem Audioeingangssignal gemäß An-

spruch 23.

32. Computerprogramm zum Ausführen, wenn es auf einem Computer läuft, eines Verfahrens gemäß einem der Verfahrensansprüche 15, 23, 29, 30 oder 31.

### Revendications

1. Appareil pour générer un signal de commande de synthétiseur multicanal audio, comprenant:

un analyseur de signal destiné à analyser un signal d'entrée multicanal audio;  
un calculateur d'informations de lissage destiné à déterminer les informations de commande de lissage dans le temps en réponse à l'analyseur de signal, le calculateur d'informations de lissage étant opérationnel pour déterminer les informations de commande de lissage dans le temps de sorte que, en réponse aux informations de commande de lissage dans le temps, un post-processeur du côté du synthétiseur séparé d'un synthétiseur multicanal audio selon la revendication 16 génère un paramètre de reconstruction post-traité ou une quantité post-traitée dérivée du paramètre de reconstruction pour une partie temporelle d'un signal d'entrée à traiter; et un générateur de données destiné à générer un signal de commande représentant les informations de commande de lissage dans le temps comme signal de commande de synthétiseur multicanal audio.

2. Appareil selon la revendication 1, dans lequel l'analyseur de signal est opérationnel pour analyser une modification d'une caractéristique de signal multicanal audio d'une première partie temporelle du signal d'entrée multicanal audio à une deuxième partie temporelle ultérieure du signal d'entrée multicanal audio, et dans lequel le calculateur d'informations de lissage est opérationnel pour déterminer une information de constante de temps de lissage sur base de la modification analysée.
3. Appareil selon la revendication 1, dans lequel l'analyseur de signal est opérationnel pour réaliser une analyse par bande du signal d'entrée multicanal audio, et dans lequel le calculateur d'informations de lissage est opérationnel pour déterminer une information de commande de lissage dans le temps par bande.
4. Appareil selon la revendication 3, dans lequel le générateur de données est opérationnel pour sortir un masque de commande de lissage dans le temps

ayant un bit pour chaque bande de fréquences, le bit pour chaque bande de fréquences indiquant si le post-processeur du côté du codeur doit réaliser un lissage dans le temps ou non.

5. Appareil selon la revendication 3, dans lequel le générateur de données est opérationnel pour générer un signal de court-circuit tout désactivé indiquant qu'il ne doit pas être réalisé de lissage dans le temps, ou pour générer un signal de court-circuit tout activé indiquant qu'un lissage dans le temps doit être réalisé dans chaque bande de fréquences, ou pour générer un signal de répétition du dernier masque indiquant qu'un statut par bande doit être utilisé pour une partie temporelle actuelle qui a déjà été utilisée par le post-processeur du côté du synthétiseur séparé pour une partie temporelle précédente.
6. Appareil selon la revendication 1, dans lequel le générateur de données est opérationnel pour générer un signal d'activation de synthétiseur indiquant si le post-processeur du côté du synthétiseur séparé doit fonctionner à l'aide des informations transmises dans un flux de données ou à l'aide d'informations dérivées d'une analyse de signal du côté du synthétiseur séparé.
7. Appareil selon la revendication 2, dans lequel le générateur de données est opérationnel pour générer, comme informations de commande de lissage dans le temps, un signal indiquant une certaine valeur de constante de temps de lissage parmi un ensemble de valeurs connues du post-processeur du côté du synthétiseur séparé.
8. Appareil selon la revendication 2, dans lequel l'analyseur est opérationnel pour déterminer s'il existe une source de points, sur base d'un paramètre de cohérence entre canaux pour une partie temporelle du signal d'entrée multicanal audio, et dans lequel le calculateur d'informations de lissage ou le générateur de données ne sont actifs que si l'analyseur de signal a déterminé qu'il existe une source de points.
9. Appareil selon la revendication 1, dans lequel le calculateur d'informations de lissage est opérationnel pour calculer une variation dans une position d'une source de points pour les parties temporelles du signal d'entrée multicanal audio successives, et dans lequel le générateur de données est opérationnel pour sortir un signal de contrôle indiquant que la variation de position est inférieure à un seuil prédéterminé, de sorte que le lissage dans le temps doit être appliqué par le post-processeur du côté du synthétiseur séparé.



10. Appareil selon la revendication 2, dans lequel l'analyseur de signal est opérationnel pour générer une différence de niveau entre canaux ou une différence d'intensité entre canaux pour différents moments, et dans lequel le calculateur d'informations de lissage est opérationnel pour calculer une constante de temps de lissage qui est inversement proportionnelle à une pente d'une courbe des paramètres de différence de niveau entre canaux ou de différence d'intensité entre canaux.
11. Appareil selon la revendication 2, dans lequel le calculateur d'informations de lissage est opérationnel pour calculer une seule constante de temps de lissage pour un groupe de plusieurs bandes de fréquences, et dans lequel le générateur de données est opérationnel pour indiquer les informations pour une ou plusieurs bandes dans le groupe de plusieurs bandes de fréquences dans lequel le post-processeur du côté du synthétiseur séparé doit être désactivé.
12. Appareil selon la revendication 1, dans lequel le calculateur d'informations de lissage est opérationnel pour effectuer une analyse par traitement de synthèse.
13. Appareil selon la revendication 12, dans lequel le calculateur d'informations de lissage est opérationnel pour calculer plusieurs constantes de temps, pour simuler un post-traitement du côté du synthétiseur à l'aide de plusieurs constantes de temps, pour sélectionner une constante de temps qui résulte en des valeurs pour des trames successives qui présentent la déviation la plus petite par rapport aux valeurs correspondantes non quantifiées.
14. Appareil selon la revendication 12, dans lequel différentes paires de test sont générées, dans lequel une paire de test présente une constante de temps de lissage et une certaine règle de quantification, et dans lequel le calculateur d'informations de lissage est opérationnel pour sélectionner des valeurs quantifiées à l'aide d'une règle de quantification et la constante de temps de lissage de la paire qui résulte en la déviation la plus petite entre les valeurs post-traitées et les valeurs correspondantes non quantifiées.
15. Procédé pour générer dans un codeur audio un signal de commande de synthétiseur multicanal audio, comprenant:
- analyser un signal d'entrée multicanal audio; déterminer les informations de commande de lissage en réponse à l'étape d'analyse de signal de sorte que, en réponse aux informations de commande de lissage dans le temps, une étape de post-traitement d'un procédé dans un synthétiseur multicanal audio séparé pour générer un signal de sortie audio à partir d'un signal d'entrée audio génère un paramètre de reconstruction post-traité ou une quantité post-traitée dérivée du paramètre de reconstruction pour une partie temporelle d'un signal d'entrée à traiter; et générer un signal de commande représentant les informations de commande de lissage dans le temps comme signal de commande de synthétiseur multicanal audio.
16. Synthétiseur multicanal audio pour générer un signal de sortie audio à partir d'un signal d'entrée audio, le signal d'entrée présentant au moins un canal d'entrée audio et une séquence de paramètres de reconstruction quantifiés, les paramètres de reconstruction quantifiés étant associés à des parties temporelles successives du signal d'entrée audio, le signal de sortie audio présentant un nombre de canaux de sortie audio synthétisés, et le nombre de canaux de sortie audio synthétisés étant supérieur au nombre de canaux d'entrée audio, le canal d'entrée audio présentant, y associé, un signal de commande de synthétiseur multicanal audio représentant les informations de commande de lissage dans le temps, comprenant:
- un fournisseur de signal de commande destiné à fournir le signal de commande présentant les informations de commande de lissage dans le temps;
- un post-processeur destiné à déterminer, en réponse au signal de commande, le paramètre de reconstruction post-traité ou la quantité post-traitée dérivée du paramètre de reconstruction pour une partie temporelle de signal d'entrée à traiter par une opération de lissage dans le temps, où le post-processeur est opérationnel pour déterminer le paramètre de reconstruction post-traité ou la quantité post-traitée de sorte que la valeur du paramètre de reconstruction post-traité ou la quantité post-traitée soit différente d'une valeur pouvant être obtenue à l'aide d'une requantification selon la règle de quantification; et
- un reconstituteur multicanal destiné à reconstruire une partie temporelle du nombre de canaux de sortie audio synthétisés à l'aide de la partie temporelle du canal d'entrée audio et du paramètre de reconstruction post-traité ou de la valeur post-traitée.
17. Synthétiseur multicanal audio selon la revendication 16, dans lequel les informations de commande de lissage dans le temps indiquent une constante de temps de lissage, et

