CLAIMS

1. A method of expanding an audio signal comprising:

receiving an audio signal; and

expanding the audio signal to an expanded dynamic range through an expansion process comprising: dividing the received audio signal into a plurality of time-segments using a defined window shape, calculating a wideband gain for each time-segment in the frequency domain using a non-energy based average of a frequency domain representation of the audio signal, and applying individual gain values to each time-segment to obtain the expanded dynamic range audio signal, wherein the application of the individual gain values amplifies segments of relatively high intensity and attenuates segments of relatively low intensity.

2. The method of claim 1 wherein the segments are overlapping.

3. The method of claim 2 wherein a first filterbank is used to analyze the audio signal to obtain a frequency domain representation, and the defined window shape corresponds to a prototype filter for the first filterbank.

4. The method of claim 3 wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

5. The method of claim 3 wherein the wideband gain for each time segment is calculated using the subband samples in a subset of subbands in the respective time segment.

6. The method of claim 5 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank.

7. The method of claim 5 wherein the gain for each respective time segment is derived from the p-norm of the subband samples in each time segment where p is a positive real number not equal to two.

8. The method of claim 5 wherein the wideband gain is applied in the domain of the first filterbank.

9. The method of claim 8 wherein each wideband gain value is calculated from a first subset of subbands of the first filterbank and applied to a second subset of subbands of the first filterbank, wherein the second set of subbands includes the first subset of subbands.

10. The method of claim 9 wherein the first and second subset of subbands are identical and correspond to a low frequency region of the audio signal.

11. The method of claim 9 wherein the first subset of subbands corresponds to a low frequency region of the audio signal and the second subset of subbands corresponds to an entire frequency range spanned by the first filterbank.

12. The method of claim 3 wherein the received audio signal has been previously compressed by a process comprising:

receiving an initial audio signal; and

compressing the initial audio signal to substantially reduce an original dynamic range of the initial audio signal through a compression process comprising dividing the initial audio signal into a plurality of segments using the defined window shape, calculating a wideband gain for each segment using a non-energy based average of frequency domain samples of the initial audio signal, and applying the gain values calculated from the initial audio signal to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.

13. The method of claim 12 wherein the wideband gain calculated by the expansion process is substantially the inverse of the wideband gain calculated for the corresponding time-segment by the compression process.

14. The method of claim 12 wherein a second filterbank is used in the compression process to analyze the initial audio signal to obtain its frequency domain representation, and the defined window shape for segmentation is the same as the prototype filter for the filterbank, and further wherein the second filterbank is the same as the first filterbank.

15. The method of claim 12 wherein the received signal for the expansion process is obtained after modification of the compressed signal by an audio encoder that generates a bitstream and decoder that decodes the bitstream.

16. The method of claim 15, wherein the audio encoder and the decoder are both transform-based; and wherein the time segments of the audio signals in the compression process and the expansion process are substantially shorter than a single window length of the transform in the audio encoder and decoder.

17. The method of claim 15 further comprising:

generating control information determining an activation state of the expansion process; and

transmitting the control information in a bitstream that is transmitted from the encoder to the decoder.

18. The method of claim 17 wherein the audio signal in the bitstream is segmented into frames with each frame corresponding to a plurality of time-segments of the expansion process, and wherein the activation state is selected from the group consisting of: applying the expansion process on each time-segment in a frame; not applying the expansion for any time-segment in a frame; applying the expansion process on each time-segment in a frame with a modified gain calculation, wherein the gain applied in each of the time-segments is the average gain of all time-segments in the frame; applying the expansion process on each time-segment in a frame with a modified gain calculation resulting in a gain value that is intermediate to not applying the expansion process at all; using a stop frame to fade out from a frame where the expansion process is applied and fade in to a frame where the expansion process is not applied; using a start frame to fade out from a frame where the expansion process is not applied and fade in to a frame where the expansion process is applied; and applying the expansion process completely.

