quantized audio spectra or time-domain representation in the bitstream payload and to decode the quantized values and other reconstruction information. Hence, the encoder performs two decisions. The first decision is to perform a signal classification for frequency domain versus linear prediction domain mode decision. The second decision is 5 to determine, within the linear prediction domain (LPD), whether a signal portion is to be encoded using ACELP or TCX.

For applying a switched audio coding scheme in scenarios, where a very low delay is necessary, particular attention has to be paid to transform-based coding parts, since these 10 coding parts introduce a certain delay which depends on the transform length and window design. Therefore, the USAC coding concept is not suitable to very low-delay applications due to the modified AAC coding branch having a considerable transform length and length adaptation (also known as block switching) involving transitional windows.

On the other hand, the AMR-WB+ coding concept was found to be problematic due to the encoder-side decision whether ACELP or TCX is to be used. ACELP provides a good coding gain, but may result in significant audio quality problems when a signal portion is not suitable for the ACELP coding mode. Hence, for quality reasons, one might be inclined to use TCX whenever the input signal does not contain speech. However, using TCX too much at low bitrates will result in bitrate problems, since TCX provides a relatively low coding gain. When one, therefore, looks more onto the coding gain, one might use ACELP whenever possible, but, as stated before, this can result in audio quality problems due to the fact that ACELP is not optimal, for example, for music and similar stationary signals.

The segmental SNR calculation is a quality measure, which determines the better coding mode only based on the result, i.e., whether the SNR between the original signal or the encoded/decoded signal is better, so that the encoding algorithm resulting in a better SNR is used. This, however, always has to operate under bitrate constraints. Therefore, it has been found that only using a quality measure such as, for example, the segmental SNR measure does not always result in the best compromise between quality and bitrate.

~~It is the object~~ Example embodiments of the present ~~invention~~ application seek to provide an improved concept for coding a portion of an audio signal.

This ~~object is achieved~~ may be provided by an apparatus for coding a portion of an audio signal in accordance with claim 1 or a method for coding a portion of an audio signal in accordance with claim 14.

In accordance with a first aspect of the present invention, there is provided an apparatus for coding a portion of an audio signal to obtain an encoded audio signal for the portion of the audio signal, comprising: a transient detector for detecting whether a transient signal is located in the portion of the audio signal to obtain a transient detection result; an encoder stage for performing a first encoding algorithm on the audio signal, the first encoding algorithm having a first characteristic, and for performing a second encoding algorithm on the audio signal, the second encoding algorithm having a second characteristic being different from the first characteristic; a processor for determining which encoding algorithm results in an encoded audio signal being a better approximation of the portion of the audio signal with respect to the other encoding algorithm to obtain a quality result; and a controller for determining whether the encoded audio signal for the portion of the audio signal is to be generated by either the first encoding algorithm or the second encoding algorithm based on the transient detection result and the quality result.

In accordance with a second aspect of the present invention, there is provided a method of coding a portion of an audio signal to obtain an encoded audio signal for the portion of the audio signal, comprising: detecting whether a transient signal is located in the portion of the audio signal to obtain a transient detection result; performing a first encoding algorithm on the audio signal, the first encoding algorithm having a first characteristic, and performing a second encoding algorithm on the audio signal, the second encoding algorithm having a second characteristic being different from the first characteristic; determining which encoding algorithm results in an encoded audio signal being a better approximation to the portion of the audio signal with respect to the other encoding algorithm to obtain a quality result; and determining whether the encoded audio signal for the portion of the audio signal is to be generated by either the first encoding algorithm or the second encoding algorithm based on the transient detection result and the quality result.

The ~~present invention is~~ example embodiments are based on the finding that a better decision between a first encoding algorithm suited for more transient signal portions and a second encoding algorithm suitable for more stationary signal portions can be obtained when the decision is not only based on a quality measure but, additionally, on a transient detection result. While the quality measure only looks at the result of the encoding/decoding chain with respect to the original signal, the transient detection result additionally relies on an analysis of the original input audio signal alone. Hence, it has been found out that a combination of both measures, i.e., the quality result on the one hand and the transient detection result on the other hand for finally determining whether a portion of an audio signal is to be encoded by which encoding algorithm leads to an improved compromise between coding gain on the one hand and audio quality on the other hand.

