Claims

1. Audio encoder for encoding an audio signal, comprising:

a first encoding processor ~~(600)~~ for encoding a first audio signal portion in a frequency domain, wherein the first encoding processor ~~(600)~~ comprises:

a time frequency converter ~~(602)~~ for converting the first audio signal portion into a frequency domain representation having spectral lines up to a maximum frequency of the first audio signal portion;

a spectral encoder ~~(606)~~ for encoding the frequency domain representation;

a second encoding processor ~~(610)~~ for encoding a second different audio signal portion in the time domain,

wherein the second encoding processor ~~(610)~~ has an associated second sampling rate,

wherein the first encoding processor ~~(600)~~ has associated therewith a first sampling rate being different from the second sampling rate;

a cross-processor ~~(700)~~ for calculating, from the encoded spectral representation of the first audio signal portion, initialization data of the second encoding processor ~~(610)~~, so that the second encoding processing ~~(610)~~ is initialized to encode the second audio signal portion immediately following the first audio signal portion in time in the audio signal; wherein the cross-processor comprises a frequency-time converter ~~(702)~~ for generating a time domain signal at the second sampling rate, wherein the frequency time converter ~~(702)~~ comprises:

a selector ~~(726)~~ for selecting a portion of a spectrum input into the frequency time converter in accordance with a ratio of the first sampling rate and the second sampling rate,

a transform processor ~~(720)~~ having a transform length being different from a transform length of the time-frequency converter ~~(602)~~; and

a synthesis windower ~~(712)~~ for windowing using a window having a different number of window coefficients compared to a window used by the time frequency converter ~~(602)~~;

a controller ~~(620)~~ configured for analyzing the audio signal and for determining, which portion of the audio signal is the first audio signal portion encoded in the frequency domain and which portion of the audio signal is the second audio signal portion encoded in the time domain; and

an encoded signal former ~~(630)~~ for forming an encoded audio signal comprising a first encoded signal portion for the first audio signal portion and a second encoded signal portion for the second audio signal portion.

2. Audio encoder of claim 1, wherein the input signal has a high band and a low band,

wherein the second encoding processor ~~(610)~~ comprises a sampling rate converter ~~(900)~~ for converting the second audio signal portion to a lower sampling rate representation, the lower sampling rate being lower than a sampling rate of the audio signal, wherein the lower sampling rate representation does not include the high band of the input signal;

a time domain low band encoder ~~(910)~~ for time domain encoding the lower sampling rate representation; and

a time domain bandwidth extension encoder ~~(920)~~ for parametrically encoding the high band.

3. Audio encoder of claim 1 or 2, further comprising:

a preprocessor ~~(1000)~~ configured for preprocessing the first audio signal portion and the second audio signal portion,

wherein the preprocessor comprises a prediction analyzer ~~(1002)~~ for determining prediction coefficients;

wherein the encoded signal former ~~(630)~~ is configured for introducing an encoded version of the prediction coefficients into the encoded audio signal.

4. Audio encoder of claims 1, 2 or 3,

wherein a preprocessor ~~(1000)~~ comprises a resampler ~~(1004)~~ for resampling the audio signal to a sampling rate of the second encoding processor; and

wherein a prediction analyzer is configured to determine the prediction coefficients using a resampled audio signal, or

wherein the preprocessor ~~(1000)~~ further comprises a long term prediction analysis stage ~~(1006)~~ for determining one or more long term prediction parameters for the first audio signal portion.

5. Audio encoder of one of the preceding claims, wherein the cross-processor ~~(700)~~ comprises:

a spectral decoder ~~(701)~~ for calculating a decoded version of the first encoded signal portion;

a delay stage ~~(707)~~ for feeding a delayed version of the decoded version into a de-emphasis stage ~~(617)~~ of the second encoding processor for initialization;

a weighted prediction coefficient analysis filtering block ~~(708)~~ for feeding a filter output into a codebook determinator ~~(613)~~ of the second encoding processor ~~(610)~~ for initialization;

an analysis filtering stage ~~(706)~~ for filtering the decoded version or a pre-emphasized ~~(709)~~ version and for feeding a filter residual into an adaptive codebook determinator ~~(612)~~ of the second encoding processor for initialization; or

a pre-emphasis filter ~~(709)~~ for filtering the decoded version and for feeding a delayed or pre-emphasized version to a synthesis filtering stage ~~(616)~~ of the second encoding processor ~~(610)~~ for initialization.

6. Audio encoder of one of the preceding claims,

wherein the first encoding processor (600) is configured to perform a shaping (606a) of spectral values of the frequency domain representation using prediction coefficients (1002, 1010) derived from the first audio signal portion, and wherein the first encoding processor (600) is furthermore configured to perform a quantization and entropy coding operation (606b) of shaped spectral values of the first spectral regions.

