Description

FIELD OF THE INVENTION

[0001] The invention relates generally to speech coding and, more particularly, to adapting the coding of a speech signal to local characteristics of the speech signal.

BACKGROUND OF THE INVENTION

[0002] Most conventional speech coders apply the same coding method regardless of the local character of the speech segment to be encoded. It is, however, recognized that enhanced quality can be achieved if the coding method is changed, or adapted, according to the local character of the speech. Such adaptive methods are commonly based on some form of classification of a given speech segment, which classification is used to select one of several coding modes (multi-mode coding). Such techniques are especially useful when there is background noise which, in order to obtain a natural sounding reproduction thereof, requires coding approaches that differ from the coding technique generally applied to the speech signal itself.

[0003] One disadvantage associated with the aforementioned classification schemes is that they are somewhat rigid; giving rise to the danger of mis-classifying a given speech segment and, as a result, selecting an improper coding mode for that segment. The improper coding mode typically results in severe degradation in the resulting coded speech signal. The classification approach thus disadvantageously limits the performance of the speech coder.

[0004] A well-known technique in multi-mode coding is to perform a closed-loop mode decision where the coder tries all modes and decides on the best according to some criterion. This alleviates the mis-classification problem to some extent, but it is a problem to find a good criterion for such a scheme. It is, as is also the case for aforementioned classification schemes, necessary to transmit information (i.e., send overhead bits from the transmitter's encoder through the communication channel to the receiver's decoder) describing which mode is chosen. This restricts the number of coding modes in practice.

[0005] It is therefore desirable to permit a speech coding (encoding or decoding) procedure to be changed or adapted based on the local character of the speech without the severe degradations associated with the aforementioned conventional classification approaches and without requiring transmission of overhead bits to describe the selected adaptation.

[0006] According to the present invention as claimed in claims 1-54, a speech coding (encoding or decoding) procedure can be adapted without rigid classifications and the attendant risk of severe degradation of the coded speech signal, and without requiring transmission of overhead bits to describe the selected adaptation. The adaptation is based on parameters already existing in the coder (encoder or decoder) and therefore no extra information has to be transmitted to describe the adaptation. This makes possible a completely soft adaptation scheme where an infinite number of modifications of the coding (encoding or decoding) method is possible. Furthermore, the adaptation is based on the coder's characterization of the signal and the adaptation is made according to how well the basic coding approach works for a certain speech segment.

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BRIEF DESCRIPTION OF THE DRAWINGS

[0007]

FIGURE 1 is a block diagram which illustrates generally a softly adaptive speech encoding scheme according to the invention.

FIGURE 1A illustrates the arrangement of FIGURE 1 in greater detail.

FIGURE 2 illustrates in greater detail the arrangement of FIGURE 1A.

FIGURE 3 illustrates the multi-level code modifier of FIGURES 2 and 21 in more detail.

FIGURE 4 illustrates one example of the softly adaptive controller of FIGURES 2 and 21.

FIGURE 5 is a flow diagram which illustrates the operation of the softly adaptive controller of FIGURE 4.

FIGURE 6 illustrates diagrammatically an anti-sparseness filter according to the invention which may be provided as one of the modifier levels in the multi-level code modifier of FIGURE 3.

FIGURES 7-11 illustrate graphically the operation of an anti-sparseness filter of the type illustrated in FIGURE 6.

FIGURE 12-16 illustrate graphically the operation of an anti-sparseness filter of the type illustrated in FIGURE 6 and at a relatively lower level of anti-spareness operation than the anti-spareness filter of FIGURES 7-11.

FIGURE 17 illustrates a pertinent portion of another speech coding arrangement according to the invention.

FIGURE 18 illustrates a pertinent portion of a further speech coding arrangement according to the invention.

FIGURE 19 illustrates a modification applicable to the speech coding arrangements of FIGURES 2, 17 and 21.

FIGURE 20 is a block diagram which illustrates generally a softly adaptive speech decoding scheme according to the invention.

FIGURE 20A illustrates the arrangement of FIGURE 20 in greater detail.

FIGURE 21 illustrates in greater the detail the arrangement of FIGURE 20A.

data processor in combination with additional external circuitry connected thereto.

[0040] Although exemplary embodiments of the present invention have been described above in detail, this does not limit the scope of the invention as defined by the appended claims, which can be practiced in a variety of embodiments.

Claims:

1. A speech encoding apparatus for producing a coded representation of an original speech signal, comprising:

   an input for receiving the original speech signal;
   an output for providing said coded representation of said original speech signal;
   a coder (11) coupled between said input and said output for selectively performing on the original speech signal either a coding operation or an adaptation of said coding operation to produce said coded representation; and characterised by
   a controller (16) coupled to said coder to receive therefrom and store information currently being used by said coder in said coding operation, said controller including an output coupled to said coder and responsive to said information currently being used by said coder in said coding operation and to previous information previously used by said coder in said coding operation and stored by said controller for signaling said coder to perform said adaptation of said coding operation. Add Rider A.

