CLAIMS

1. A method for decoding an encoded input signal to generate an audio output signal, wherein the method comprises:

receiving the encoded input signal that includes encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and optionally including one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that were set according to an encoding process that generated the encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that were set according to the encoding process that generated the encoded audio information to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level that is higher than the first reference reproduction level;

applying a decoding process to the encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

modifying the subband signals to obtain modified subband signals with changed dynamic range characteristics, wherein the modifying is adapted in response to the one or more second parameters if the metadata includes the one or more second parameters or is adapted in response to the one or more first parameters if the metadata does not include the one or more second parameters;

applying a synthesis filter bank to the modified subband signals to obtain a time-domain audio signal; and

if the metadata does not include the one or more second parameters, applying a gain and a limiter to the time-domain audio signal in response to the metadata, wherein the application of the gain modifies the time-domain audio signal to obtain the audio output signal with amplitudes for playback at the second reference reproduction level, and wherein the application of the limiter prevents the amplitudes of the audio output signal from exceeding the clipping level.

2. The method of claim 1, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

3. The method of claim 1 or 2, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

4. A method for encoding an audio input signal representing aural stimuli, wherein the method comprises:

receiving the audio input signal;

applying an analysis filter bank to the audio input signal to generate subband signals representing spectral content of the audio input signal;

analyzing one or more signals derived from the audio input signal to calculate metadata including one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level;

applying an encoding process to the subband signals to obtain encoded audio information; and

assembling the encoded audio information and the metadata into an encoded output signal having a format suitable for transmission or storage, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

5. The method of claim 4, wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

6. A method for transcoding an encoded input signal to generate an encoded output signal, wherein the method comprises:

receiving the encoded input signal that includes first encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile, wherein the one or more first parameters have values that were set according to a first encoding process that generated the first encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level;

applying a decoding process to the first encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

analyzing one or more signals obtained from the subband signals to calculate one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level; and

assembling second encoded audio information, the one or more first parameters and the one or more second parameters into an encoded output signal having a format suitable for transmission or storage, wherein the second encoded audio information is an encoded representation of the subband signals.

7. The method of claim 6, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

8. The method of claim 6 or 7 that comprises applying a synthesis filter bank to the subband signals to obtain the one or more signals that are analyzed to calculate the one or more second parameters specifying dynamic range compression.

9. The method of any one of claims 6 through 8 that comprises applying a second encoding process to the subband signals to generate the second encoded audio information.

10. The method of any one of claims 6 through 8, wherein the second encoded audio information is the first encoded audio information.

11. The method of any one of claims 6 through 10, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level.

12. The method of any one of claims 6 through 10, wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

13. An apparatus that comprises means for performing the steps of the method for any one of claims 1 through 12.

14. A storage medium recording a program of instructions that is executable by a device to perform the steps of the method for any one of claims 1 through 12.

CLAIMS

1. A method for decoding an encoded input signal to generate an audio output signal, wherein the method comprises:

receiving the encoded input signal that includes encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and optionally including one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that were set according to an encoding process that generated the encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that were set according to the encoding process that generated the encoded audio information to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level that is higher than the first reference reproduction level;

applying a decoding process to the encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

modifying the subband signals to obtain modified subband signals with changed dynamic range characteristics, wherein the modifying is adapted in response to the one or more second parameters if the metadata includes the one or more second parameters or is adapted in response to the one or more first parameters if the metadata does not include the one or more second parameters;

applying a synthesis filter bank to the modified subband signals to obtain a time-domain audio signal; and

if the metadata does not include the one or more second parameters, applying a gain and a limiter to the time-domain audio signal in response to the metadata, wherein the application of the gain modifies the time-domain audio signal to obtain the audio output signal with amplitudes for playback at the second reference reproduction level, and wherein the application of the limiter prevents the amplitudes of the audio output signal from exceeding the clipping level.

2. The method of claim 1, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

3. A method for encoding an audio input signal representing aural stimuli, wherein the method comprises:

receiving the audio input signal;

applying an analysis filter bank to the audio input signal to generate subband signals representing spectral content of the audio input signal;

analyzing one or more signals derived from the audio input signal to calculate metadata including one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level;

applying an encoding process to the subband signals to obtain encoded audio information; and

assembling the encoded audio information and the metadata into an encoded output signal having a format suitable for transmission or storage, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

4. The method of claim 3, wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

5. A method for transcoding an encoded input signal to generate an encoded output signal, wherein the method comprises:

receiving the encoded input signal that includes first encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile, wherein the one or more first parameters have values that were set according to a first encoding process that generated the first encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level;

applying a decoding process to the first encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

analyzing one or more signals obtained from the subband signals to calculate one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level; and

assembling second encoded audio information, the one or more first parameters and the one or more second parameters into an encoded output signal having a format suitable for transmission or storage, wherein the second encoded audio information is an encoded representation of the subband signals.

6. The method of claim 5, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

7. The method of claim 5 that comprises applying a synthesis filter bank to the subband signals to obtain the one or more signals that are analyzed to calculate the one or more second parameters specifying dynamic range compression.