7. Audio encoder of any one of the preceding claims, wherein the cross-processor (700) comprises:

a noise shaper (703) for shaping quantized spectral values of the frequency domain representation using LPC coefficients (1010) derived from the first audio signal portion;

a spectral decoder (704, 705) for decoding spectrally shaped spectral portions of the frequency domain representation with a high spectral resolution to obtain a decoded spectral representation;

a frequency-time converter (702) for converting the spectral representation into a time domain to obtain a decoded first audio signal portion, wherein a sampling rate associated with the decoded first audio signal portion is different from a sampling rate of the audio signal, and a sampling rate associated with an output signal of the frequency-time converter (702) is different from a sampling rate associated with the audio signal input into the frequency-time converter (602).

8. Audio encoder of one of the preceding claims,

wherein the second encoding processor comprises at least one block of the following group of blocks:

a prediction analysis filter (611);

an adaptive codebook stage (612);

an innovative codebook stage ~~(614)~~;

an estimator ~~(613)~~ for estimating an innovative codebook entry;

an ACELP/gain coding stage ~~(615)~~;

a prediction synthesis filtering stage ~~(616)~~;

a de-emphasis stage ~~(617)~~; and

a bass post-filter analysis stage ~~(618)~~.

9. Audio decoder for decoding an encoded audio signal, comprising:

a first decoding processor ~~(1120)~~ for decoding a first encoded audio signal portion in a frequency domain, the first decoding processor ~~(1120)~~ comprising a frequency-time converter ~~(1120)~~ for converting a decoded spectral representation into a time domain to obtain a decoded first audio signal portion;

a second decoding processor ~~(1140)~~ for decoding a second encoded audio signal portion in the time domain to obtain a decoded second audio signal portion;

a cross-processor ~~(1170)~~ for calculating, from the decoded spectral representation of the first encoded audio signal portion, initialization data of the second decoding processor ~~(1140)~~, so that the second decoding processor ~~(1140)~~ is initialized to decode the encoded second audio signal portion following in time the first audio signal portion in the encoded audio signal; and

a combiner ~~(1160)~~ for combining the decoded first spectral portion and the decoded second spectral portion to obtain a decoded audio signal,

wherein the cross-processor further comprises

a further frequency-time converter ~~(1171)~~ operating at a first effective sampling rate being different from a second effective sampling rate associated with the frequency-time converter ~~(1124)~~ of the first decoding

processor ~~(1120)~~ to obtain a further decoded first signal portion in the time domain,

wherein the signal output by the further frequency-time converter ~~(1171)~~ has the second sampling rate being different from the first sampling rate associated with an output of the frequency-time converter ~~(1124)~~ of the first decoding processor,

wherein the further frequency-time converter ~~(1171)~~ comprises a selector ~~(726)~~ for selecting a portion of a spectrum input into the further frequency-time converter ~~(1171)~~ in accordance with a ratio of the first sampling rate and the second sampling rate;

a transform processor ~~(720)~~ having a transform length being different from a transform length ~~(710)~~ of the time-frequency converter ~~(1124)~~ of the first decoding processor ~~(1120)~~; and

a synthesis windower ~~(722)~~ using a window having a different number of coefficients compared to a window used by the frequency-time converter ~~(1124)~~ of the first decoding processor ~~(1120)~~.

10. Audio decoder of claim 9, wherein the second decoding processor comprises:

a time domain low band decoder ~~(1200)~~ for decoding a low band time domain signal;

a resampler ~~(1210)~~ for resampling the low band time domain signal;

a time domain bandwidth extension decoder ~~(1220)~~ for synthesizing a high band of a time domain output signal; and

a mixer ~~(1230)~~ for mixing a synthesized high band of the time domain signal and a resampled low band time domain signal.

11. Audio decoder of one of the claims 9 to 10,

wherein the first decoding processor ~~(1120)~~ comprises an adaptive long term prediction post-filter ~~(1420)~~ for post-filtering the first decoded first signal portion, wherein the filter ~~(1420)~~ is controlled by one or more long term prediction parameters included in the encoded audio signal.