2. The apparatus of Claim 1, wherein said information currently being used in said coding operation includes voicing information indicative of a voicing level of said original speech signal.

3. The apparatus of Claim 2, wherein said coding operation and said adaptation thereof include adaptive gainshape coding, and wherein said voicing information includes a gain signal associated with said adaptive gainshape coding.

4. The apparatus of Claim 2, wherein said controller includes a memory for maintaining a record of previous voicing levels as indicated by said voicing information, and refining logic operable when said voicing information indicates that a current voicing level exceeds a predetermined threshold to evaluate said current voicing level with respect to said previous voicing levels to determine whether said voicing information indicative of said current voicing level should be used by said controller.

5. The apparatus of Claim 1, wherein said information currently being used in said coding operation includes signal energy information indicative of a signal energy in the original speech signal.

6. The apparatus of Claim 5, wherein said coding operation and said adaptation thereof include fixed gainshape coding, and wherein said signal energy information includes a gain signal associated with said fixed gainshape coding.

7. The apparatus of Claim 5, wherein said information currently being used in said coding operation includes voicing information indicative of a voicing level of said original speech signal.

8. The apparatus of Claim 7, wherein said controller includes a memory for maintaining a record of a previous signal energy as indicated by said signal energy information, and refining logic operable when said voicing information indicates that a current voicing level exceeds a predetermined threshold to evaluate a current signal energy with respect to said previous signal energy to determine whether said voicing information indicative of said current voicing level should be used by said controller.

9. The apparatus of Claim 1, wherein said coding operation and said adaptation thereof include linear predictive coding.

10. The apparatus of Claim 1, wherein said coder is operable to perform any selected one of a plurality of different adaptations of said coding operation in response to said controller output, and wherein said controller includes map logic having an input to receive said information currently being used in said coding operation and having an output that indicates which of said adaptations should be signaled to said coder.

11. The apparatus of Claim 10, wherein said controller includes further logic coupled to said map logic output for determining whether the adaptation indicated by said map logic output differs by more than a threshold amount from said coding operation.

12. ~~The apparatus of Claim 1~~, wherein said coder includes an algebraic codebook and said performance of said adaptation includes performing anti-sparseness filtering on a signal received ~~from~~ said algebraic codebook. Rider A from

13. A speech encoding method for producing a coded representation of an original speech signal, comprising the steps of 2,

   receiving the original speech signal;
   performing on the original speech signal a

current coding operation to produce the coded representation; and characterised by the steps of

responsive to information currently being used in the current coding operation and information used previously in the current coding operation, adapting the current coding operation to produce an adapted coding operation; and

performing the adapted coding operation on the original speech signal. Add Rider B.

14. The method of Claim 13, wherein the information currently being used in the current coding operation includes voicing information indicative of a voicing level of the original speech signal.

15. The method of Claim 14, wherein said performing steps include performing adaptive gainshape coding, and wherein said voicing information includes a gain signal associated with the adaptive gainshape coding.

16. The method of Claim 14, including maintaining a record of previous voicing levels as indicated by said voicing information and, if said voicing information indicates that a current voicing level exceeds a predetermined threshold, evaluating the current voicing level with respect to the previous voicing levels.

17. The method of Claim 16, including modifying the voicing information indicative of the current voicing level to indicate a different voicing level.

18. The method of Claim 17, wherein said different voicing level is a lower voicing level.

19. The method of Claim 13, wherein the information currently being used in the current coding operation includes signal energy information indicative of a signal energy in the original speech signal.

20. The method of Claim 19, wherein said performing steps include performing fixed gainshape coding, and wherein the signal energy information includes a gain signal associated with the fixed gainshape coding.

21. The method of Claim 19, wherein the information currently being used in the current coding operation includes voicing information indicative of a voicing level of the original speech signal.

22. The method of Claim 21, including maintaining a record of a previous signal energy as indicated by the signal energy information and, if the voicing information indicates that a current voicing level exceeds a predetermined threshold, evaluating a current signal energy with respect to the previous signal energy to determine whether the current voicing level should be accepted.

23. The method of Claim 13, wherein said performing steps include performing linear predicative coding.

24. The method of Claim 13, wherein said adapting step includes adapting the current coding operation to produce any selected one of a plurality of different adaptations of the current coding operation.

25. The method of Claim 24, wherein said adapting step includes selecting, in response to the information currently being used in the current coding operation, one of said adaptations to be produced in said adapting step, and thereafter determining a difference between the selected adaptation and the current coding operation.