An apparatus for coding a portion of an audio signal to obtain an encoded audio signal for the portion of an audio signal comprises a transient detector for detecting whether a transient signal is located in the portion of the audio signal to obtain a transient detection result. The apparatus furthermore comprises an encoder stage for performing a first encoding algorithm on the audio signal, the first encoding algorithm having a first characteristic, and for performing a second encoding algorithm on the audio signal, the second encoding algorithm having a second characteristic being different from the first characteristic. In an embodiment, the first characteristic associated with the first encoding algorithm is better suited for a more transient signal, and the second encoding characteristic associated with the second encoding algorithm is better suited for more stationary audio signals. Exemplarily, the first encoding algorithm is an ACELP encoding algorithm and the second encoding algorithm is a TCX encoding algorithm which may be based on a modified discrete cosine transform, an FFT transform or any other transform or filterbank. Furthermore, a processor is provided for determining, which encoding algorithm results in an encoded audio signal being a better approximation to the portion of the audio signal to obtain a quality result. Furthermore, a controller is provided, where the controller is configured for determining whether the encoded audio signal for the portion of the audio signal is generated by either the first encoding algorithm or the second encoding algorithm. In accordance with the ~~invention~~ example embodiments of the present application, the controller is configured for performing this determination not only based on the quality result but, additionally, on the transient detection result.

In an embodiment, the controller is configured for determining the second encoding algorithm, although the quality result indicates a better quality for the first encoding algorithm, when the transient detection result indicates a non-transient signal. Furthermore, the controller is configured for determining the first encoding algorithm, although the

quality result indicates a better quality for the second encoding algorithm, when the transient detection result indicates a transient signal.

In a further embodiment, this determination, in which the transient result can negate the quality result, is enhanced using a hysteresis function such that the second encoding algorithm is only determined when a number of earlier signal portions, for which the first encoding algorithm has been determined, is smaller than a predetermined number. Analogously, the controller is configured to only determine the first encoding algorithm when a number of earlier signal portions, for which the second encoding algorithm has been determined in the past, is smaller than a predetermined number. An advantage from the hysteresis processing is that the number of switch-overs between coding modes is reduced for certain input signals. A too frequent switch-over at critical points in the signal may generate audible artifacts specifically for low bitrates. The probability of such artifacts is reduced by implementing the hysteresis.

In a further embodiment, the quality result is favored with respect to the transient detection result when the quality result indicates a strong quality advantage for one coding algorithm. Then, the encoding algorithm having the much better quality result than the other encoding algorithm is selected irrespective of whether the signal is a transient signal or not. On the other hand, the transient detection result can become decisive when the quality difference between both encoding algorithms is not so high. To this end, it is preferred to not only determine a binary quality result, but a quantitative quality result. A binary quality result would only indicate which encoding algorithm results in a better quality, whereas a quantitative quality result not only determines which encoding algorithm results in a better quality, but how much better the corresponding encoding algorithm is. On the other hand, one could also use a quantitative transient detection result but, basically, a binary transient detection result would be sufficient as well.

Hence, the example embodiments of the present ~~invention~~ application provide~~s~~ a particular advantage with respect to a good compromise between bitrate on the one hand and quality on the other hand, since, for transient signals, the coding algorithm resulting in less quality is selected. When the quality result favors e.g. a TCX decision, nevertheless the ACELP mode is taken, which might result in a slightly reduced audio quality but, in the end, results in a higher coding gain associated with using the ACELP mode.

When, on the other hand, the quality result favors an ACELP frame, a TCX decision is, nevertheless, taken for non-transient signals. Hence, the slightly less coding gain is accepted in favor of a better audio quality.

Thus, the example embodiments of the present ~~invention~~application result~~s~~ in an improved compromise between quality and bitrate due to the fact that not only the quality of the encoded and again decoded signal is considered but, in addition, also the actually to be encoded input signal is analyzed with 5 respect to its transient characteristic and the result of this transient analysis is used to additionally influence the decision for an algorithm better suited for transient signals or an algorithm better suited for stationary signals.

Further embodiments of the present invention are subsequently illustrated by reference to the accompanying drawings, in which:

Fig. 1 illustrates a block diagram of an apparatus for coding a portion of an audio signal in accordance with an embodiment;

Fig. 2 illustrates a table for two different encoding algorithms and the signals for which they are suited;

Fig. 3 illustrates an overview over the quality condition, the transient condition and the hysteresis condition, which can be applied independently of each other, but which are, preferably, applied jointly;

Fig. 4 illustrates a state table indicating whether a switch-over is performed or not for different situations;

Fig. 5 illustrates a flowchart for determining the transient result in an embodiment;

Fig. 6a illustrates a flowchart for determining the quality result in an embodiment;

Fig. 6b illustrates more details on the quality result of Fig. 6a; and

Fig. 7 illustrates a more detailed block diagram of an apparatus for coding in accordance with an embodiment.

Fig. 1 illustrates an apparatus for coding a portion of an audio signal provided at an input 35 line 10. The portion of the audio signal is input into a transient detector 12 for detecting whether a transient signal is located in the portion of the audio signal to obtain a transient detection result on line 14. Furthermore, an encoder stage 16 is provided where the encoder stage is configured for performing a first encoding algorithm on the audio signal, the first