8. The method of claim 5, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

9. An apparatus for decoding an encoded input signal to generate an audio output signal, wherein the method comprises:

means for receiving the encoded input signal that includes encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and optionally including one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that were set according to an encoding process that generated the encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that were set according to the encoding process that generated the encoded audio information

to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level that is higher than the first reference reproduction level;

means for applying a decoding process to the encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

means for modifying the subband signals to obtain modified subband signals with changed dynamic range characteristics, wherein the modifying is adapted in response to the one or more second parameters if the metadata includes the one or more second parameters or is adapted in response to the one or more first parameters if the metadata does not include the one or more second parameters;

means for applying a synthesis filter bank to the modified subband signals to obtain a time-domain audio signal; and

means for applying a gain and a limiter to the time-domain audio signal in response to the metadata if the metadata does not include the one or more second parameters, wherein the application of the gain modifies the time-domain audio signal to obtain the audio output signal with amplitudes for playback at the second reference reproduction level, and wherein the application of the limiter prevents the amplitudes of the audio output signal from exceeding the clipping level.

10. The apparatus of claim 9, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

11. An apparatus for encoding an audio input signal representing aural stimuli, wherein the method comprises:

means for receiving the audio input signal;

means for applying an analysis filter bank to the audio input signal to generate subband signals representing spectral content of the audio input signal;

means for analyzing one or more signals derived from the audio input signal to calculate metadata including one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level;

means for applying an encoding process to the subband signals to obtain encoded audio information; and

means for assembling the encoded audio information and the metadata into an encoded output signal having a format suitable for transmission or storage, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

12. The apparatus of claim 11, wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

13. An apparatus for transcoding an encoded input signal to generate an encoded output signal, wherein the method comprises:

means for receiving the encoded input signal that includes first encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile, wherein the one or more first parameters have values that were set according to a first encoding process that generated the first encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level;

means for applying a decoding process to the first encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

means for analyzing one or more signals obtained from the subband signals to calculate one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level; and

means for assembling second encoded audio information, the one or more first parameters and the one or more second parameters into an encoded output signal having a format suitable for transmission or storage, wherein the second encoded audio information is an encoded representation of the subband signals.

14. The apparatus of claim 13, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

15. The apparatus of claim 13 that comprises means for applying a synthesis filter bank to the subband signals to obtain the one or more signals that are analyzed to calculate the one or more second parameters specifying dynamic range compression.

16. The apparatus of claim 13, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

17. A non-transitory medium recording a program of instructions that is executable by a device to perform a method for decoding an encoded input signal to generate an audio output signal, wherein the method comprises:

receiving the encoded input signal that includes encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and optionally including one or more second parameters specifying' dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that were set according to an encoding process that generated the encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that were set according to the encoding process that generated the encoded audio information to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level that is higher than the first reference reproduction level;

applying a decoding process to the encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

modifying the subband signals to obtain modified subband signals with changed dynamic range characteristics, wherein the modifying is adapted in response to the one or more second parameters if the metadata includes the one or more second parameters or is adapted in response to the one or more first parameters if the metadata does not include the one or more second parameters;

applying a synthesis filter bank to the modified subband signals to obtain a time-domain audio signal; and

if the metadata does not include the one or more second parameters, applying a gain and a limiter to the time-domain audio signal in response to the metadata, wherein the application of the gain modifies the time-domain audio signal to obtain the audio output signal with amplitudes for playback at the second reference reproduction level, and wherein the application of the limiter prevents the amplitudes of the audio output signal from exceeding the clipping level.

18. The medium of claim 17, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

19. A non-transitory medium recording a program of instructions that is executable by a device to perform a method for encoding an audio input signal representing aural stimuli, wherein the method comprises:

receiving the audio input signal;

applying an analysis filter bank to the audio input signal to generate subband signals representing spectral content of the audio input signal;

analyzing one or more signals derived from the audio input signal to calculate metadata including one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile and one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more first parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level, and wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level;

applying an encoding process to the subband signals to obtain encoded audio information; and

assembling the encoded audio information and the metadata into an encoded output signal having a format suitable for transmission or storage, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

20. The medium of claim 19, wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.

21. A non-transitory medium recording a program of instructions that is executable by a device to perform a method for transcoding an encoded input signal to generate an encoded output signal, wherein the method comprises:

receiving the encoded input signal that includes first encoded audio information and associated metadata including one or more decoding-control parameters and one or more first parameters specifying dynamic range compression according to a first dynamic range compression profile, wherein the one or more first parameters have values that were set according to a first encoding process that generated the first encoded audio information to represent aural stimuli with amplitudes that do not exceed a clipping level for playback at a first reference reproduction level;

applying a decoding process to the first encoded audio information to obtain subband signals representing spectral content of the aural stimuli, wherein the decoding process is adapted in response to the one or more decoding-control parameters;

analyzing one or more signals obtained from the subband signals to calculate one or more second parameters specifying dynamic range compression according to a second dynamic range compression profile, wherein the one or more second parameters have values that are set to represent the aural stimuli with amplitudes that do not exceed the clipping level for playback at a second reference reproduction level; and

assembling second encoded audio information, the one or more first parameters and the one or more second parameters into an encoded output signal having a format suitable for transmission or storage, wherein the second encoded audio information is an encoded representation of the subband signals.

22. The medium of claim 21, wherein the one or more second parameters represent differences between corresponding parameters for the first dynamic range compression profile and the second dynamic range compression profile.

23. The medium of claim 21, wherein the method comprises applying a synthesis filter bank to the subband signals to obtain the one or more signals that are analyzed to calculate the one or more second parameters specifying dynamic range compression.

24. The medium of claim 21, wherein the encoded input signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the first reference reproduction level corresponds to an amplitude 20 dB below the clipping level, and wherein the encoded output signal conforms to the ATSC Standard, the MPEG-2 AAC Standard, or the MPEG-4 Audio Standard, and the second reference reproduction level corresponds to an amplitude 11 dB below the clipping level.