(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 1 334 485 B1

(12) EUROPEAN PATENT SPECIFICATION

(45) Date of publication and mention of the grant of the patent:
31.08.2005 Bulletin 2005/35

(21) Application number: **01993000.7**

(22) Date of filing: **22.10.2001**

(51) Int Cl.[7]: **G10L 19/06**

(86) International application number:
PCT/EP2001/012403

(87) International publication number:
WO 2002/037477 (10.05.2002 Gazette 2002/19)

(54) **SPEECH CODEC AND METHOD FOR GENERATING A VECTOR CODEBOOK AND ENCODING/DECODING SPEECH SIGNALS**

SPRACH-CODEC UND VERFAHREN ZUR ERZEUGUNG EINES VEKTORCODEBUCHS UND ZUM CODIEREN/DECODIEREN VON SPRACHSIGNALEN

CODEC VOCAL ET PROCEDE DE GENERATION D'UN CODE VECTORIEL ET DE CODAGE/DECODAGE DE SIGNAUX VOCAUX

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE TR

(30) Priority: **30.10.2000 GB 0026463**

(43) Date of publication of application:
13.08.2003 Bulletin 2003/33

(73) Proprietor: **MOTOROLA, INC.**
Schaumburg, IL 60196 (US)

(72) Inventors:
- **GIBBS, Jonathan Alastair**
  Southampton, Hampshire SO30 2XF (GB)
- **HOSKIN, James Malcolm**
  Chessington, Surrey KT9 1BP (GB)

(74) Representative: **McCormack, Derek James**
Optimus
Grove House, Lutyens Close,
Chineham Court,
Basingstoke,
Hampshire RG24 8AG (GB)

(56) References cited:
EP-A- 0 911 807
US-A- 5 621 852
US-A- 5 966 688

- **GIBBS J A ET AL: "LSF split vector quantization for wideband codecs with narrowband tandemming" 2000 IEEE WORKSHOP ON SPEECH CODING. PROCEEDINGS. MEETING THE CHALLENGES OF THE NEW MILLENNIUM (CAT. NO.00EX421), DELAVAN, WI, USA,, 17 - 20 September 2000, pages 120-122, XP002194619 2000, Piscataway, NJ, USA, IEEE, USA ISBN: 0-7803-6416-3**
- **RAMAKRISHNAN S ET AL: "Constrained-storage vector quantization with a universal codebook" IEEE TRANSACTIONS ON IMAGE PROCESSING, JUNE 1998, IEEE, USA, vol. 7, no. 6, pages 785-793, XP002194620 ISSN: 1057-7149**

EP 1 334 485 B1

Printed by Jouve, 75001 PARIS (FR)

Description

Field of the Invention

[0001] This invention relates to speech coding and methods of optimising the performance of speech codecs in communications systems. The invention is applicable to, but not limited to, speech codecs that accommodate wideband and narrowband speech signals without compromising the overall performance of the speech codec quantiser.

Background of the Invention

[0002] Many present day voice communications systems, such as the global system for mobile communications (GSM) cellular telephony standard and the TErrestrial Trunked RAdio (TETRA) system for private mobile radio users, use speech processing units to encode and decode speech patterns. In such voice communications systems a speech encoder converts the analogue speech pattern into a suitable digital format for transmission and a speech decoder converts a received digital speech signal into an audible analog speech pattern.

[0003] As frequency spectrum for such voice communications systems is a valuable resource, it is desirable to limit the channel bandwidth used by such speech signals, in order to maximise the number of users per frequency band. Hence, a primary objective in the use of speech coding techniques is to reduce the occupied capacity of the speech patterns as much as possible, by use of compression techniques, without losing fidelity.

[0004] A popular solution in speech coding technology is the application of vector quantisation (VQ). A prime incentive in using VQ can be found in Shannon's rate distortion theory, as known to those skilled in the art, which states that better performance can always be achieved by coding vectors instead of scalars.

