Claims

1. Apparatus ~~(10)~~ for selecting one of a first encoding algorithm having a first characteristic and a second encoding algorithm having a second characteristic for encoding a portion of an audio signal ~~(40)~~ to obtain an encoded version of the portion of the audio signal ~~(40)~~, comprising:

a long-term prediction filter configured to receive the audio signal, to reduce the amplitude of harmonics in the audio signal and to output a filtered version of the audio signal;

a first estimator ~~(12)~~ for using the filtered version of the audio signal in estimating a SNR (signal to noise ratio) or a segmental SNR of the portion of the audio signal as a first quality measure for the portion of the audio signal, the first quality measure being associated with the first encoding algorithm, wherein estimating said first quality measure comprises performing an approximation of the first encoding algorithm to obtain a distortion estimate of the first encoding algorithm and to estimate the first quality measure based on the portion of the audio signal and the distortion estimate of the first encoding algorithm without actually encoding and decoding the portion of the audio signal using the first encoding algorithm;

a second estimator ~~(14)~~ for estimating a SNR or a segmental SNR as a second quality measure for the portion of the audio signal, the second quality measure being associated with the second encoding algorithm, wherein estimating said second quality measure comprises performing an approximation of the second encoding algorithm to obtain a distortion estimate of the second encoding algorithm and to estimate the second quality measure using the portion of the audio signal and the distortion estimate of the second encoding algorithm without actually encoding and decoding the portion of the audio signal using the second encoding algorithm; and

a controller ~~(16)~~ for selecting the first encoding algorithm or the second encoding algorithm based on a comparison between the first quality measure and the second quality measure,

wherein the first encoding algorithm is a transform coding algorithm, a MDCT (modified discrete cosine transform) based coding algorithm or a TCX (transform coding excitation) coding algorithm and wherein the second encoding algorithm is a CELP (code excited linear prediction) coding algorithm or an ACELP (algebraic code excited linear prediction) coding algorithm.

2. Apparatus ~~(10)~~ of claim 1, wherein a transfer function of the long-term prediction filter comprises an integer part of a pitch lag and a multi tap filter depending on a fractional part of the pitch lag.

3. Apparatus ~~(10)~~ of claim 1, wherein the long-term prediction filter has the transfer function:

$$P(z) = 1 - \beta g B(z, T_{fr}) z^{-T_{int}}$$

with T_{int} and T_{fr} are the integer and fractional part of a pitch-lag, g is a gain, β is a weight and $B(z,T_{fr})$ is a FIR low-pass filter whose coefficients depend on the fractional part of the pitch.

4. Apparatus of one of claims 1 to 3, further comprising a disabling unit for disabling the filter based on a combination of one or more harmonicity measures and/or one or more temporal structure measures.

5. Apparatus of claim 4, wherein the one or more harmonicity measures comprise at least one of a normalized correlation or a prediction gain and wherein the one or more temporal structure measures comprise at least one of a temporal flatness measure and an energy change.

6. Apparatus of one of claims 1 to 5, wherein the filter is applied to the audio signal on a frame-by-frame basis, said apparatus further comprising a unit for removing discontinuities in the audio signal caused by the filter.

7. Apparatus ~~(10)~~ of one of claims 1 to 6, wherein the first and second estimators are configured to estimate a SNR or segmental SNR of a portion of a weighted version of the audio signal.

8. Apparatus ~~(10)~~ of one of claims 1 to 7, wherein the first estimator ~~(12)~~ is configured to determine an estimated quantizer distortion which a quantizer used in the first encoding algorithm would introduce when quantizing the portion of the audio signal and to estimate the first quality measure based on an energy of a portion of a weighted version of the audio signal and the estimated quantizer distortion, wherein the first estimator ~~(12)~~ is configured to estimate a global gain for the portion of the audio signal such that the portion of the audio signal would produce a given target bitrate when encoded with a quantizer and an entropy coder used in the first encoding algorithm, wherein the first estimator ~~(12)~~ is further configured to determine the estimated quantizer distortion based on the estimated global gain.