12. Audio decoder of one of claims 9 to 11, wherein the cross-processor ~~(1170)~~ comprises:

a delay stage ~~(1172)~~ for delaying the further decoded first signal portion and for feeding a delayed version of the decoded first signal portion into a de-emphasis stage ~~(1144)~~ of the second decoding processor for initialization;

a pre-emphasis filter ~~(1173)~~ and a delay stage ~~(1175)~~ for filtering and delaying the further decoded first signal portion and for feeding a delay stage output into a prediction synthesis filter ~~(1143)~~ of the second decoding processor for initialization;

a prediction analysis filter ~~(1174)~~ for generating a prediction residual signal from the further decoded first spectral portion or a pre-emphasized ~~(1173)~~ further decoded first signal portion and for feeding a prediction residual signal into a codebook synthesizer ~~(1141)~~ of the second decoding processor ~~(1200)~~; or

a switch ~~(1480)~~ for feeding the further decoded first signal portion into an analysis stage ~~(1471)~~ of a resampler ~~(1210)~~ of the second decoding processor for initialization.

13. Audio decoder of one of claims 9 to 12,

wherein the second decoding processor ~~(1200)~~ comprises at least one block of the group of blocks comprising:

a stage for decoding ACELP gains and an innovative codebook;

an adaptive codebook synthesis stage ~~(1141)~~;

an ACELP post-processor ~~(1142)~~

;

a prediction synthesis filter (1143); and

a de-emphasis stage (1144).

14. Method of encoding an audio signal, comprising:

encoding (600) a first audio signal portion in a frequency domain, comprising:

converting (602) the first audio signal portion into a frequency domain representation having spectral lines up to a maximum frequency of the first audio signal portion;

encoding (606) the frequency domain representation;

encoding (610) a second different audio signal portion in the time domain;

wherein the encoding (610) the second audio signal portion has an associated second sampling rate,

wherein the encoding (600) the first audio signal portion has associated therewith a first sampling rate being different from the second sampling rate

calculating (700), from the encoded spectral representation of the first audio signal portion, initialization data for the step of encoding the second different audio signal portion, so that the step of encoding (610) the second different audio signal portion is initialized to encode the second audio signal portion immediately following the first audio signal portion in time in the audio signal wherein the calculating (700) comprises generating (702), by a frequency-time converter, a time domain signal at the second sampling rate, wherein the generating (702) comprises:

selecting (726) a portion of a spectrum input into the frequency-time converter in accordance with a ratio of the first sampling rate and the second sampling rate,

processing using a transform processor (720) having a transform length being different from a transform length of a time-frequency converter used in the converting (602) the first audio signal portion; and

synthesis windowing (712) using a window having a different number of window coefficients compared to a window used by the time frequency converter (602) used in the converting (602) the first audio signal portion;

analyzing (620) the audio signal and determining, which portion of the audio signal is the first audio signal portion encoded in the frequency domain and which portion of the audio signal is the second audio signal portion encoded in the time domain; and

forming (630) an encoded audio signal comprising a first encoded signal portion for the first audio signal portion and a second encoded signal portion for the second audio signal portion.

15. Method of decoding an encoded audio signal, comprising:

decoding (1120), by a first decoding processor, a first encoded audio signal portion in a frequency domain, the decoding (1120) comprising: converting (1120), by a frequency-time converter (1124), a decoded spectral representation into a time domain to obtain a decoded first audio signal portion;

decoding (1140) a second encoded audio signal portion in the time domain to obtain a decoded second audio signal portion;

calculating (1170), from the decoded spectral representation of the first encoded audio signal portion, initialization data of the step of decoding (1140) the second encoded audio signal portion, so that the step of decoding the second encoded audio signal portion is initialized to decode the encoded second audio signal portion following in time the first audio signal portion in the encoded audio signal; and

combining (1160) the decoded first spectral portion and the decoded second spectral portion to obtain a decoded audio signal,

wherein the calculating ~~(1170)~~ further comprises

using a further frequency-time converter ~~(1171)~~ operating at a first effective sampling rate being different from a second effective sampling rate associated with the frequency-time converter ~~(1124)~~ of the first decoding processor ~~(1120)~~ to obtain a further decoded first signal portion in the time domain,

wherein the signal output by the further frequency-time converter ~~(1171)~~ has the second sampling rate being different from the first sampling rate associated with an output of the frequency-time converter ~~(1124)~~ of the first decoding processor,

wherein the using the further frequency-time converter ~~(1171)~~ comprises:

selecting ~~(726)~~ a portion of a spectrum input into the further frequency-time converter ~~(1171)~~ in accordance with a ratio of the first sampling rate and the second sampling rate;

using a transform processor ~~(720)~~ having a transform length being different from a transform length ~~(710)~~ of the time-frequency converter ~~(1124)~~ of the first decoding processor ~~(1120)~~; and

using a synthesis windower ~~(722)~~ using a window having a different number of coefficients compared to a window used by the frequency-time converter ~~(1124)~~ of the first decoding processor ~~(1120)~~.

16. Computer program for performing, when running on a computer or a processor, the method of claim 14 or claim 15.