[0005] The process of vector quantisation is to represent an input vector as a member of a set of fixed vectors.. This set of fixed vectors is known as the VQ codebook. The fixed vector in the VQ codebook which best represents the input vector is found by exhaustively searching all members of the VQ codebook and selecting the fixed vector which gives the minimum distance measure (or Euclidean distance) between it and the input vector.

[0006] This procedure requires that every fixed vector in the VQ codebook is searched in order to find the best representation of the input vector. Consequently searching a full VQ codebook is computationally expensive and memory hungry. Although VQ has been shown to be very attractive and efficient in many areas of speech coding, it is not without its drawbacks.

[0007] In the field of this invention it is known that wideband speech codecs are likely to find application in telephone conferencing. Wideband speech codecs have an input speech bandwidth covering the 50Hz to 7KHz range, compared to narrowband or telephone-band codecs that have an input speech bandwidth of 250Hz to 3.3KHz.

[0008] The consequence of this is that wideband speech codecs will often be tandemmed with narrowband speech codecs. Tandemming is a term which is used to describe the situation where speech previously processed by one speech encoder/decoder is processed by a second speech encoder/decoder pair.

[0009] Furthermore, in these situations, the speech quality requirement of such tandemmed codecs is to achieve equivalence to the best narrowband codecs i.e. GSM Enhanced full-rate codec (EFR). It is therefore appropriate to consider the performance of any wideband line spectral frequency (LSF) VQ scheme in the presence of narrowband speech.

[0010] To accommodate the processing of both wideband and narrowband speech signals in a speech synthesis unit, one option may be to use a classified VQ scheme with two sets of codebooks: one to represent the wideband speech and one to represent the narrowband speech. In this case, a respective codebook would be selected by a special "mode" bit, where the mode bit indicates whether the subsequent data bits represent a wideband or narrowband speech signal.

[0011] One problem that is inherent with this technique is that two individual codebooks are required, one arranged for wideband performance and one arranged for narrowband performance. Subsequent bit errors applied to the mode bit would result in large de-quantiser errors. To resolve any such mode bit error problems, heavy forward error correction (FEC) protection is required in the resulting speech codec.

[0012] One disadvantage associated with quantising wideband LSFs, when there is a possibility that narrowband codec tandemming may be present, is highlighted below in Table 1.

[0013] The representative codecs have been simulated with each predictor of the speech codec arranged to be an 18^{th} order split vector quantiser, with the eighteen associated line spectral frequencies split into six groups of three bits each.

Table 1:

7KHz & 3KHz Spectral Distortion Results for the 1st order MA-PVQ 40 bit Quantisers.						
Quantiser Configuration	Wideband Performance			Narrowband Performance		
	Mean SD (dB)	% Frames 3-5 dB	% Frames > 5 dB	Mean SD (dB)	% Frames 2-4 dB	% Frames > 4 dB
6,8,7,7,6,6	1.418	0.887	0.019	1.292	5.754	0.058
6,7,7,7,7,6	1.417	1.012	0.010	1.384	8.047	0.087
7,7,7,7,6,6	1.418	0.916	0.019	1.301	6.062	0.077
7,8,7,7,6,5	1.389	0.771	0.019	1.202	4.163	0.058
6,8,8,7,6,5	1.396	0.684	0.019	1.231	4.597	0.029
7,7,7,7,7,5	1.387	0.810	0.010	1.300	6.081	0.077
7,7,8,7,6,5	1.397	0.723	0.019	1.240	4.857	0.058
7,8,8,7,6,4	1.379	0.646	0.019	1.137	3.094	0.039
7,9,9,6,5,4	1.453	0.896	0.029	1.033	1.744	0.000
7,9,9,7,5,3	1.431	0.752	0.000	1.017	1.677	0.010
8,8,8,7,6,3	1.390	0.723	0.010	1.073	2.274	0.029
8,9,8,7,6,2	1.419	0.964	0.010	0.993	1.484	0.019
8,8,8,8,5,3	1.419	0.867	0.019	1.067	2.101	0.029
8,9,8,7,5,3	1.428	0.848	0.000	0.995	1.503	0.029
8,9,8,8,5,2	1.447	1.224	0.010	0.986	1.378	0.019
8,9,9,7,5,2	1.459	1.245	0.010	0.947	1.051	0.010