- dans lequel le post-processeur est opérationnel pour réaliser une filtration passe-bas, où une caractéristique de filtre est réglée en réponse à la constante de temps de lissage.
18. Synthétiseur multicanal audio selon la revendication 16, dans lequel le signal de commande comporte des informations de commande de lissage dans le temps pour chaque bande d'une pluralité de bandes de l'au moins un canal d'entrée audio, et dans lequel le post-processeur est opérationnel pour réaliser le post-traitement par bande en réponse au signal de commande.
19. Synthétiseur multicanal audio selon la revendication 16, dans lequel le signal de commande comporte un masque de commande de lissage dans le temps ayant un bit pour chaque bande de fréquences, le bit pour chaque bande de fréquences indiquant si le post-processeur doit ou non réaliser un lissage dans le temps, et dans lequel le post-processeur est opérationnel pour réaliser le lissage dans le temps en réponse au masque de commande de lissage dans le temps uniquement lorsqu'un bit pour la bande de fréquences dans le masque de commande de lissage dans le temps a une valeur prédéterminée.
20. Synthétiseur multicanal audio selon la revendication 16, dans lequel le signal de commande comporte un signal de court-circuit tout désactivé, un signal de court-circuit tout activé ou un signal de court-circuit de répétition du dernier masque, et dans lequel le post-processeur est opérationnel pour réaliser une opération de lissage dans le temps en réponse au signal de court-circuit tout désactivé, au signal de court-circuit tout activé ou au signal de court-circuit de répétition du dernier masque.
21. Synthétiseur multicanal audio selon la revendication 16, dans lequel le signal de données comporte un signal d'activation de décodeur indiquant si le post-processeur doit fonctionner à l'aide des informations transmises dans le signal de données ou à l'aide des informations dérivées d'une analyse de signal du côté du décodeur, et dans lequel le post-processeur est opérationnel pour fonctionner à l'aide des informations de commande de lissage dans le temps ou sur base d'une analyse de signal du côté du décodeur en réponse au signal de commande.
22. Synthétiseur multicanal audio selon la revendication 21, comprenant par ailleurs un analyseur de signal d'entrée destiné à analyser le signal d'entrée audio pour déterminer une caractéristique de signal de la partie temporelle du signal d'entrée audio à traiter, dans lequel le post-processeur est opérationnel pour
- déterminer le paramètre de reconstruction post-traité en fonction de la caractéristique de signal, dans lequel la caractéristique de signal est une caractéristique de tonalité ou une caractéristique de transitoires de la partie du signal d'entrée audio à traiter.
23. Procédé pour générer un signal de sortie audio à partir d'un signal d'entrée audio, le signal d'entrée audio présentant au moins un canal d'entrée audio et une séquence de paramètres de reconstruction quantifiés, les paramètres de reconstruction quantifiés étant quantifiés selon une règle de quantification, et étant associés à de parties temporelles successives du signal d'entrée audio, le signal de sortie audio présentant un nombre de canaux de sortie audio synthétisés, et le nombre de canaux de sortie audio synthétisés étant supérieur au nombre de canaux d'entrée audio, le signal d'entrée audio présentant, y associé, un signal de commande de synthétiseur multicanal audio représentant les informations de commande de lissage dans le temps, comprenant:
- fournir le signal de commande présentant les informations de commande de lissage dans le temps;
- déterminer, en réponse au signal de commande, le paramètre de reconstruction post-traité ou la quantité post-traitée dérivée du paramètre de reconstruction pour une partie temporelle du signal d'entrée à traiter par une opération de lissage dans le temps; et
- reconstruire une partie temporelle du nombre de canaux de sortie audio synthétisés à l'aide de la partie temporelle du canal d'entrée audio et du paramètre de reconstruction post-traité ou de la valeur post-traitée.
24. Signal de commande de synthétiseur multicanal audio présentant des informations de commande de lissage dans le temps en fonction d'un signal d'entrée multicanal audio, le signal de commande de synthétiseur multicanal audio étant tel que, lorsqu'il est entré dans un synthétiseur multicanal audio selon la revendication 16, le post-processeur du synthétiseur multicanal audio génère, en réponse aux informations de commande de lissage dans le temps, un paramètre de reconstruction post-traité ou une quantité post-traitée dérivée du paramètre de reconstruction pour une partie temporelle du signal d'entrée audio à traiter par une opération de lissage dans le temps qui est différente d'une valeur pouvant être obtenue à l'aide d'une requantification selon une règle de quantification.
25. Signal de commande de synthétiseur multicanal selon la revendication 24, qui est mémorisé sur un sup-

port de mémoire lisible en machine.

- 26.** Emetteur ou enregistreur audio présentant un appareil pour générer un signal de commande de synthétiseur multicanal audio selon la revendication 1. 5
- 27.** Récepteur ou reproducteur audio présentant un synthétiseur multicanal audio selon la revendication 16.
- 28.** Système de transmission présentant un émetteur et un récepteur, 10  
l'émetteur présentant un appareil pour générer un signal de commande de synthétiseur multicanal audio selon la revendication 1, et  
le récepteur présentant un synthétiseur multicanal audio selon la revendication 16. 15
- 29.** Procédé pour émettre ou enregistrer audio, le procédé présentant un procédé de génération d'un signal de commande de synthétiseur multicanal audio selon la revendication 15. 20
- 30.** Procédé pour recevoir ou reproduire audio, le procédé comportant un procédé pour générer un signal de sortie audio à partir d'un signal d'entrée audio selon la revendication 23. 25
- 31.** Procédé pour recevoir et émettre, le procédé comportant un procédé d'émission présentant un procédé de génération d'un signal de commande de synthétiseur multicanal audio selon la revendication 15, 30  
et  
comportant un procédé de réception présentant un procédé pour générer un signal de sortie audio à partir d'un signal d'entrée audio selon la revendication 23. 35
- 32.** Programme d'ordinateur pour réaliser, lorsqu'il est exécuté sur un ordinateur, un procédé selon l'une quelconque des revendications de procédé 15, 23, 29, 30 ou 31. 40

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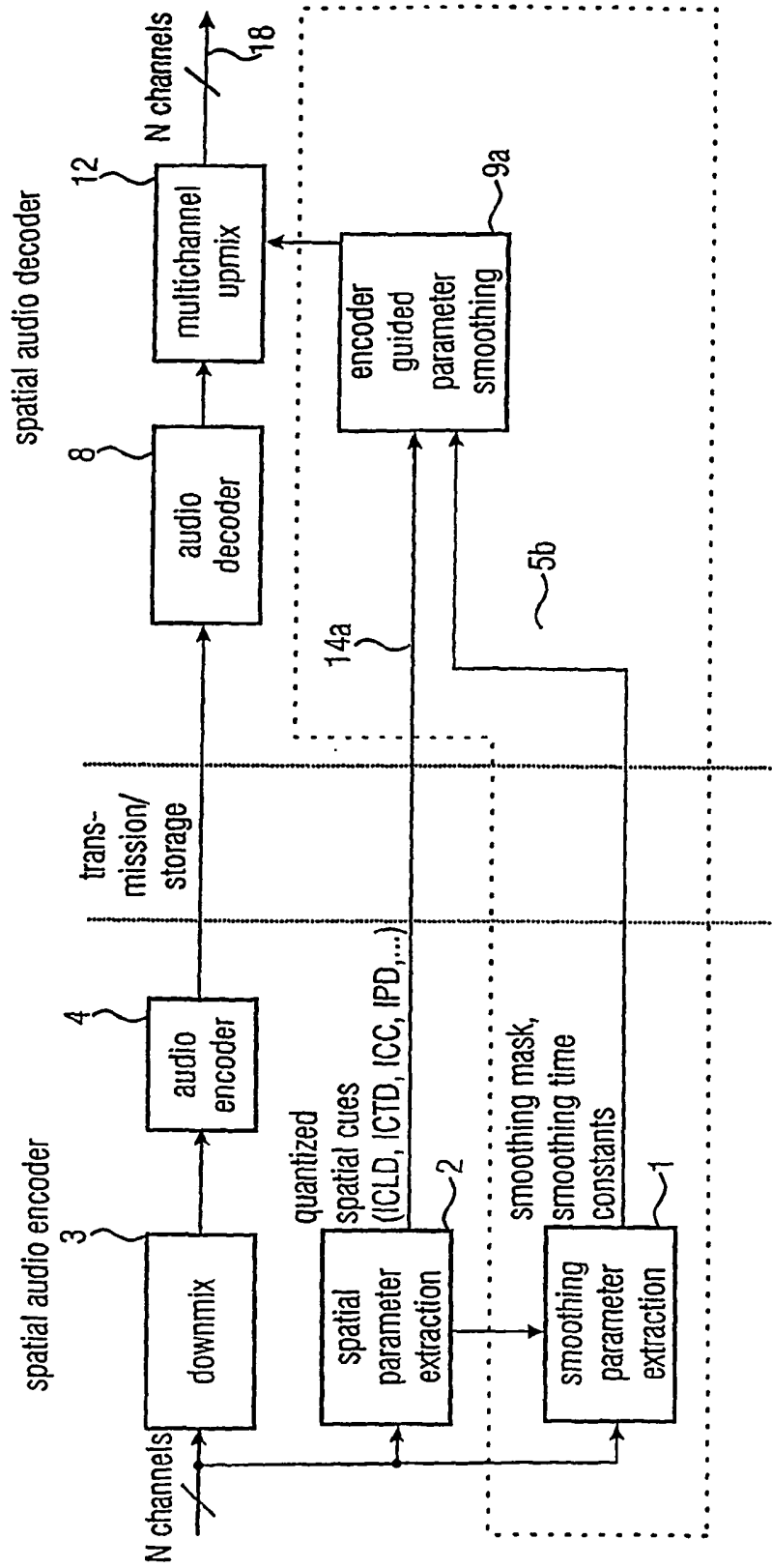