19. The method of claim 18 wherein, the control information for the expansion process is determined by the compressing step based on one or more characteristics of the initial audio signal, including at least one of a content type of the audio signal and a stationary versus transient nature of the audio signal.

20. The method of claim 19 wherein the control information is determined such that switching between activation states minimizes an occurrence of signal discontinuities.

21. The method of claim 20 wherein, the control information also controls the compression process and has the effect of turning off the compression process when the expansion process will also be switched off; turning on the compression process when the expansion process will also be switched on; and enabling a modified gain calculation for compression when a modified gain calculation for expansion will be effected; using a stop frame when a stop frame will be used at the expander; and using a start frame when a start frame will be used at the expander.

22. The method of claim 15 wherein:

the compressed audio signal and the audio signal received by the expander have a number, N, channels, where N is greater than 1;

the channels are grouped into one or more discrete subsets;

the grouping at the compressor and the expander being the same;

the channels in each group compressed sharing the same gains at the compressor and expanded sharing the same gains at the expander.

23. The method of 22 wherein the grouping is pre-determined and known at the compressor and expander.

24. The method of 23 wherein each group contains exactly one channel and there are N groups.

25. The method of 22 wherein the grouping of channels comprises:

calculating at the compressor a metric of similarity between channels;

grouping similar channels together based on the similarity metric; and

sending the grouping information through the bitstream.

26. The method of claim 22 wherein at least two channels are encoded as a mono downmix with additional parametric spatial information that is applied in the first filterbank domain to reconstruct a stereo output, and wherein the parametric spatial information is used either above a defined frequency with discrete stereo information used below the defined

frequency, or parametric spatial information is used below the defined frequency with discrete stereo information used above the defined frequency.

27. A method of compressing an audio signal comprising:

receiving an initial audio signal; and

compressing the initial audio signal to substantially reduce an original dynamic range of the initial audio signal through a compression process comprising dividing the initial audio signal into a plurality of segments using a defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal, and applying individual gain value to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.

28. The method of claim 27 wherein the segments are overlapping and wherein a first filterbank is used to analyze the audio signal to obtain a frequency domain representation and the defined window shape corresponds to a prototype filter for the first filterbank.

29. The method of claim 28 wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

30. The method of claim 28 wherein each individual gain value is calculated using the subband samples in a subset of subbands in a respective time segment.

31. The method of claim 30 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank, and wherein the gain is applied in the domain of the first filterbank.

32. The method of claim 30 wherein the gain for each time segment is derived from the p-norm of the subband samples in each time segment where p is a positive real number not equal to two.

33. The method of claim 30 wherein the gain is calculated from a first subset of subbands of the first filterbank and applied to a second subset of subbands of the first filterbank, wherein the second set of subbands includes the first subset of subbands.

34. The method of claim 33 wherein the first and second subset of subbands are identical and correspond to a low frequency region of the audio signal.

35. The method of claim 33 wherein the first subset of subbands corresponds to a low frequency region of the audio signal and the second subset of subbands corresponds to an entire frequency range spanned by the first filterbank.

36. The method of claim 27 further comprising: transmitting a compressed version of the initial audio signal to a expansion component performing an expansion process comprising receiving the compressed version of audio signal; and expanding the compressed version of the audio signal to substantially restore it to an original dynamic range of the initial audio signal through process comprising dividing the initial audio signal into a plurality of time segments using the defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal; and applying individual gain values of the wideband gain to each time segment to amplify segments of relatively high intensity and attenuate segments of relatively low intensity.

37. The method of claim 36 wherein the gain calculated by the compressing step is substantially the inverse of the gain calculated by the expanding step for the same time-segment.

38. The method of claim 36 wherein a second filterbank is used in the expansion process to analyze the initial audio signal to obtain its frequency domain representation, and the defined window shape for segmentation is the same as the prototype filter for the filterbank, and further wherein the second filterbank is the same as the first filterbank.

39. The method of claim 36 wherein the received signal for the expanding step is obtained after modification of the compressed signal by an audio encoder that generates a bitstream and decoder that decodes the bitstream.