[0014] The wideband and narrowband distortion figures are presented in the form suggested by; "LSF Quantization in Wideband Speech Coders", Proceedings of 1999 IEEE Workshop on Speech Coding, pp 25-27, M. Ferhaoui & S. Van Gerven and "Efficient Vector Quantization of LPC Parameters at 24 Bits/Frame", IEEE Transactions on Speech and Audio Processing, Vol 1, No 1, March 1993, K.K. Paliwal & B.S. Atal.

[0015] Table 1 details the wideband and narrowband spectral distortion figures for a 40-bit first order moving average quantiser trained on 50:50 wideband:narrowband speech. As mentioned, the configuration column denotes the number of bits allocated to each of six, moving-average predictive split-vector quantisers, applied to LSFs 1-3, 4-6, 7-9, 10-12, 13-15 and 16-18 respectively.

[0016] It is worth noting that a wideband speech codec would typically be represented by an even distribution of bits allocated to each of the six split-vector quantisers, to provide an approximately even frequency response across the full range of the line spectral frequencies. In contrast, a narrowband speech codec would have an uneven distribution of bits associated with each quantiser, with more bits allocated to the lower frequencies of the LSFs.

[0017] It is clear that the best wideband performance is obtained by a first configuration (7,8,8,7,6,4), whilst the best narrowband performance is obtained with a different second configuration (8,9,9,7,5,2). Both configurations consist of the same total number of bits, but with different distributions of the numbers of bits per quantiser.

[0018] A compromise quantiser such as 8,9,8,7,6,2 provides substantially inferior performance to both of these. In addition, it is noteworthy that this dual quantiser approach would require a total of:

$$3 \text{ x } (640 + 1444) \text{ words} = 6252 \text{ words in memory.}$$

[0019] Hence, a need exists to provide a bit-error robust speech codec, preferably complementary to popular, proven speech codecs such as embedded split VQ codecs, that can provide improved quantisation of wideband line spectral frequencies (lsfs) for wideband speech and also in narrowband tandemming.

[0020] The present invention aims to provide a speech codec and method of optimising a performance of the speech codec to at least alleviate some of the aforementioned disadvantages.

[0021] Published prior art documents known to the applicant include WO99/65017, WO97/27578, WO95/10760 and EP-A-411655.

Summary of the Invention

[0022] In a first aspect of the present invention, a speech coder for a speech communications unit in accordance with claim 1 is provided.

[0023] In a second aspect of the present invention, a speech communications unit adapted to include the speech coder of any one of claims 1 to 10 is provided.

[0024] In a third aspect of the present invention, a method of generating a speech vector codebook in a speech communications unit in accordance with claim 12 is provided.

[0025] In a fourth aspect of the present invention, a speech communications unit adapted to include a speech vector codebook generated, in accordance with any one of claims 12 to 20, is provided.

[0026] In a fifth aspect of the present invention, a method of encoding a speech signal, in accordance with claim 22, is provided.

[0027] In a sixth aspect of the present invention, a speech communications unit adapted to employ a speech encoding method in accordance with any one of claims 22 to 26 is provided.

[0028] In a seventh aspect of the present invention, a method of decoding a speech signal, in accordance with claim 28, is provided.

[0029] In an eighth aspect of the present invention, a speech communications unit adapted to employ a speech decoding method in accordance with any one of claims 28 to 31 is provided.