FIGURE 1A

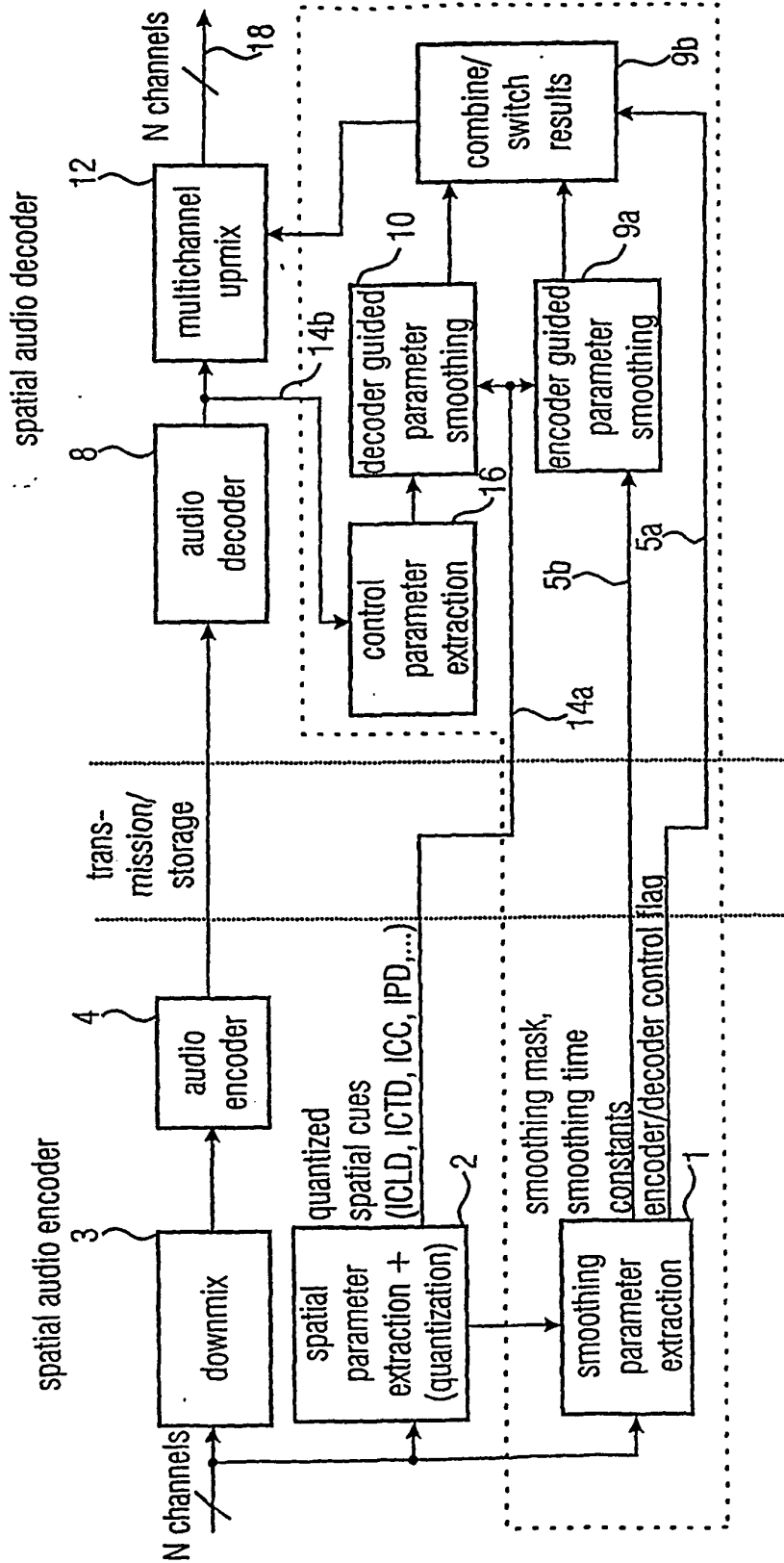


FIGURE 1B

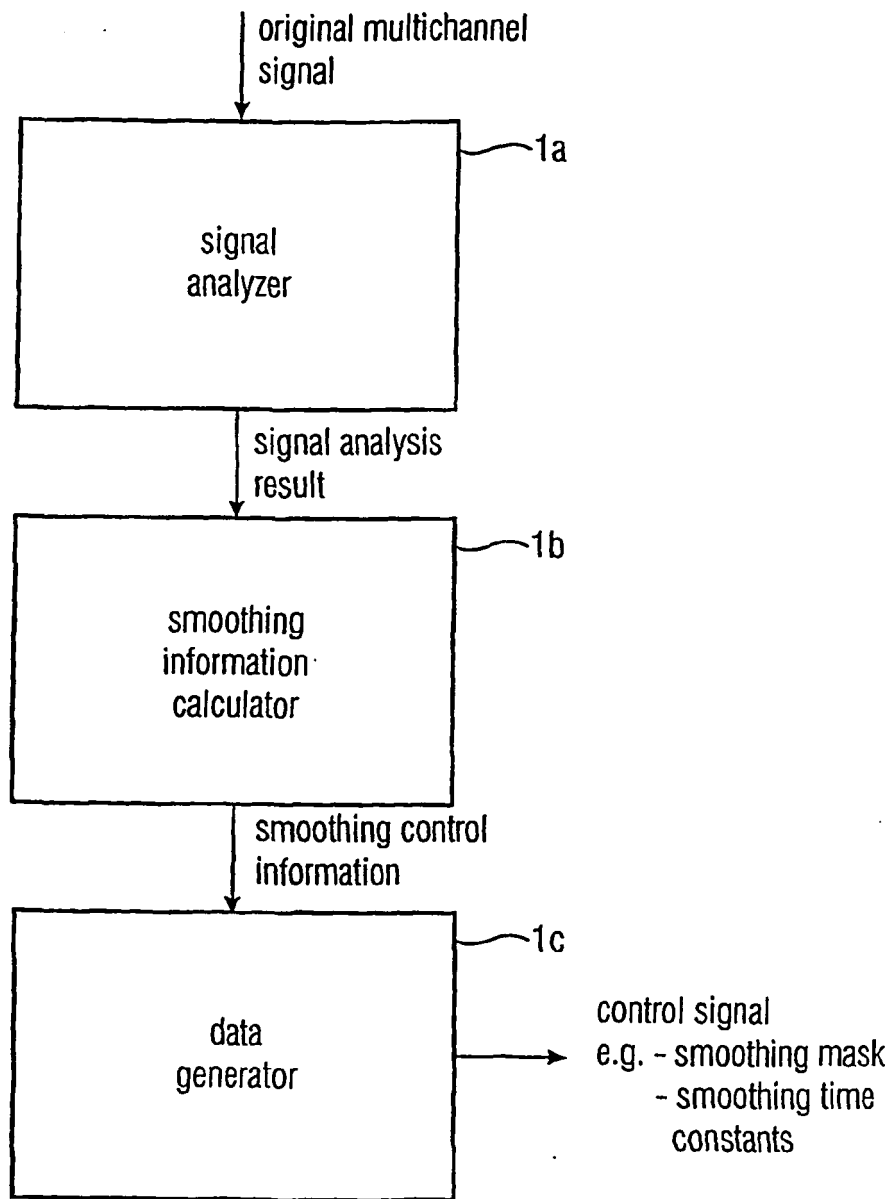
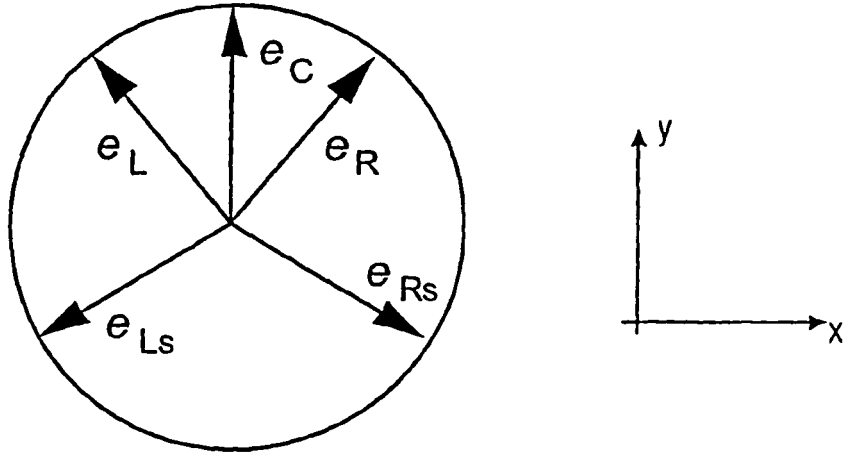