40. The method of claim 39, wherein the audio encoder and the decoder are both transform-based; and wherein the time segments of the audio signals in the compressing and

expanding steps are substantially shorter than a single window length of the transform in the audio encoder and decoder.

41. The method of claim 39 further comprising:

generating control information determining an activation state of the expanding step; and

transmitting the control information in a bitstream that is transmitted from the encoder to the decoder.

42. The method of claim 41 wherein the audio signal in the bitstream is segmented into frames with each frame corresponding to a plurality of time-segments of the expansion process, and wherein the activation state is selected from the group consisting of: applying the expansion process on each time-segment in a frame; not applying the expansion for any time-segment in a frame; applying the expansion process on each time-segment in a frame with a modified gain calculation, wherein the gain applied in each of the time-segments is the average gain of all time-segments in the frame; applying the expansion process on each time-segment in a frame with a modified gain calculation resulting in a gain value that is intermediate to not applying the expansion process at all; using a stop frame to fade out from a frame where the expansion process is applied and fade in to a frame where the expansion process is not applied; using a start frame to fade out from a frame where the expansion process is not applied and fade in to a frame where the expansion process is applied; and applying the expansion process completely.

43. The method of claim 42 wherein, the control information for the expansion process is determined by the compressing step based on one or more characteristics of the initial audio signal, including at least one of a content type of the audio signal and a stationary versus transient nature of the audio signal.

44. The method of claim 43 wherein the control information is determined such that switching between activation states minimizes an occurrence of signal discontinuities.

45. The method of claim 44 wherein, the control information also controls the compression process and has the effect of turning off the compressing step when the expanding step will also be switched off; turning on the compressing step when the

expanding step will also be switched on; and enabling a modified gain calculation for compression when a modified gain calculation for expansion will be effected; using a stop frame when a stop frame will be used at the expander; and using a start frame when a start frame will be used at the expander.

46. The method of claim 39 wherein:

the compressed audio signal and the audio signal received by the expander have a number, N, channels, where N is greater than 1;

the channels are grouped into one or more discrete subsets;

the grouping at the compressor and the expander being the same;

the channels in each group compressed sharing the same gains at the compressor and expanded sharing the same gains at the expander.

47. The method of 46 wherein the grouping is pre-determined and known at the compressor and expander.

48. The method of 47 wherein each group contains exactly one channel and there are N groups.

49. The method of 46 wherein the grouping of channels comprises:

calculating at the compressor a metric of similarity between channels;

grouping similar channels together based on the similarity metric; and

sending the grouping information through the bitstream.

50. The method of claim 49 wherein at least two channels are encoded as a mono downmix with additional parametric spatial information that is applied in the first filterbank domain to reconstruct a stereo output, and wherein the parametric spatial information is used either above a defined frequency with discrete stereo information used below the defined frequency, or parametric spatial information is used below the defined frequency with discrete stereo information used above the defined frequency.

51. An apparatus for compressing an audio signal comprising:

a first interface receiving an initial audio signal; and

a compressor compressing the initial audio signal to substantially reduce an original dynamic range of the initial audio signal by dividing the initial audio signal into a plurality of segments using a defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal, and applying individual gain values to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.

52. The apparatus of claim 51 further comprising a first filterbank analyzing the audio signal to obtain a frequency domain representation and wherein the defined window shape corresponds to a prototype filter for the first filterbank, and further wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

53. The apparatus of claim 52 wherein the individual gain values are calculated using the subband samples in a subset of subbands in each respective time segment.

54. The apparatus of claim 53 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank, and wherein the gain is applied in the domain of the first filterbank.

55. The apparatus of claim 52 further comprising a second interface transmitting a compressed version of the initial audio signal to an expander that receives the compressed version of audio signal, and expands the compressed version of the audio signal to substantially restore it to an original dynamic range of the initial audio signal by dividing the initial audio signal into a plurality of segments using the defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal; and applying a respective gain value to each segment of the plurality of segments to amplify segments of relatively high intensity and attenuate segments of relatively low intensity.