Brief Description of the Drawings

[0030] Exemplary embodiments of the present invention will now be described, with reference to the accompanying drawings, in which:

FIG. 1 shows a block diagram of a code excited linear predictive speech encoder that can be adapted to support the various inventive concepts of a preferred embodiment of the present invention;

FIG. 2 shows a block diagram of a code excited linear predictive speech decoder that can be adapted to support the various inventive concepts of a preferred embodiment of the present invention;

FIG. 3 shows a 2-way split VQ codebook applied to eighteen wideband line spectral frequencies (LSFs) adapted to support the inventive concepts of the preferred embodiments of the present invention;

FIG. 4 shows a preferred packing arrangement of bits for the 2-way split VQ codebook of FIG. 3, adapted to support the inventive concepts of the preferred embodiments of the present invention;

FIG. 5 shows an octagon partitioned to reflect eight separate locations for the 2-way split VQ codebook of FIG. 3, in accordance with a preferred embodiment of the present invention;

FIG. 6 shows a graph that demonstrates the error resilience of a preferred embodiment of the present invention in the presence of 10% bit errors applied on a per-bit basis with natural ordering; and

FIG. 7 shows a further graph that demonstrates the error resilience of a preferred embodiment of the present invention in the presence of 10% bit errors applied on a per-bit basis with ranked ordering.

Description of Preferred Embodiments

[0031] Referring first to FIG. 1, a block diagram of a code excited linear predictive speech encoder 100, according to a preferred embodiment of the present invention, is shown. An acoustic input signal to be analysed is applied to speech coder 100 at microphone 102. The input signal is then applied to filter 104. Filter 104 will generally exhibit band-pass filter characteristics. However, if the speech bandwidth is already adequate, filter 104 may comprise a direct wire connection.

[0032] The analog speech signal from filter 104 is then converted into a sequence of N pulse samples, and the amplitude of each pulse sample is then represented by a digital code in analog-to-digital (A/D) converter 108, as known in the art. The sampling rate is determined by sample clock (SC). The sample clock SC is generated along with the frame clock (FC) via clock 112.

[0033] The digital output of A/D 108, which may be represented as input speech vector s(n), is then applied to coefficient analyser 110. This input speech vector s(n) is repetitively obtained in separate frames, i.e., blocks of time, the length of which is determined by the frame clock (FC), as is known in the art.

[0034] For each block of speech, a set of linear predictive coding (LPC) parameters is produced in accordance with a preferred embodiment of the invention by coefficient analyser 110. The generated speech coder parameters may include the following: LPC parameters, long-term predictor (LTP) parameters, excitation gain factor (√) (along with the best excitation codeword I). Such speech coding parameters are applied to multiplexer 150 and sent over the channel 152 for use by the speech synthesizer at the decoder. The input speech vector s(n) is also applied to subtractor 130, the function of which is described later.

[0035] In the preferred embodiment of the invention, a specially constructed family of embedded codebooks is used in order to best represent the LPC parameters of the input speech signal. Hence, coefficient analyser 110 has been adapted to incorporate the specially constructed family of embedded codebooks.

[0036] It is within the contemplation of the invention that any number of embedded codebooks would benefit from the inventive concepts described herein. The inventive concepts of the embedded codebook arrangement are further described with reference to FIG. 3.

[0037] Within the conventional CELP encoder of FIG. 1, the codebook search controller 140 selects the best indices and gains from the adaptive codebook within block 116 and the stochastic codebook within block 114 in order to produce a minimum weighted error in the summed chosen excitation vector used to represent the input speech sample. The output of the stochastic codebook 114 and the adaptive codebook 116 are input into respective gain functions 122 and 118. The gain-adjusted outputs are then summed in summer 120 and input into the LPC filter 124, as is known in the art.

[0038] For each individual excitation vector $u_i(n)$, a reconstructed speech vector $s'_i(n)$ is generated for comparison to the input speech vector s(n). Gain block 122 scales the excitation gain factor '√'. Such gain may be pre-computed by coefficient analyser 110 and used to analyse all excitation vectors, or may be optimised jointly with the search for the best excitation codeword I, generated by codebook search controller 140.

[0039] The scaled excitation signal $\sqrt{}u_i$ (n) is then filtered by the linear predictive coding filter 124, which preferably includes a long-term predictor (LTP) filter and a short-term predictor (STP) filter, to generate the reconstructed speech vector $s'_i(n)$.