FIGURE 1C



spatial  
position

$$p = \frac{L \cdot e_L + C \cdot e_C + R \cdot e_R + Ls \cdot e_{Ls} + Rs \cdot e_{Rs}}{L + C + R + Ls + Rs}$$

FIGURE 2A

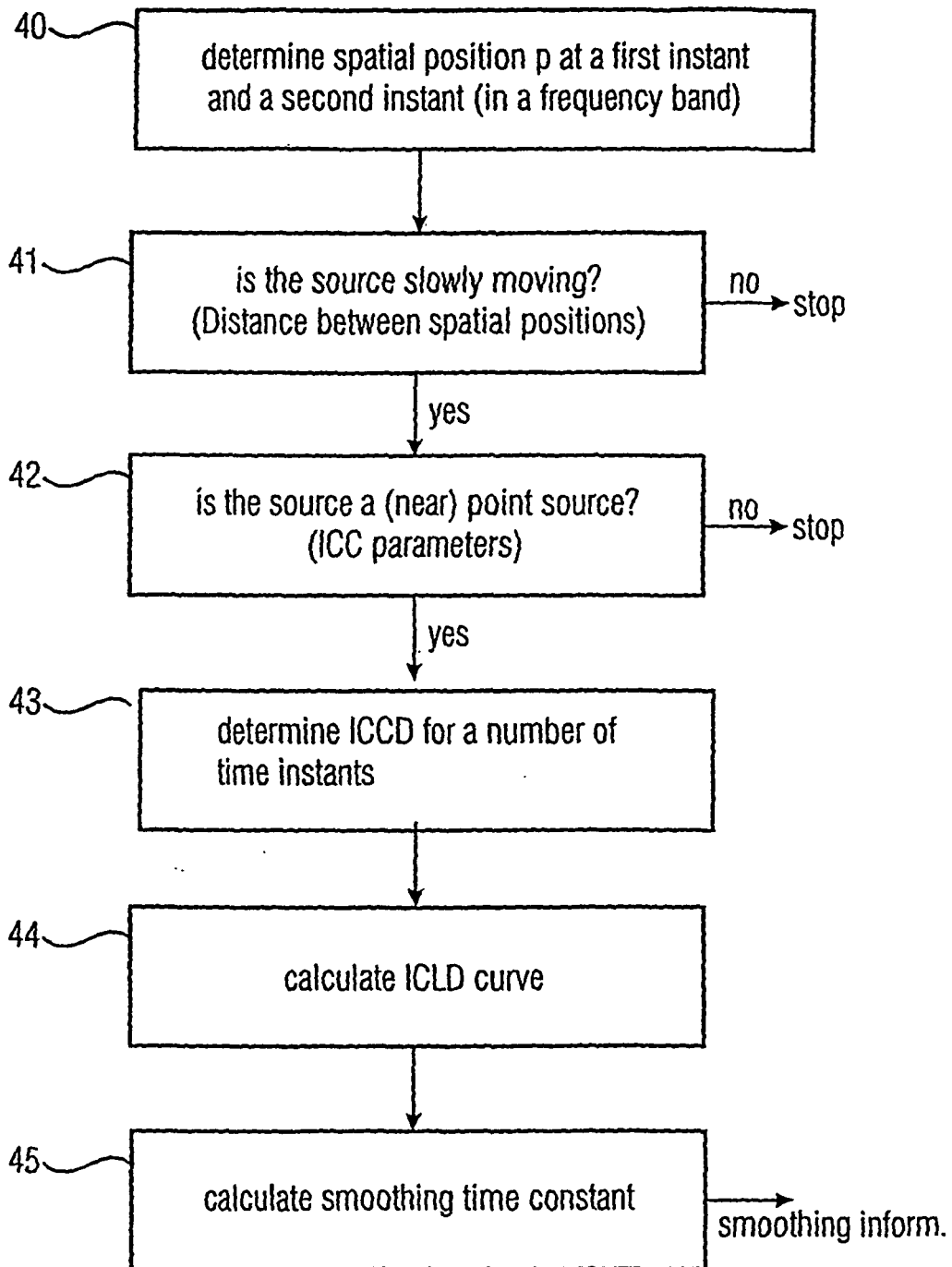


FIGURE 2B



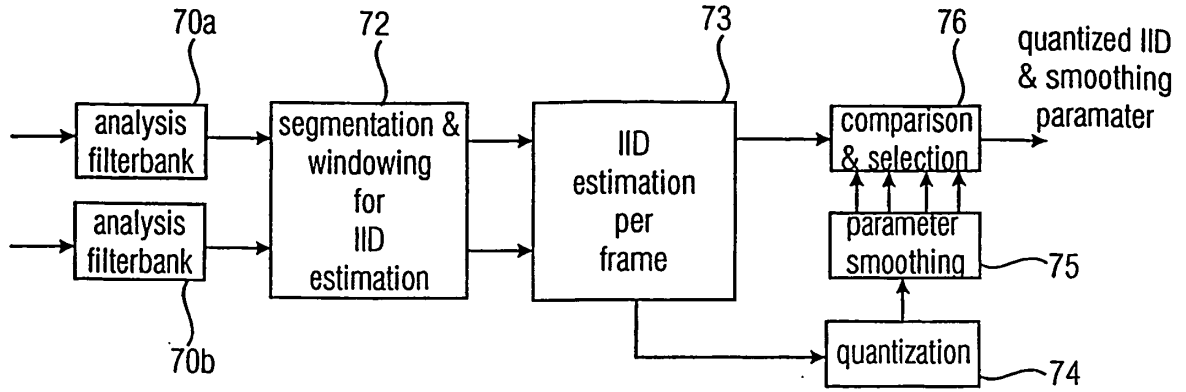


FIGURE 3A

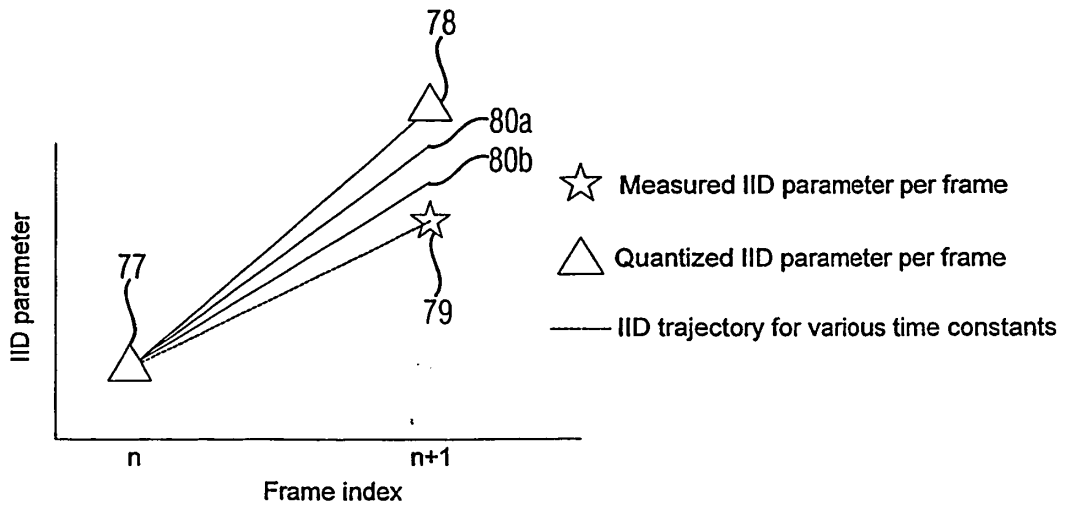