56. The apparatus of claim 55 wherein the gain calculated by the expander is substantially the inverse of the gain calculated by the compressor for the same time segment.

57. The apparatus of claim 55 further comprising a second filterbank analyzing the initial audio signal to obtain its frequency domain representation, and wherein the defined window shape for segmentation is the same as the prototype filter for the filterbank, and further wherein the second filterbank is the same as the first filterbank.

58. The apparatus of claim 55 further comprising encoder and decoder stages of an audio codec configured to transmit a bitstream of the compressed version of the audio signal from the compressor to the expander, and wherein the encoder and decoder are both transform-based.

59. The apparatus of claim 58 further comprising a control component generating control information determining an activation state of the expander and transmitting the control information in the bitstream, and wherein, the control information for the expansion process is determined by the compressor based on one or more characteristics of the initial audio signal, including at least one of a content type of the audio signal and a stationary versus transient nature of the audio signal.

60. The apparatus of claim 55 further comprising a parametric spatial information component applying parametric spatial information in the first filterbank domain to reconstruct a stereo output, and wherein the parametric spatial information is used either above a defined frequency with discrete stereo information used below the defined frequency, or parametric spatial information is used below the defined frequency with discrete stereo information used above the defined frequency.

61. An apparatus for expanding an audio signal comprising:

a first interface receiving a compressed audio signal; and

an expander expanding the compressed audio signal to substantially restore its original uncompressed dynamic range by dividing the initial audio signal into a plurality of segments using a defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal, and applying individual gain values to each segment of the plurality of segments to amplify segments of relatively high intensity and attenuate segments of relatively low intensity.

62. The apparatus of claim 61 further comprising a first filterbank analyzing the audio signal to obtain a frequency domain representation and wherein the defined window shape corresponds to a prototype filter for the first filterbank, and further wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

63. The apparatus of claim 62 wherein the wideband gain comprises an individual gain value for each time segment, and wherein each individual gain value is calculated using the subband samples in a subset of subbands in each respective time segment.

64. The apparatus of claim 63 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank, and wherein the gain is applied in the domain of the first filterbank.

65. The apparatus of claim 62 further comprising a second interface receiving the compressed audio signal from a compressor that receives an initial audio signal, and compresses the initial audio signal to substantially reduce the original dynamic range of the initial audio signal by dividing the initial audio signal into a plurality of segments using the defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal; and applying a respective gain value to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.

66. The apparatus of claim 65 wherein the gain calculated by the compressor is substantially the inverse of the gain calculated by the expander for the same time segment.

67. The apparatus of claim 65 further comprising a second filterbank analyzing the initial audio signal to obtain its frequency domain representation, and wherein the defined window shape for segmentation is the same as the prototype filter for the filterbank, and further wherein the second filterbank is the same as the first filterbank.

68. The apparatus of claim 65 further comprising encoder and decoder stages of an audio codec configured to transmit a bitstream of the compressed version of the audio signal from the compressor to the expander, and wherein the encoder and decoder are both transform-based.

69. The apparatus of claim 68 further comprising a control component generating control information determining an activation state of the expander and transmitting the control information in the bitstream, and wherein, the control information for the expansion process is determined by the compressor based on one or more characteristics of the initial audio signal, including at least one of a content type of the audio signal and a stationary versus transient nature of the audio signal.

70. The apparatus of claim 65 further comprising a parametric spatial information component applying parametric spatial information in the first filterbank domain to reconstruct a stereo output, and wherein the parametric spatial information is used either above a defined frequency with discrete stereo information used below the defined frequency, or parametric spatial information is used below the defined frequency with discrete stereo information used above the defined frequency.

CLAIMS

1. A method of expanding an audio signal comprising:

receiving an audio signal; and

expanding the audio signal to an expanded dynamic range through an expansion process comprising: dividing the received audio signal into a plurality of time-segments using a defined window shape, calculating a wideband gain for each time-segment in the frequency domain using a non-energy based average of a frequency domain representation of the audio signal, and applying individual gain values to each time-segment to obtain the expanded dynamic range audio signal, wherein the application of the individual gain values amplifies segments of relatively high intensity and attenuates segments of relatively low intensity.