26. The method of Claim 25, wherein said adapting step includes, if the selected adaptation differs from the current coding operation by more than a threshold amount, selecting another adaptation which differs less from the current coding operation.

~~27. The method of Claim 16,~~ wherein said last-mentioned performing step includes performing anti-sparseness filtering on a signal received from an algebraic codebook. Rider B.

~~28.~~ 3. A speech decoding apparatus for producing a decoded speech signal from a coded representation of an original speech signal, comprising:

an input for receiving the coded representation of the original speech signal;

an output for providing said decoded speech signal;

a decoder (200) coupled between said input and said output for selectively performing on said coded representation either a decoding operation or an adaptation of said decoding operation to produce said decoded speech signal; and characterised by

a controller (19) coupled to said decoder to receive therefrom and store information currently being used by said decoder in said decoding operation, said controller including an output coupled to said decoder and responsive to said information currently being used by said decoder in said decoding operation and to previous information used previously by said decoder in said decoding operation and previously stored by said controller for signaling said decoder to perform said adaptation of said decoding operation. Add Rider C.

~~29. The apparatus of Claim 28, wherein said inferma-~~

tion currently being used in said decoding operation includes voicing information indicative of a voicing level of said original speech signal.

30. The apparatus of Claim 29, wherein said decoding operation and said adaptation thereof include adaptive gainshape coding, and wherein said voicing information includes a gain signal associated with said adaptive gainshape coding.

31. The apparatus of Claim 29, wherein said controller includes a memory for maintaining a record of previous voicing levels as indicated by said voicing information, and refining logic operable when said voicing information indicates that a current voicing level exceeds a predetermined threshold to evaluate said current voicing level with respect to said previous voicing levels to determine whether said voicing information indicative of said current voicing level should be used by said controller.

32. The apparatus of Claim 28, wherein said information currently being used in said decoding operation includes signal energy information indicative of a signal energy in the original speech signal.

33. The apparatus of Claim 32, wherein said decoding operation and said adaptation thereof include fixed gainshape coding, and wherein said signal energy information includes a gain signal associated with said fixed gainshape coding.

34. The apparatus of Claim 32, wherein said information currently being used in said decoding operation includes voicing information indicative of a voicing level of said original speech signal.

35. The apparatus of Claim 34, wherein said controller includes a memory for maintaining a record of a previous signal energy as indicated by said signal energy information, and refining logic operable when said voicing information indicates that a current voicing level exceeds a predetermined threshold to evaluate a current signal energy with respect to said previous signal energy to determine whether said voicing information indicative of said current voicing level should be used by said controller.

36. The apparatus of Claim 28, wherein said decoding operation and said adaptation thereof include linear predictive coding.

37. The apparatus of Claim 28, wherein said decoder is operable to perform any selected one of a plurality of different adaptations of said decoding operation in response to said controller output, and wherein said controller includes map logic having an input to receive said information currently being used in said decoding operation and having an output that indicates which of said adaptations should be signaled to said decoder.

38. The apparatus of Claim 37, wherein said controller includes further logic coupled to said map logic output for determining whether the adaptation indicated by said map logic output differs by more than a threshold amount from said decoding operation.

~~39. The apparatus of Claim 28,~~ wherein said decoder includes an algebraic codebook and said performance of said adaptation includes performing anti-sparseness filtering on a signal received from said algebraic codebook. Rider C

Add Rider D.

40. A speech decoding method for producing a decoded speech signal from a coded representation of an original speech signal, comprising the steps of:

receiving the coded representation of the original speech signal;

performing on the coded representation a current decoding operation to produce the decoded speech signal; and characterised by the steps of

responsive to information currently being used in the current decoding operation and to information previously used in the current decoding operation, adapting the current decoding operation to produce an adapted decoding operation; and

performing the adapted decoding operation on the coded representation. Add Rider E.

41. The method of Claim 40, wherein the information currently being used in the current decoding operation includes voicing information indicative of a voicing level of the original speech signal.

42. The method of Claim 41, wherein said performing steps include performing adaptive gainshape coding, and wherein said voicing information includes a gain signal associated with the adaptive gainshape coding.

43. The method of Claim 41, including maintaining a record of previous voicing levels as indicated by said voicing information and, if said voicing information indicates that a current voicing level exceeds a predetermined threshold, evaluating the current voicing level with respect to the previous voicing levels.

44. The method of Claim 43, including modifying the voicing information indicative of the current voicing level to indicate a different voicing level.

45. The method of Claim 44, wherein said different voicing level is a lower voicing level.

46. The method of Claim 40, wherein the information currently being used in the current decoding operation includes signal energy information indicative of a signal energy in the original speech signal.

47. The method of Claim 46, wherein said performing steps include performing fixed gainshape coding, and wherein the signal energy information includes a gain signal associated with the fixed gainshape coding.