FIGURE 3B

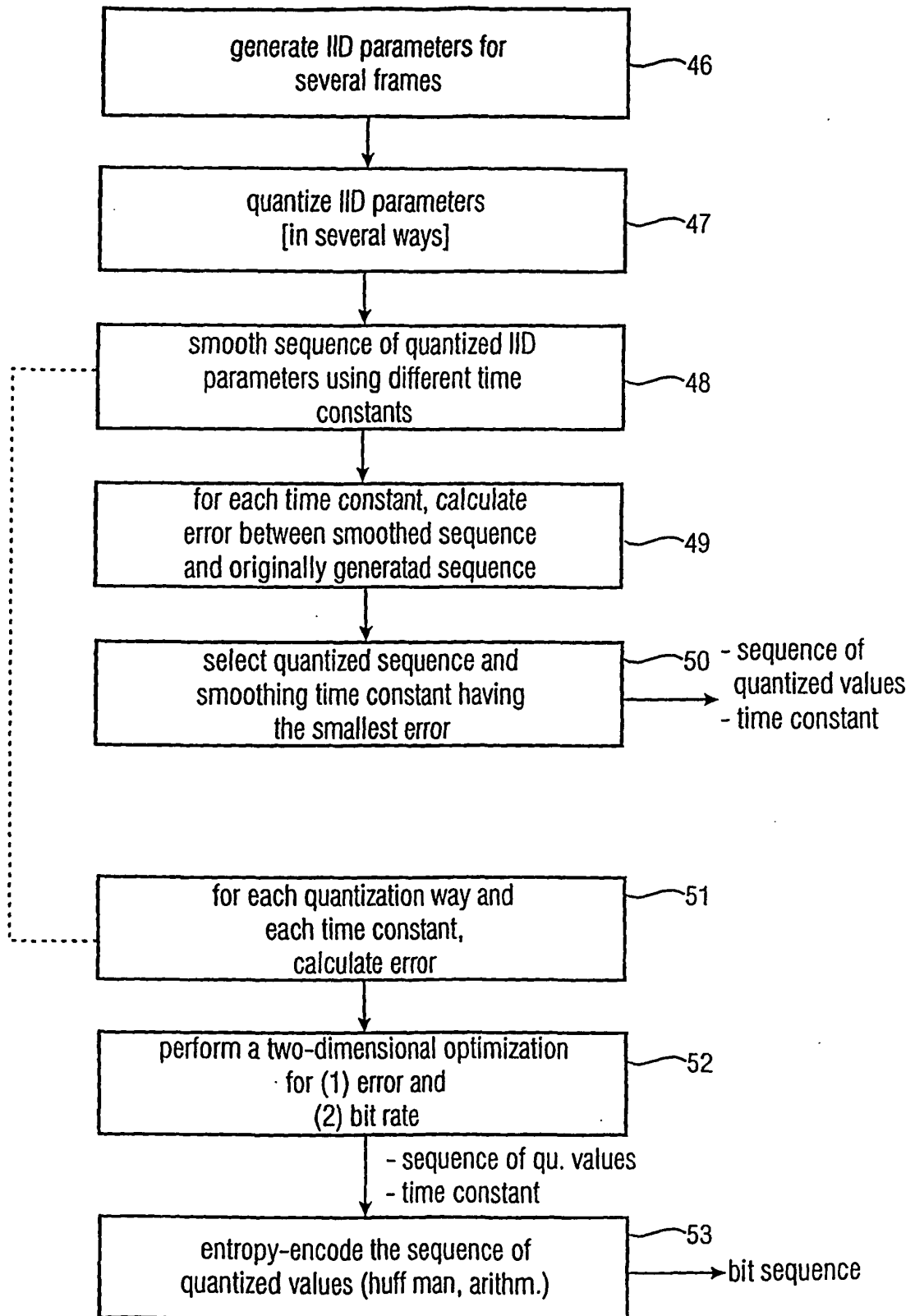


FIGURE 3C

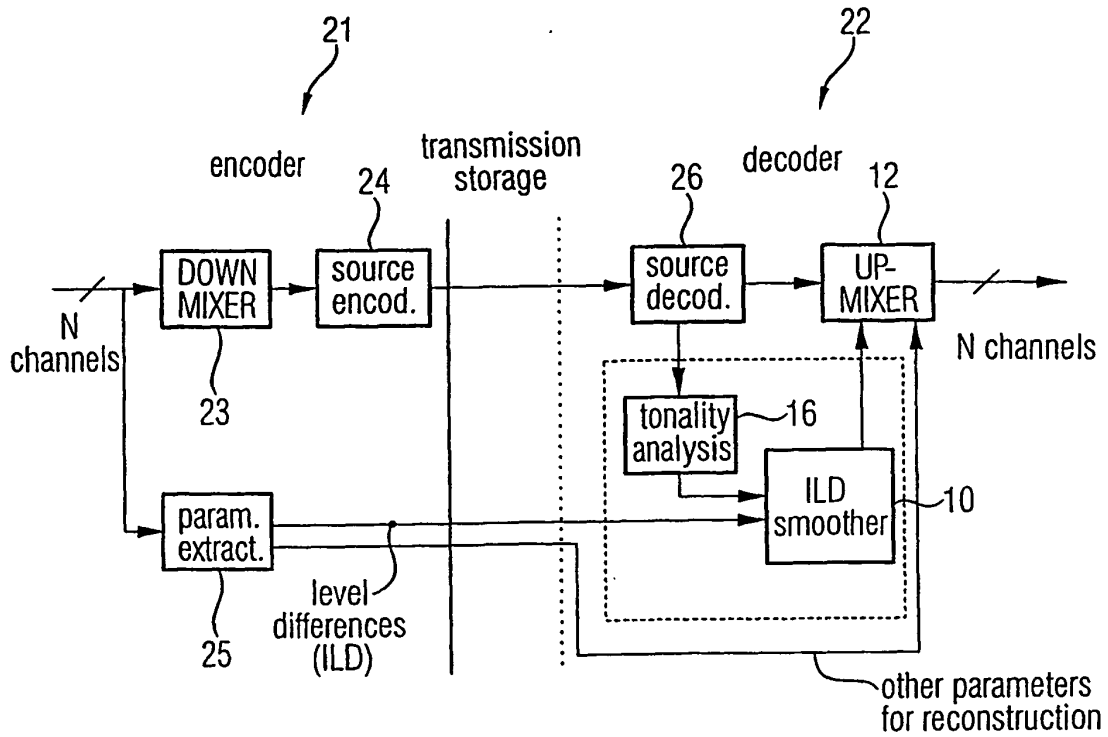


FIGURE 4A

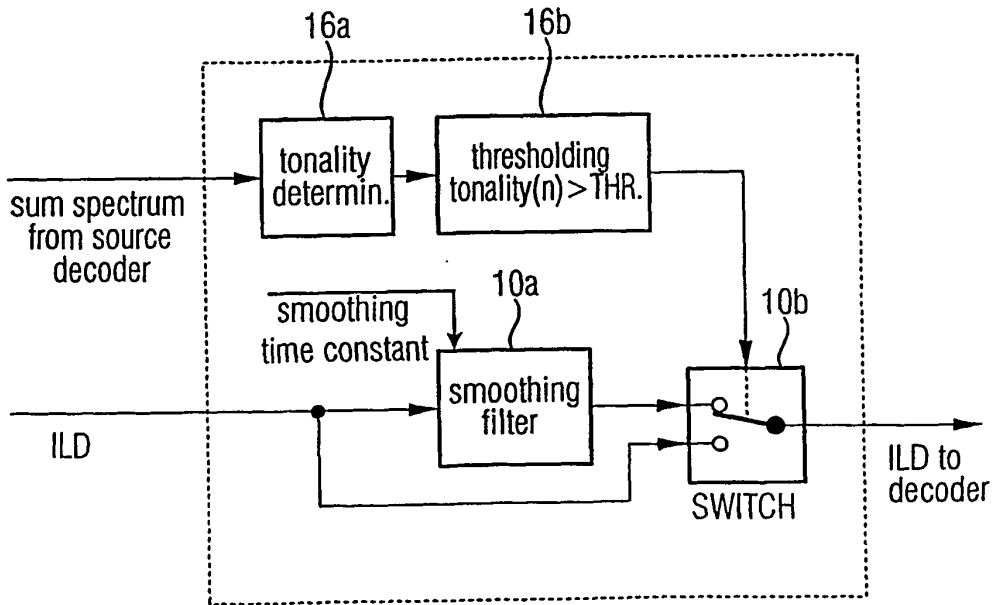


FIGURE 4B

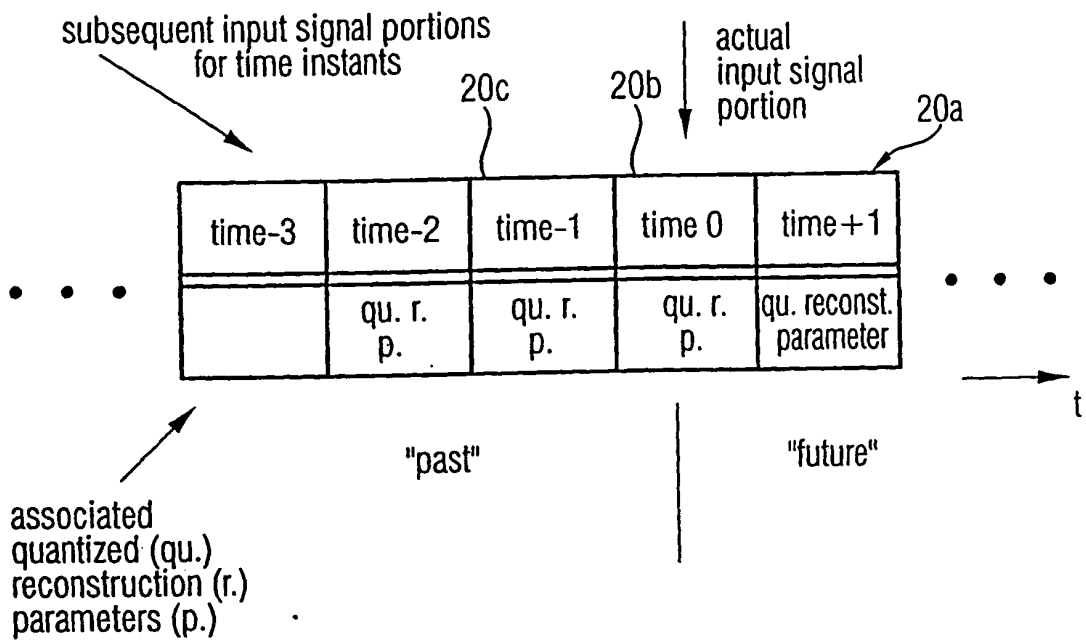


FIGURE 4C

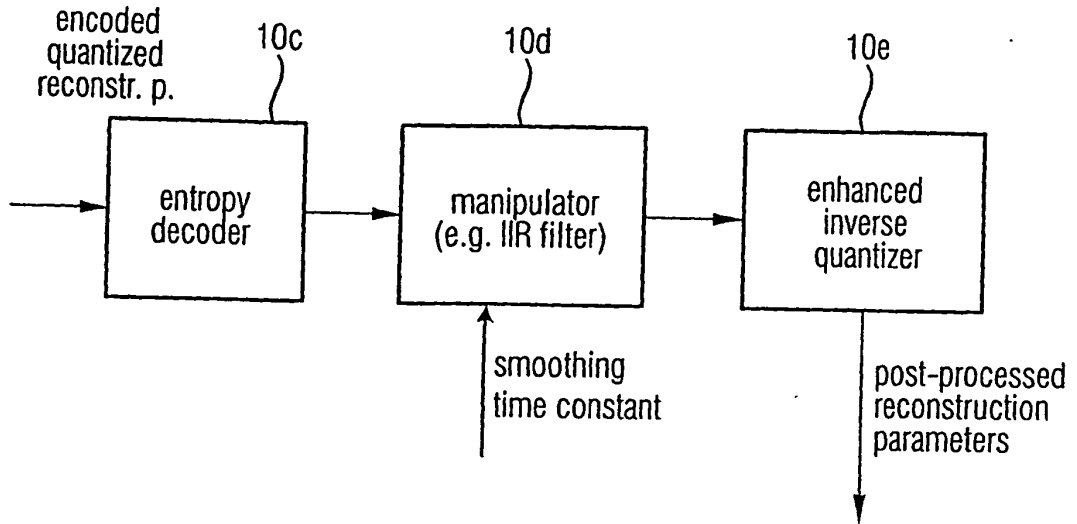