The reconstructed speech vector $s'_i(n)$ for the i-th excitation code vector is compared to the same block of input speech vector s(n) by subtracting these two signals in subtractor 130.

[0040] The difference vector e_i (n) represents the difference between the original and the reconstructed blocks of speech. The difference vector is perceptually-weighted by weighting filter 132, utilising the weighting filter parameters (WTP) generated by coefficient analyser 110. Perceptual weighting accentuates those frequencies where the error is perceptually more important to the human ear, and attenuates other frequencies.

[0041] An energy calculator function inside the codebook search controller 140 computes the energy of the weighted difference vector $e'_i(n)$. The codebook search controller compares the i-th error signal for the present excitation vector $u_i(n)$ against previous error signals to determine the excitation vector producing the minimum error. The code of the i-th excitation vector having a minimum error is then output over the channel as the best excitation code I.

[0042] In the alternative, codebook search controller 140 may determine a particular codeword that provides an error signal having some predetermined criteria, such as meeting a predefined error threshold.

[0043] A more detailed description of the functionality of a typical speech encoding unit can be found in "Digital speech coding for low-bit rate communications systems" by A. M. Kondoz, published by John Wiley in 1994.

[0044] As mentioned earlier, the coefficient analyzer 110 has also been adapted to employ at least some of the inventive concepts of the present invention. To accommodate vectors in either the wideband or narrowband vector space, the coefficient analyser 110 is used to train the quantisers and to determine whether the input speech comprises wideband or narrowband speech.

[0045] The inventors of the present invention have recognised the opportunity to use the same training data, or at least very similar data, to train each of the quantisers. The different sized quantisers cover much the same signal vector space and hence a smaller quantiser is embedded within the larger quantiser leading to a more compact representation. In the present case, the storage may be reduced from the dual quantiser approach requiring a total of 3 x (640 + 1444) words = 6252 words to 3 x 1600 = 4800 words.

[0046] In the preferred embodiment of the invention, it is desirable for the coefficient analyser 110 to send an additional mode bit to indicate which quantiser set (i.e. which of the specially constructed family of embedded codebooks) is being used. The quantiser set will preferably refer to a wideband or narrowband arrangement. The codebook index transmission is structured in order to minimise the effect of errors to this mode bit, as described later with respect to FIG. 4. The consequence of such a careful structuring of the codebook index transmission means that any bit error(s) in the mode bit have much less impact than in any two or more independent-codebook prior art approach.

[0047] Referring now to FIG. 2, a block diagram of a code excited linear predictive speech decoder 200 is shown, according to a preferred embodiment of the present invention. The decoder functionality is substantially the reverse of that of the encoder.

[0048] The received multiplexed signal is input into demultiplexer 202, which separates the excitation parameters 204 from the LPC parameters 206.

[0049] For each block of speech, a set of linear predictive coding (LPC) parameters were produced in the encoder by coefficient analyser 110 as described with reference to FIG. 1.

[0050] In the preferred embodiment of the present invention, the LPC parameters are input into an LPC de-quantiser, stability check and correction block 210 to obtain a local stable version of the synthesis filter even in the presence of channel bit errors. The LPC de-quantise, stability check and correction block 210 has been adapted to encompass the inventive concepts contained herein.

[0051] The LPC de-quantise, stability check and correction block 210 receives the LPC parameters and mode bit sent from the corresponding encoder function. The LPC de-quantise function of block 210 includes the corresponding embedded codebook arrangement of the encoder, such that the determination of the at least one mode bit can select the embedded codebook arrangement that best describes the encoded and transmitted speech signal. The LPC de-quantise, stability check and correction block 210 also controls the filter co-efficients of the LPC synthesis filter 222 in order to reconstruct the transmitted speech vector $s'_i(n)$. The output from the LPC synthesis filter 222 is input to a post filter process 224, which subsequently outputs the reconstructed speech 226.

