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CLAIMS

1. A method for reducing distortion artifacts preceding a signal transient in an audio signal stream processed by a transform-based low-bit-rate audio coding system employing coding blocks, comprising

detecting a transient in the audio signal stream prior to processing by said coding system, and

shifting the temporal relationship of said transient with respect to said coding blocks such that the time duration of said distortion artifacts is reduced.

2. The method of claim 1 wherein said shifting shifts the temporal relationship of said transient with respect to said coding blocks prior to forward transforming in the encoder of said coding system.

3. The method of claim 2 wherein said transient is shifted to a temporal position closely following the next block end or closely following the last block end.

4. The method of claim 3 wherein said transient is shifted to a temporal position closely following the next block end or closely following the last block end which results in the shorter shift of temporal position.

5. A method according to claim 1 or claim 3 further comprising removing at least a portion of remaining distortion artifacts after inverse transformation in the decoder of said coding system.

6. The method of claim 5 wherein the portion of remaining distortion artifacts is determined at least in part by metadata information carried in said coding system.

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7. The method of claim 5 wherein the portion of remaining distortion artifacts is determined at least in part by a default parameter.

8. The method of claim 5 wherein the portion of remaining distortion artifacts is determined at least in part by a measure of high frequency audio components in said audio signal steam.

9. The method of claim 2 or claim 3 wherein the temporal relationship of said transient with respect to said coding blocks is shifted by time scaling a segment of said audio signal stream preceding said signal transient.

10. The method of claim 9 further comprising applying a compensating time scaling to the audio signal stream subsequent to inverse transformation in the decoder of said coding system such that the time evolution of the processed audio signal stream is substantially the same as that of the audio signal stream prior to said shifting.

11. The method of claim 10 wherein said compensating time scaling is applied to a segment of said audio signal stream preceding said signal transient.

12. The method of claim 10 wherein said coding system includes an encoder and a decoder, said encoder transmitting metadata to said decoder along with an encoded version of said audio signal stream, said metadata including information useful for applying said compensating time scaling.

13. The method of claim 9 wherein said time scaling is performed on a segment of said audio stream closely preceding said transient.

14. The method of claim 13 wherein said time scaling is performed on a segment of said audio stream that is at least partially temporally pre-masked by transient.

15. The method of claim 9 wherein said time scaling has the effect of deleting signal components from or adding signal components to the audio signal stream applied to the coding system.

16. The method of claim 15 wherein a further time scaling is applied following said signal transient, said further time scaling acting in the opposite sense to the said first-recited time scaling.

17. The method of claim 16 wherein said further time scaling is applied prior to forward transforming in the encoder of said coding system.

18. The method of claim 16 wherein said further time scaling is applied subsequent to inverse transformation in the decoder of said coding system.

19. The method of claim 16 wherein the time duration of the signal components added or deleted by said further time scaling is substantially the same as the time duration of signal components deleted or added by said first-recited time scaling, respectively, whereby the time duration of said audio signal stream is substantially unchanged.

20. The method of claim 15 further comprising applying compensating time scaling to the audio signal stream preceding said distortion artifacts, which precede said transient, and subsequent to inverse transformation in the decoder of said coding system such that the time evolution of the processed audio signal stream is

substantially the same as that of the audio signal stream prior to said shifting and the time duration of said audio signal stream is substantially unchanged.

21. The method of claim 20 wherein said coding system includes an encoder and a decoder, said encoder transmitting metadata to said decoder, said metadata including information useful for applying said compensating time scalings.

22. The method of claim 9 wherein said audio signal stream applied to the coding system is a digital signal stream in which the audio information is represented by samples, the order of said samples representing time, and wherein said time scaling has the effect of deleting samples from or adding samples to the digital signal stream applied to the coding system.

23. The method of claim 9 wherein a further time scaling is applied following said signal transient, said further time scaling acting in the opposite sense to the said first-recited time scaling.

24. The method of claim 23 wherein said further time scaling is performed on a segment of said audio stream closely following said transient.

25. The method of claim 24 wherein said time scaling is performed on a segment of said audio stream that is at least partially temporally post-masked by transient.

26. The method of claim 23 wherein said first-recited time scaling has the effect of deleting signal components from or adding signal components to the audio signal stream applied to the coding system and said further time scaling has the effect of adding signal components to the audio signal stream when said first-recited time scaling deletes signal components and said further time scaling has the effect of

deleting signal components to the audio signal stream when said first-recited time scaling adds signal components.

27. The method of claim 26 wherein the time duration of the signal components added or deleted by said further time scaling is substantially the same as the time duration of signal components deleted or added by said first-recited time scaling, respectively, whereby the time duration of said audio signal stream is substantially unchanged.

28. The method of claim 23 wherein said audio signal stream applied to the coding system is a digital signal stream in which the audio information is represented by samples, the order of said samples representing time, and wherein said first-recited time scaling has the effect of deleting samples from or adding samples to the digital signal stream applied to the coding system and said further time scaling has the effect of adding samples to the digital signal stream when said first-recited time sampling deletes samples from the digital signal stream and said further time scaling has the effect of deleting samples from the digital signal stream when said first-recited time sampling adds samples to the digital signal stream.

29. The method of claim 1 wherein said detecting detects multiple transients and said shifting shifts the temporal location of the first of said transients to reduce distortion artifacts prior to the first of said transients.

30. The method of claim 29 wherein the temporal location of the first of said transients with respect to said coding blocks is shifted by time scaling said audio signal stream preceding the first of said signal transients.

31. The method of claim 30 wherein a further time scaling is applied following the first of said transients and before one or more other of said multiple

transients, said further time scaling acting in the opposite sense to the said first-recited time scaling.

32. The method of claim 30 wherein a further time scaling is applied following said transients, said further time scaling acting in the opposite sense to the said first-recited time scaling.

33. In a decoder of a transform-based low-bit-rate audio coding system employing coding blocks, a method for reducing distortion artifacts preceding a signal transient in an audio signal stream subsequent to inverse transformation, comprising

detecting a transient in the audio signal stream, and

time compressing at least a portion of said distortion artifacts such that the time duration of said distortion artifacts is reduced.

34. The method of claim 33 wherein the portion of the distortion artifacts is determined at least in part by the location of the detected transient and a default parameter.

35. The method of claim 33 the portion of the distortion artifacts is determined at least in part by the location of the detected transient and signal characteristics preceding said transient.

36. The method of claim 35 wherein said signal characteristics include a measure of high-frequency components of the audio signal stream.

37. The method of claim 34 or 35 further comprising time expanding prior to said time compression such that the time evolution and length of the audio signal stream is substantially unchanged.

38. The method of claim 34 or 35 further comprising time expanding subsequent to said time compression such that the length of the audio signal stream is substantially unchanged.

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