FIGURE 5

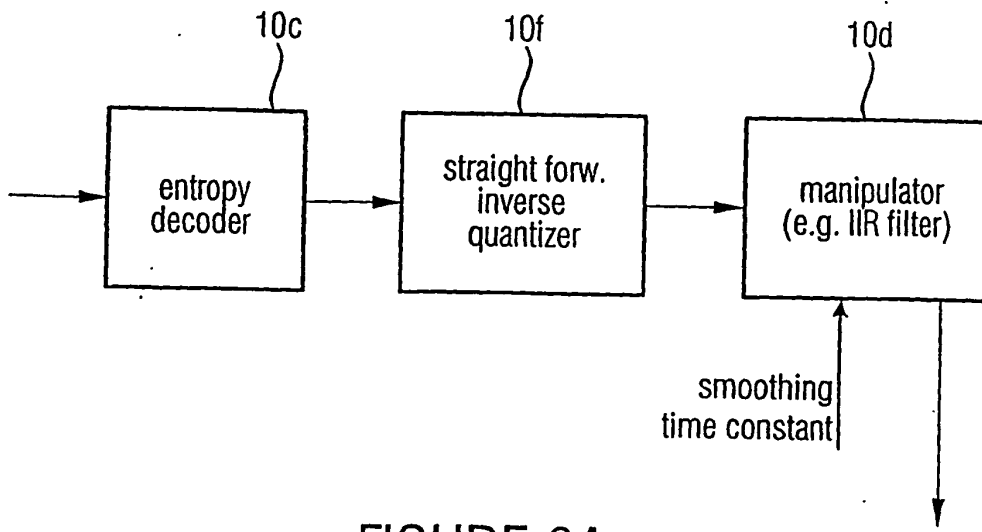


FIGURE 6A

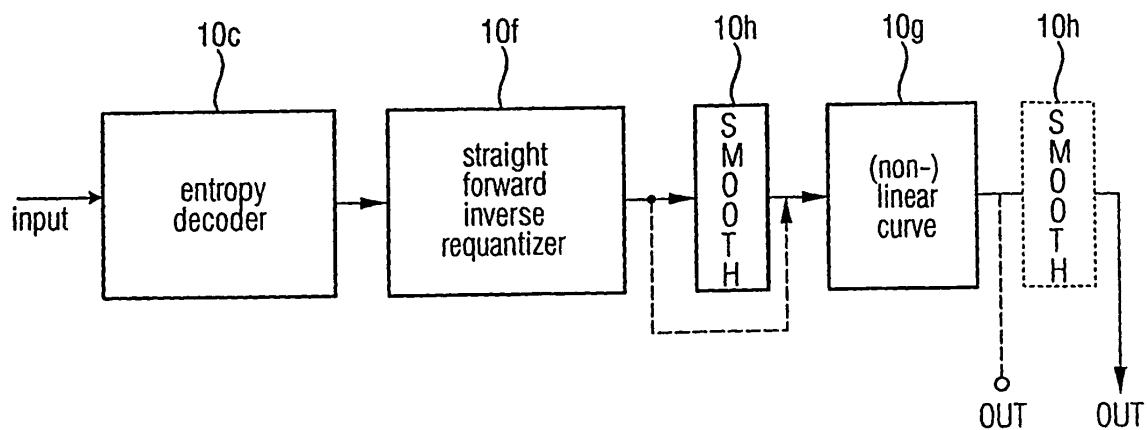


FIGURE 6B

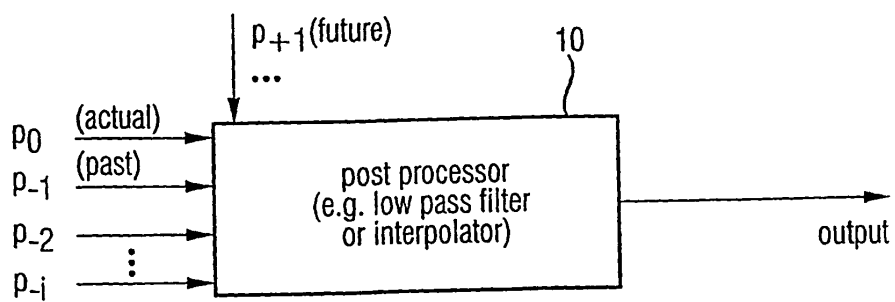


FIGURE 7A

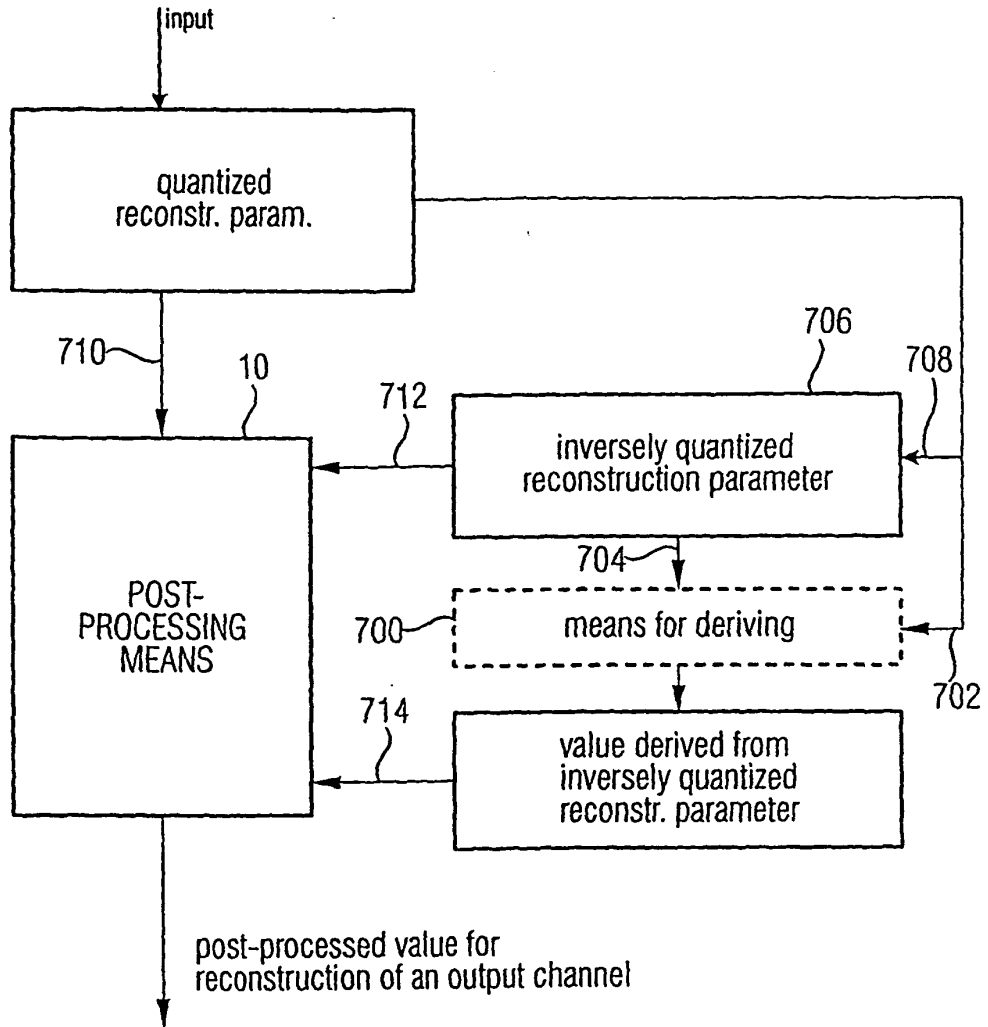
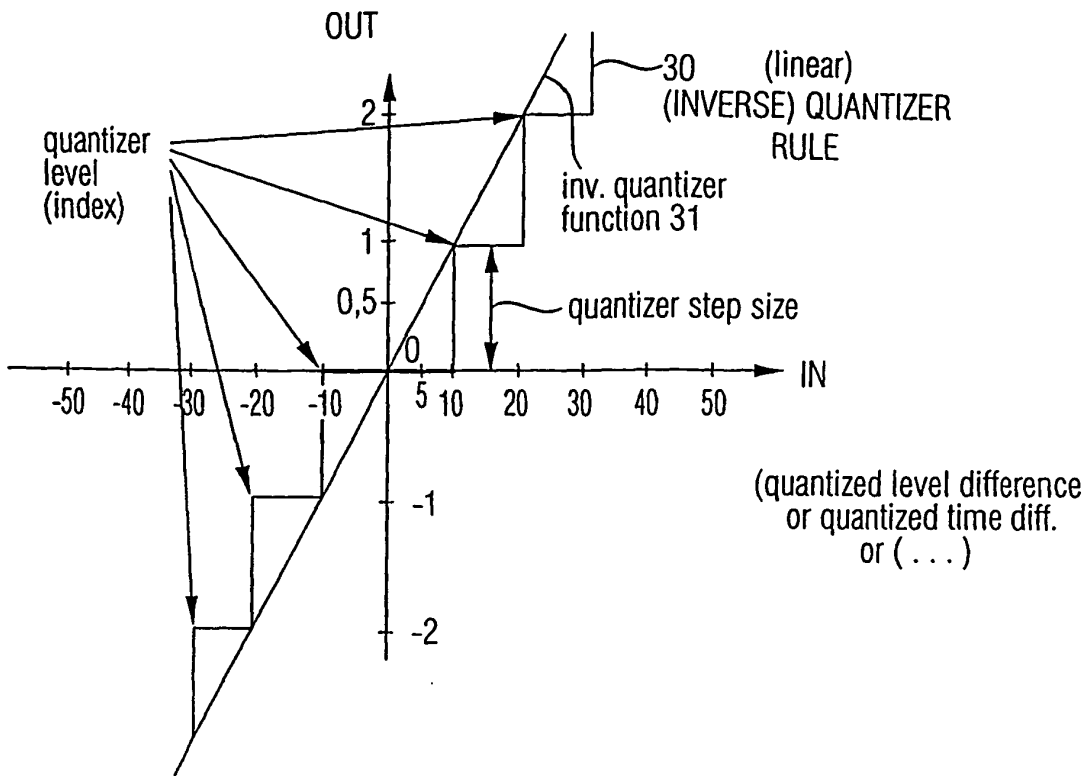


FIGURE 7B



- Straight-forward inverse quantizer maps integer quantizer levels to quantized values (e.g. 0, 10, 20, ...) only
- enhanced inverse quantizer maps non-integer (post-processed) quantizer "levels" to "inverse quantized values" between values determined by the (inverse) QUANTIZER RULE (e.g. 5, 15, 25, ...)