[0052] The excitation parameters 204 may include: excitation gain factor √ together with the best excitation codeword I, and are input into an adaptive non-linear smoothing function 208. The output from the adaptive non-linear smoothing function 208 provides the precise adaptive and stochastic codebook indices and gains that form the excitation for the synthesis filter. As such, the outputs from the adaptive non-linear smoothing function 208 are input to stochastic codebook 218 and adaptive codebook 212.

[0053] The gain controls are input to adaptive codebook gain block 214, which receives an output from the adaptive codebook 212, and stochastic codebook gain block 220, which receives an output from the stochastic codebook 218. The output from the respective gain blocks 214, 220 are input to summing junction 216, whose output is fed into the LPC synthesis filter 222 and fed back to the adaptive codebook 212, as known in the art.

[0054] It is again within the contemplation of the invention that any number of embedded codebooks would benefit from the inventive concepts described herein. The inventive concepts of the embedded codebook arrangement, applicable to either the encoder or decoder function are further described with reference to FIG. 3.

[0055] A more detailed description of the functionality of a typical speech decoding unit can also be found in "Digital speech coding for low-bit rate communications systems" by A. M. Kondoz, published by John Wiley in 1994.

[0056] The inventive concepts of the preferred embodiment of the invention are best illustrated with an example. FIG. 3 shows a 2-way split VQ codebook applied to eighteen wideband line spectral frequencies (LSFs). The number of split VQ codebooks together with the number of LSFs have only been selected to more easily show the benefits of the inventive concepts of the present invention.

[0057] The input LSFs (L1-L18) 250 are quantised by first quantiser 254 and second quantiser 268 to derive estimates and the two binary indices "I1" 270 and "I2" 272 using a respective first embedded codebook 256 and second embedded codebook 262. The LSFs (L1-L18) 250 are fed into a mode-bit detector 252, that selects the respective embedded codebook to provide the most appropriate one for the speech signal presented.

[0058] The first embedded codebook 256 contains a first set of core entries, in this case appropriate for wideband speech 260, and additional entries appropriate for narrowband speech 258. The second embedded codebook 262 contains a second set of core entries, this time appropriate for narrowband speech 266, and additional wideband entries 264.

[0059] In each case, the core entries are always searched in each quantiser and are indexed by a set of core bits. Additional entries are searched, depending upon the mode, and a set of "extra" bits are formed. The codebook is structured such that when the full codebook is searched, the "extra" bits are effectively zero for the core entries. This is depicted in FIG. 4.

[0060] This arrangement of core bits provides for a constant sum of the bit allocations for each of the two modes, wideband or narrowband.

[0061] FIG. 4 shows the preferred packing of bits for the 2-way split VQ of FIG. 3. The configuration of the bit stream 320 comprises the mode bit 322 (indicating a wideband or narrowband input signal) followed by the "I1" core bits 324 (either wideband or narrowband) and the "I2" core bits. Finally the "I1" narrowband extra bits or the "I2" wideband extra bits complete the preferred packing configuration.

[0062] Since one of the codebooks is fully searched, and for the other codebook only the core entries are searched, advantageously only one set of extra bits needs to be sent. The extra bits for the partially searched codebook will effectively be zero i.e. either 302, 306 and 308 or 310, 312, 314 need to be sent since either 304 or 316 will convey no information.

[0063] By packing the bits as shown in FIG. 4, the mode bit and core bits are beneficially always in the same locations. Hence, the impact of a mode bit error can be arranged to result in much smaller errors in the two quantisers than in prior art arrangements.

[0064] Clearly, in order to first generate the embedded vector codebook arrangement described in FIGs 1-4, a series of "test" input speech signals may be used, to obtain the optimum set of vectors to represent all input speech signals.

[0065] FIG. 5 shows two octagons 350 and 354, partitioned to reflect eight separate locations identified by a respective 3-bit address. The two octagons 350, 354 individually each represent one of the two split VQ codebooks of FIG. 3.

