Claims

1. An apparatus for decoding an encoded audio signal to obtain a reconstructed audio signal, wherein the apparatus is configured to receive a plurality of frames, and wherein the apparatus comprises:

an inverse modified discrete cosine transform module (683) for decoding the plurality of frames by conducting an inverse modified discrete cosine transform to obtain audio signal samples of the decoded audio signal, and

a long-term prediction unit (684) for conducting long-term prediction, comprising:

a delay buffer (1020) for storing the audio signal samples of the decoded audio signal,

a sample selector (1030) for selecting a plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer (1020), and

a sample processor (1040) for processing the selected audio signal samples to obtain reconstructed audio signal samples of the reconstructed audio signal,

wherein the sample selector (1030) is configured to select, if a current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer (1020) depending on a pitch lag information being comprised by the current frame, and

wherein the sample selector (1030) is configured to select, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted, the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer (1020) depending on a pitch lag information being comprised by another frame being received previously by the apparatus,

wherein the sample selector (1030) is configured to obtain the reconstructed audio signal samples by rescaling the selected audio signal samples depending on a modified gain, wherein the modified gain is defined according to the formula:

```
gain = gain_past * damping;
```

wherein `gain` is the modified gain,

wherein the sample selector (1030) is configured to set `gain_past` to `gain` after `gain` has been calculated, and

wherein `damping` is a real value with $0 \leq damping \leq 1$.

2. An apparatus according to claim 1,

wherein the sample processor (1040) is configured to obtain the reconstructed audio signal samples, if the current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, by rescaling the selected audio signal samples depending on a gain information being comprised by the current frame, and

wherein the sample selector (1030) is configured to obtain the reconstructed audio signal samples, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted, by rescaling the selected audio signal samples depending on the gain information being comprised by said another frame being received previously by the apparatus.

3. An apparatus according to claim 2,

wherein the sample processor (1040) is configured to obtain the reconstructed audio signal samples, if the current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, by multiplying the selected audio signal samples and a value depending on the gain information being comprised by the current frame, and

wherein the sample selector (1030) is configured to obtain the reconstructed audio signal samples, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted, by multiplying the

selected audio signal samples and a value depending on the gain information being comprised by said another frame being received previously by the apparatus.

4. An apparatus according to one of the preceding claims, wherein the sample processor (1040) is configured to store the reconstructed audio signal samples into the delay buffer (1020).

5. An apparatus according to claim 4, wherein the sample processor (1040) is configured to store the reconstructed audio signal samples into the delay buffer (1020) before a further frame is received by the apparatus.

6. An apparatus according to claim 4, wherein the sample processor (1040) is configured to store the reconstructed audio signal samples into the delay buffer (1020) after a further frame is received by the apparatus.

7. An apparatus according to claim 1, wherein the sample processor (1040) is configured to rescale the selected audio signal samples depending on the gain information to obtain rescaled audio signal samples and by combining the rescaled audio signal samples with input audio signal samples to obtain the processed audio signal samples.

8. An apparatus according to claim 7,

wherein the sample processor (1040) is configured to store the processed audio signal samples, indicating the combination of the rescaled audio signal samples and the input audio signal samples, into the delay buffer (1020), and to not store the rescaled audio signal samples into the delay buffer (1020), if the current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, and

wherein the sample processor (1040) is configured to store the rescaled audio signal samples into the delay buffer (1020) and to not store the processed audio signal samples into the delay buffer (1020), if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted.

9. An apparatus according to claim 7, wherein the sample processor (1040) is configured to store the processed audio signal samples into the delay buffer

(1020), if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted.

10. An apparatus according to one of the preceding claims, wherein the sample selector (1030) is configured to calculate the modified gain.

11. An apparatus according to one of the preceding claims, wherein the modified gain `gain` is set to zero, if at least a predefined number of frames have not been received by the apparatus since a frame last has been received by the apparatus.

12. A method for decoding an encoded audio signal to obtain a reconstructed audio signal, wherein the method comprises:

receiving a plurality of frames,

decoding the plurality of frames by conducting an inverse modified discrete cosine transform to obtain audio signal samples of the decoded audio signal,

conducting long-term prediction by

storing the audio signal samples of the decoded audio signal,

selecting a plurality of selected audio signal samples from the audio signal samples being stored in a delay buffer (1020), and

processing the selected audio signal samples to obtain reconstructed audio signal samples of the reconstructed audio signal,

wherein, if a current frame is received and if the current frame being received is not corrupted, the step of selecting the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer (1020) is conducted depending on a pitch lag information being comprised by the current frame, and

wherein, if the current frame is not received or if the current frame being received is corrupted, the step of selecting the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer (1020) is conducted

depending on a pitch lag information being comprised by another frame being received previously,

wherein the method further comprises the step of rescaling the selected audio signal samples depending on a modified gain, wherein the modified gain is defined according to the formula:

```
gain = gain_past * damping;
```

wherein `gain` is the modified gain,

wherein `gain_past` is set to `gain` after `gain` has been calculated, and

wherein `damping` is a real value with $0 \leq$ `damping` ≤ 1.