FIGURE 8



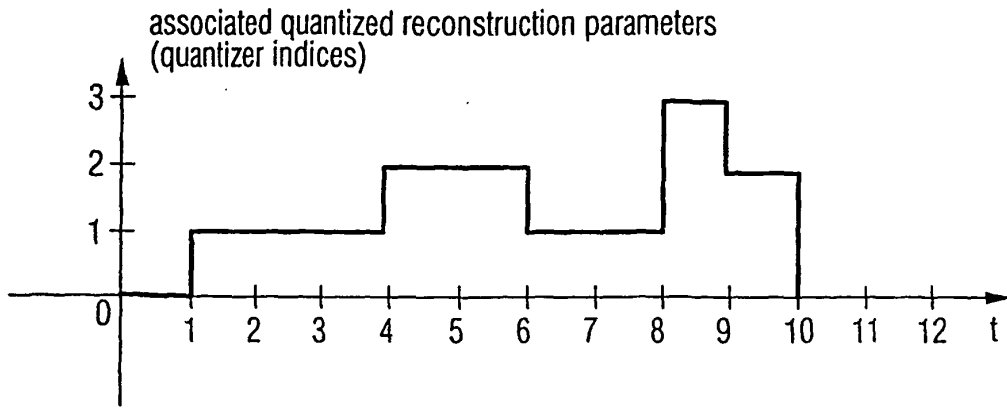


FIGURE 9A

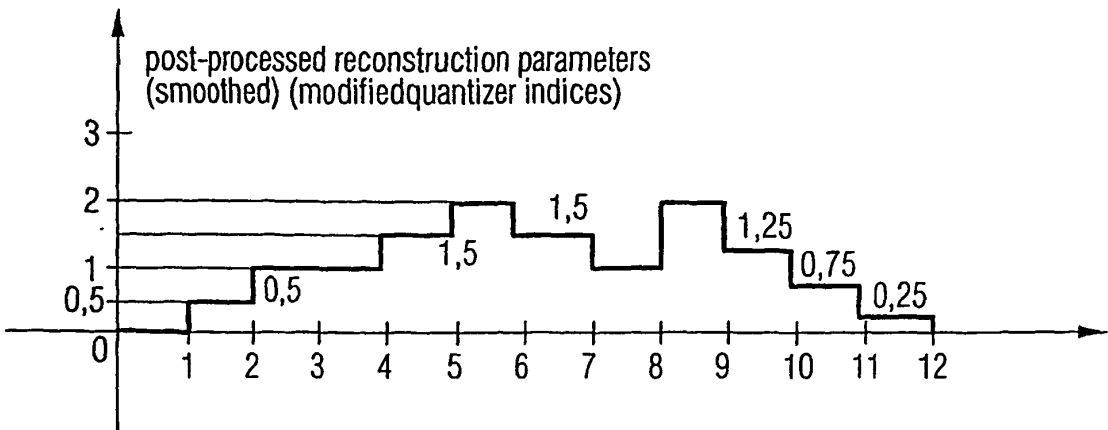


FIGURE 9B

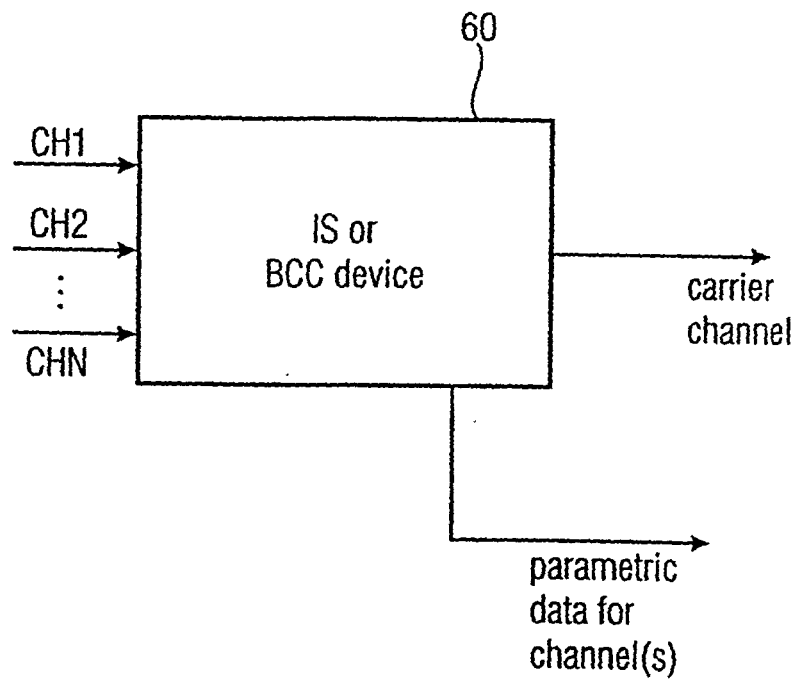


FIGURE 10  
(Prior Art)

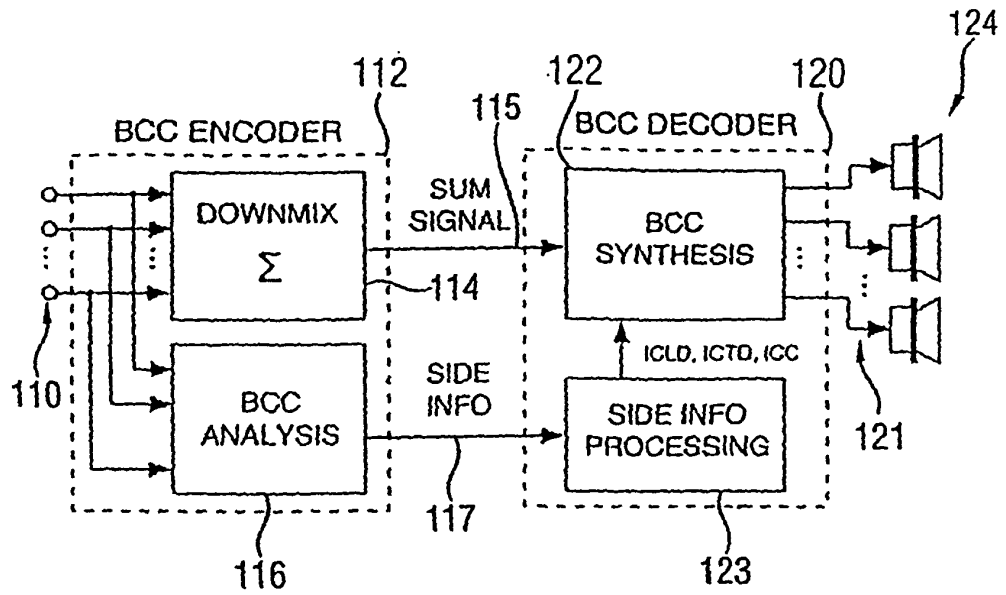


FIGURE 11  
(Prior Art)

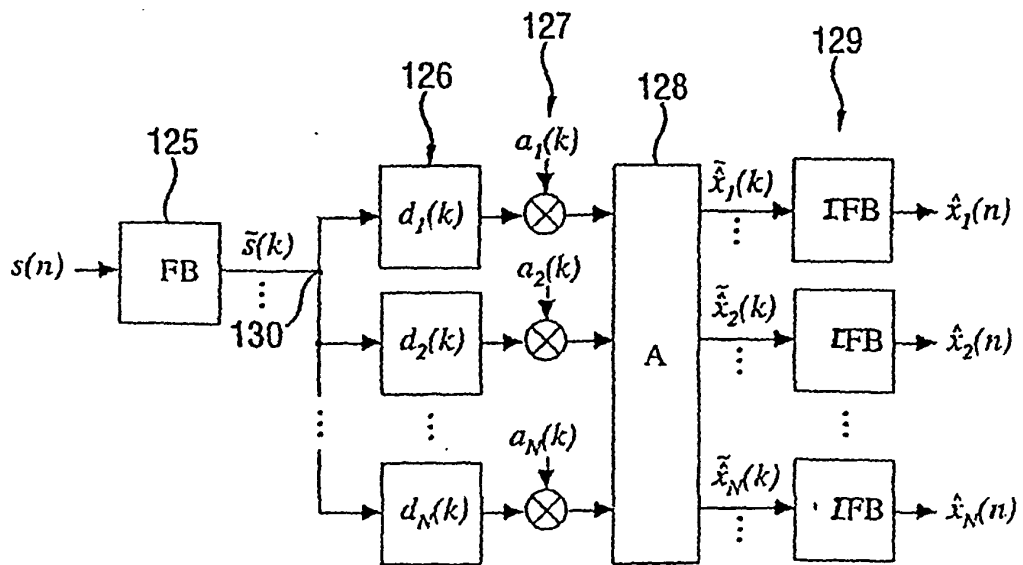


FIGURE 12  
(Prior Art)

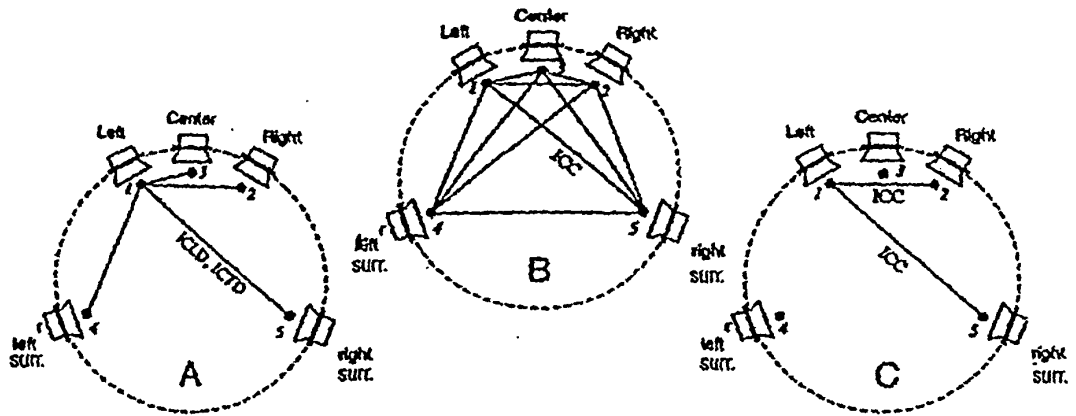


FIGURE 13  
(Prior Art)

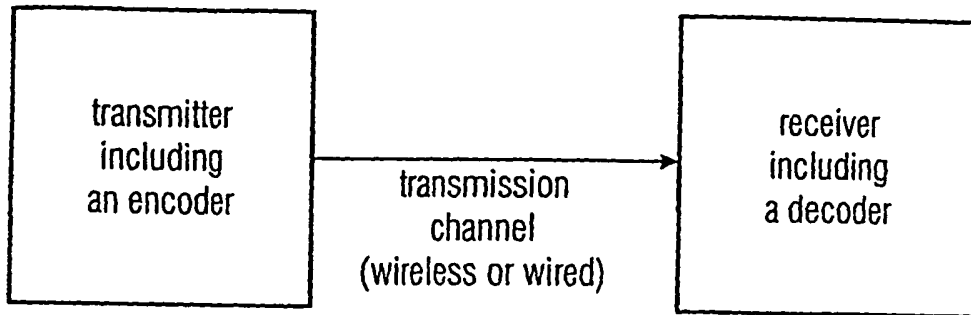


FIGURE 14



FIGURE 15

## REFERENCES CITED IN THE DESCRIPTION

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