2. The method of claim 1 wherein the segments are overlapping.

3. The method of claim 2 wherein a first filterbank is used to analyze the audio signal to obtain a frequency domain representation, and the defined window shape corresponds to a prototype filter for the first filterbank.

4. The method of claim 3 wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

5. The method of claim 3 wherein the wideband gain for each time segment is calculated using the subband samples in a subset of subbands in the respective time segment.

6. A method of compressing an audio signal comprising:

receiving an initial audio signal; and

compressing the initial audio signal to substantially reduce an original dynamic range of the initial audio signal through a compression process comprising dividing the initial audio signal into a plurality of segments using a defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency

domain samples of the initial audio signal, and applying individual gain value to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.

7. The method of claim 6 wherein the segments are overlapping and wherein a first filterbank is used to analyze the audio signal to obtain a frequency domain representation and the defined window shape corresponds to a prototype filter for the first filterbank.

8. The method of claim 7 wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

9. The method of claim 7 wherein each individual gain value is calculated using the subband samples in a subset of subbands in a respective time segment.

10. The method of claim 9 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank, and wherein the gain is applied in the domain of the first filterbank.

11. An apparatus for compressing an audio signal comprising:

a first interface receiving an initial audio signal; and

a compressor compressing the initial audio signal to substantially reduce an original dynamic range of the initial audio signal by dividing the initial audio signal into a plurality of segments using a defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal, and applying individual gain values to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.

12. The apparatus of claim 11 further comprising a first filterbank analyzing the audio signal to obtain a frequency domain representation and wherein the defined window shape corresponds to a prototype filter for the first filterbank, and further wherein the first

filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

13. The apparatus of claim 12 wherein the individual gain values are calculated using the subband samples in a subset of subbands in each respective time segment.

14. The apparatus of claim 13 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank, and wherein the gain is applied in the domain of the first filterbank.

15. The apparatus of claim 12 further comprising a second interface transmitting a compressed version of the initial audio signal to an expander that receives the compressed version of audio signal, and expands the compressed version of the audio signal to substantially restore it to an original dynamic range of the initial audio signal by dividing the initial audio signal into a plurality of segments using the defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal; and applying a respective gain value to each segment of the plurality of segments to amplify segments of relatively high intensity and attenuate segments of relatively low intensity.

16. An apparatus for expanding an audio signal comprising:

a first interface receiving a compressed audio signal; and

an expander expanding the compressed audio signal to substantially restore its original uncompressed dynamic range by dividing the initial audio signal into a plurality of segments using a defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal, and applying individual gain values to each segment of the plurality of segments to amplify segments of relatively high intensity and attenuate segments of relatively low intensity.

17. The apparatus of claim 16 further comprising a first filterbank analyzing the audio signal to obtain a frequency domain representation and wherein the defined window shape corresponds to a prototype filter for the first filterbank, and further wherein the first filterbank is one of a quadrature modulated filter (QMF) bank or a short-time Fourier transform.

18. The apparatus of claim 17 wherein the wideband gain comprises an individual gain value for each time segment, and wherein each individual gain value is calculated using the subband samples in a subset of subbands in each respective time segment.

19. The apparatus of claim 18 wherein the subset of subbands corresponds to the entire frequency range spanned by the first filterbank, and wherein the gain is applied in the domain of the first filterbank.

20. The apparatus of claim 16 further comprising a second interface receiving the compressed audio signal from a compressor that receives an initial audio signal, and compresses the initial audio signal to substantially reduce the original dynamic range of the initial audio signal by dividing the initial audio signal into a plurality of segments using the defined window shape, calculating a wideband gain in the frequency domain using a non-energy based average of frequency domain samples of the initial audio signal; and applying a respective gain value to each segment of the plurality of segments to amplify segments of relatively low intensity and attenuate segments of relatively high intensity.