48. The method of Claim 46, wherein the information currently being used in the current decoding operation includes voicing information indicative of a voicing level of the original speech signal.

49. The method of Claim 48, including maintaining a record of a previous signal energy as indicated by the signal energy information and, if the voicing information indicates that a current voicing level exceeds a predetermined threshold, evaluating a current signal energy with respect to the previous signal energy to determine whether the current voicing level should be accepted.

50. The method of Claim 40, wherein said performing steps include performing linear predicative coding.

51. The method of Claim 40, wherein said adapting step includes adapting the current decoding operation to produce any selected one of a plurality of different adaptations of the current decoding operation.

52. The method of Claim 51, wherein said adapting step includes selecting, in response to the information currently being used in the current decoding operation, one of said adaptations to be produced in said adapting step, and thereafter determining a difference between the selected adaptation and the current decoding operation.

53. The method of Claim 52, wherein said adapting step includes, if the selected adaptation differs from the current decoding operation by more than a threshold amount, selecting another adaptation which differs less from the current decoding operation.

~~54. The method of Claim 40,~~ wherein said last-mentioned performing step includes performing anti-sparseness filtering on a signal received from an algebraic codebook. Rider E.

Add Rider F.

13. ~~55.~~ Wireless speech communication device adapted for executing the speech decoding method in accordance with any of the claims ~~40–54~~. 8 to 12.

14. ~~56.~~ Wireless speech communication device adapted for executing the speech coding method in accordance with any ~~of the~~ claims ~~1–39~~. 2.

15. ~~57.~~ Wireless speech communication device comprising the speech decoding apparatus in accordance with any of the claims ~~38–39~~. 3 to 7.

16. ~~58.~~ Wireless speech communication device comprising the speech coding apparatus in accordance with ~~any of the~~ claims ~~1–37~~.

Patentansprüche

1. Eine Sprachcodiervorrichtung zum Erzeugen einer codierten Darstellung eines ursprünglichen Sprachsignals, umfassend:

einen Eingang zum Empfangen des ursprünglichen Sprachsignals;

einen Ausgang zum Bereitstellen der codierten Darstellung des ursprünglichen Sprachsignals;

einen Codierer (11), angeschlossen zwischen dem Eingang und dem Ausgang, zum selektiven Durchführen entweder einer Codierungsoperation mit dem ursprünglichen Sprachsignal oder einer Adaptierung der Codierungsoperation, um die codierte Darstellung zu erzeugen; und

gekennzeichnet durch

einen Controller (19) der mit dem Codierer gekoppelt ist, um momentan durch den Codierer bei der Codierungsoperation verwendete Information von dort zu empfangen und zu speichern, wobei der Controller einen mit dem Codierer gekoppelten Ausgang umfasst und auf die momentan durch den Codierer bei der Codierungsoperation verwendete Information anspricht, und auf vorhergehende Information, die durch den Codierer bei der Codierungsoperation vorhergehend verwendet wurde, und durch den Controller gespeichert wurde, um dem Codierer zu signalisieren, eine Adaption der Codierungsoperation durchzuführen.

2. Die Vorrichtung nach Anspruch 1, wobei die momentan bei der Codierungsoperation verwendete Information eine Stimminformation umfasst, die einen Stimmpegel des ursprünglichen Sprachsignals anzeigt.

3. Die Vorrichtung nach Anspruch 2, wobei die Codierungsoperation und die Adaption davon ein adapti-

RIDER D

New Claim 4: *The apparatus of Claim 3 wherein the anti-sparseness filtering includes a convolver that performs a circular convolution of the coded speech estimate received from the said algebraic codebook with an impulse response associated with an all pass filter.*

New Claim 5: *The apparatus of Claim 4 wherein the entries of said algebraic codebook have only two non-zero samples out of a total of forty samples.*

New Claim 6: *The apparatus of Claims 4 or 5 wherein the controller selects the said impulse response to use in said circular convolution from a plurality of impulse responses.*

New Claim 7: *The apparatus of Claim 6 wherein the controller uses said information to determine which of said impulse responses to use in said circular convolution.*

RIDER F

New Claim 9: *The method of Claim 8 wherein the anti-sparseness filtering includes a convolver that performs a circular convolution of the coded speech estimate received from the said algebraic codebook with an impulse response associated with an all pass filter.*

New Claim 10: *The method of Claim 9 wherein the entries of said algebraic codebook have only two non-zero samples out of a total of forty samples.*

New Claim 11: *The method of Claims 9 or 10 wherein the controller selects the said impulse response to use in said circular convolution from a plurality of impulse responses.*

New Claim 12: *The method of Claim 11 wherein the controller uses said information to determine which of said impulse responses to use in said circular convolution.*