9. Apparatus ~~(10)~~ of one of claims 1 to 8, wherein the second estimator ~~(14)~~ is configured to determine an estimated adaptive codebook distortion which an adaptive codebook used in the second encoding algorithm would introduce when using the adaptive codebook to encode the portion of the audio signal, and wherein the second estimator ~~(14)~~ is configured to estimate the second quality measure based on an energy of a portion of a weighted version of the audio signal and the estimated adaptive codebook distortion, wherein, for each of a plurality of sub-portions of the portion of the audio signal, the second estimator ~~(14)~~ is configured to approximate the adaptive codebook based on a version of the sub-portion of the weighted audio signal shifted to the past by a pitch-lag determined in a pre-processing stage, to estimate an adaptive codebook gain such that an error between the sub-portion of the portion of the weighted audio signal and the approximated adaptive codebook is minimized, and to determine the estimated adaptive codebook distortion based on the energy of an error between the sub-portion of the portion of the weighted audio signal and the approximated adaptive codebook scaled by the adaptive codebook gain.

10. Apparatus ~~(10)~~ of claim 9, wherein the second estimator ~~(14)~~ is further configured to reduce the estimated adaptive codebook distortion determined for each sub-portion of the portion of the audio signal by a constant factor.

11. Apparatus ~~(10)~~ of one of claims 1 to 8, wherein the second estimator ~~(14)~~ is configured to determine an estimated adaptive codebook distortion which an adaptive codebook used in the second encoding algorithm would introduce when using the adaptive codebook to encode the portion of the audio signal, and

wherein the second estimator ~~(14)~~ is configured to estimate the second quality measure based on an energy of a portion of a weighted version of the audio signal and the estimated adaptive codebook distortion, wherein the second estimator ~~(14)~~ is configured to approximate the adaptive codebook based on a version of the portion of the weighted audio signal shifted to the past by a pitch-lag determined in a pre-processing stage, to estimate an adaptive codebook gain such that an error between the portion of the weighted audio signal and the approximated adaptive codebook is minimized, and to determine the estimated adaptive codebook distortion based on the energy of an error between the portion of the weighted audio signal and the approximated adaptive codebook scaled by the adaptive codebook gain.

12. Apparatus ~~(20)~~ for encoding a portion of an audio signal, comprising the apparatus ~~(10)~~ according to one of claims 1 to 11, a first encoder stage ~~(26)~~ for performing the first encoding algorithm and a second encoder stage ~~(28)~~ for performing the second encoding algorithm, wherein the apparatus for encoding ~~(20)~~ is configured to encode the portion of the audio signal using the first encoding algorithm or the second encoding algorithm depending on the selection by the controller ~~(16)~~.

13. System for encoding and decoding comprising an apparatus ~~(20)~~ for encoding according to claim 12 and a decoder configured to receive the encoded version of the portion of the audio signal and an indication of the algorithm used to encode the portion of the audio signal and to decode the encoded version of the portion of the audio signal using the indicated algorithm.

14. Method for selecting one of a first encoding algorithm having a first characteristic and a second encoding algorithm having a second characteristic for encoding a portion of an audio signal to obtain an encoded version of the portion of the audio signal, comprising:

filtering the audio signal using a long-term prediction filter to reduce the amplitude of harmonics in the audio signal and to output a filtered version of the audio signal;

using the filtered version of the audio signal in estimating a SNR or a segmented SNR of the portion of the audio signal as a first quality measure for the portion of the audio signal, the first quality measure being associated with the first encoding

algorithm, wherein estimating said first quality measure comprises performing an approximation of the first encoding algorithm to obtain a distortion estimate of the first encoding algorithm and to estimate the first quality measure based on the portion of the first audio signal and the distortion estimate of the first encoding algorithm without actually encoding and decoding the portion of the audio signal using the first encoding algorithm;

estimating a SNR or a segmented SNR as a second quality measure for the portion of the audio signal, the second quality measure being associated with the second encoding algorithm, wherein estimating said second quality measure comprises performing an approximation of the second encoding algorithm to obtain a distortion estimate of the second encoding algorithm and to estimate the second quality measure using the portion of the audio signal and the distortion estimate of the second encoding algorithm without actually encoding and decoding the portion of the audio signal using the second coding algorithm; and

selecting the first encoding algorithm or the second encoding algorithm based on a comparison between the first quality measure and the second quality measure,

wherein the first encoding algorithm is a transform coding algorithm, a MDCT (modified discrete cosine transform) based coding algorithm or a TCX (transform coding excitation) coding algorithm and wherein the second encoding algorithm is a CELP (code excited linear prediction) coding algorithm or an ACELP (algebraic code excited linear prediction) coding algorithm.

15. Computer program having a program code for performing, when running on a computer, the method of one of claims 14.