13. A computer program for implementing the method of claim 12 when being executed on a computer or signal processor.

Claims

1. An apparatus for decoding an encoded audio signal to obtain a reconstructed audio signal, wherein the apparatus is configured to receive a plurality of frames, and wherein the apparatus comprises:

an inverse modified discrete cosine transform module for decoding the plurality of frames by conducting an inverse modified discrete cosine transform to obtain audio signal samples of the decoded audio signal, and

a long-term prediction unit for conducting long-term prediction, comprising:

a delay buffer for storing the audio signal samples of the decoded audio signal,

a sample selector for selecting a plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer, and

a sample processor for processing the selected audio signal samples to obtain reconstructed audio signal samples of the reconstructed audio signal,

wherein the sample selector is configured to select, if a current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer depending on a pitch lag information being comprised by the current frame, and

wherein the sample selector is configured to select, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted, the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer depending on a pitch lag information being comprised by another frame being received previously by the apparatus,

wherein the sample selector is configured to obtain the reconstructed audio signal samples by rescaling the selected audio signal samples depending on a modified gain, wherein the modified gain is defined according to the formula:

```
gain = gain_past * damping;
```

wherein `gain` is the modified gain,

wherein the sample selector is configured to set `gain_past` to `gain` after `gain` has been calculated, and

wherein `damping` is a real value with $0 \leq damping \leq 1$.

2. An apparatus according to claim 1,

wherein the sample processor is configured to obtain the reconstructed audio signal samples, if the current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, by rescaling the selected audio signal samples depending on a gain information being comprised by the current frame, and

wherein the sample selector is configured to obtain the reconstructed audio signal samples, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted, by rescaling the selected audio signal samples depending on the gain information being comprised by said another frame being received previously by the apparatus.

3. An apparatus according to claim 2,

wherein the sample processor is configured to obtain the reconstructed audio signal samples, if the current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, by multiplying the selected audio signal samples and a value depending on the gain information being comprised by the current frame, and

wherein the sample selector is configured to obtain the reconstructed audio signal samples, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted, by multiplying the

selected audio signal samples and a value depending on the gain information being comprised by said another frame being received previously by the apparatus.

4. An apparatus according to one of the preceding claims, wherein the sample processor is configured to store the reconstructed audio signal samples into the delay buffer.

5. An apparatus according to claim 4, wherein the sample processor is configured to store the reconstructed audio signal samples into the delay buffer before a further frame is received by the apparatus.

6. An apparatus according to claim 4, wherein the sample processor is configured to store the reconstructed audio signal samples into the delay buffer after a further frame is received by the apparatus.

7. An apparatus according to claim 1, wherein the sample processor is configured to rescale the selected audio signal samples depending on the gain information to obtain rescaled audio signal samples and by combining the rescaled audio signal samples with input audio signal samples to obtain the processed audio signal samples.

8. An apparatus according to claim 7,

wherein the sample processor is configured to store the processed audio signal samples, indicating the combination of the rescaled audio signal samples and the input audio signal samples, into the delay buffer, and to not store the rescaled audio signal samples into the delay buffer, if the current frame is received by the apparatus and if the current frame being received by the apparatus is not corrupted, and

wherein the sample processor is configured to store the rescaled audio signal samples into the delay buffer and to not store the processed audio signal samples into the delay buffer, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted.

9. An apparatus according to claim 7, wherein the sample processor is configured to store the processed audio signal samples into the delay buffer

, if the current frame is not received by the apparatus or if the current frame being received by the apparatus is corrupted.

10. An apparatus according to one of the preceding claims, wherein the sample selector is configured to calculate the modified gain.

11. An apparatus according to one of the preceding claims, wherein the modified gain `gain` is set to zero, if at least a predefined number of frames have not been received by the apparatus since a frame last has been received by the apparatus.

12. A method for decoding an encoded audio signal to obtain a reconstructed audio signal, wherein the method comprises:

receiving a plurality of frames,

decoding the plurality of frames by conducting an inverse modified discrete cosine transform to obtain audio signal samples of the decoded audio signal,

conducting long-term prediction by

storing the audio signal samples of the decoded audio signal,

selecting a plurality of selected audio signal samples from the audio signal samples being stored in a delay buffer, and

processing the selected audio signal samples to obtain reconstructed audio signal samples of the reconstructed audio signal,

wherein, if a current frame is received and if the current frame being received is not corrupted, the step of selecting the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer is conducted depending on a pitch lag information being comprised by the current frame, and

wherein, if the current frame is not received or if the current frame being received is corrupted, the step of selecting the plurality of selected audio signal samples from the audio signal samples being stored in the delay buffer is conducted

depending on a pitch lag information being comprised by another frame being received previously,

wherein the method further comprises the step of rescaling the selected audio signal samples depending on a modified gain, wherein the modified gain is defined according to the formula:

```
gain = gain_past * damping;
```

wherein `gain` is the modified gain,

wherein `gain_past` is set to `gain` after `gain` has been calculated, and

wherein `damping` is a real value with $0 \leq damping \leq 1$.

13. A computer program for implementing the method of claim 12 when being executed on a computer or signal processor.