[0066] For simplicity, the example shows the case where three "extra" LSBs are used. The xxxx & yyyy represent core entry bit patterns for each of the respective embedded codebooks. For embedded entries, a potentially "non-zero extra" position will be appended instead of all zeros and the "extra" LSBs of the larger codebooks will be set to zero. As can be seen from error arrows 352 and 356, the maximum error for a (WB/NB) mode bit is equivalent to that of several LSB errors in each codebook.

[0067] In order to achieve this advantageous feature, the codebook entries of the embedded codebook, trained using relevant and appropriately varied speech patterns, must be interlaced regularly within the large codebook. In addition, index reassignment of the combined codebook must be performed such that LSB errors in the indices result in small perceptual distances. This may be arranged using a simulated annealing method as is well known to those skilled in the art.

[0068] A set of codebooks was derived and selectively searched. In order to determine which codebook configuration to search during each frame an appropriate narrowband speech indicator was employed.

[0069] Any reliable indicator may be used, but in this case the squared sum of the LSF VQ weights, w13-w18, corresponding to LSFs 13-18, (neglecting the split adjacent weighting) was filtered:

$$\text{(first-order recursive filter} = 0.1/(1-0.9{*}z^{-1})) \tag{1}$$

and thresholded.

[0070] This provides a reliable indication of whether the input is narrowband or wideband speech, except during silence periods, when the bias is slightly in favour of wideband speech.

[0071] The results of the hybrid wideband/narrowband scheme, in accordance with a preferred embodiment of the invention, is shown in Table 2.

Table 2:

7KHz and 3KHz Spectral Distortion Results for the Hybrid 41-bit Quantiser.						
Quantiser Configuration	Wideband Performance			Narrowband Performance		
	Mean SD (dB)	% Frames 3-5 dB	% Frames > 5 dB	Mean SD (dB)	% Frames 2-4 dB	% Frames > 4 dB
Hybrid 7,8,8,7,6,4 & 8,9,9,7,5,2	1.396	0.761	0.019	0.985	1.359	0.019

[0072] As can be seen by comparing the results of Table 2 with the values in Table 1, the hybrid scheme performs very well for both wideband and narrowband speech and offers only slightly degraded performance to the optimum in either configuration.

[0073] The graph 400 shown in FIG. 6 demonstrates the error resilience of the preferred embodiment of the invention in the presence of 10% bit errors, applied on a per-bit basis measured using an objective distortion measure, such as the perceptual speech quality measure (PSQM value), as defined by the ITU-T Recommendation P.861.

[0074] Graph 400 shows the bit error sensitivity profiles, this time for two 43-bit quantisers according to the embodiment. The distortion 402 (PSQM value) is shown plotted against bit number 404 on a bit-by-bit basis.

[0075] The two quantisers shown are the hybrid 8,8,8,7,6,5 & 8,9,9,8,6,2 wideband/narrowband scheme 406, according to the preferred embodiment of the invention, and an 8,8,8,7,7,5 wideband-only scheme 408.

[0076] For the hybrid quantiser the mode bit is the first bit and then the other core quantiser bits (8,8,8,7,6,2) are presented MSB first for each quantiser in-turn, followed finally by the three extra bits as depicted in FIG. 4. For the 8,8,8,7,7,5 wideband-only scheme the bits are presented in MSB first natural order.

[0077] The overall performance of the new quantiser in the presence of bit errors can be seen to be only very slightly worse than the wideband-only scheme (see rank-ordered sensitivities). In particular, the graph highlights that the sensitivity of the mode bit is 33rd out of 43, i.e. near, but not quite at, the bottom of the rank-ordered results.

[0078] The explanation for the mode bit not being the least sensitive bit (as in the optimal case) positioned at the bottom of the rank ordering (see FIG. 7) is that when a mode error occurs, several LSB changes (in the three-quantiser tables) occur which together are more significant than a single LSB change (bottom of the rank ordering). This clearly shows that the embedded structuring of the LSF VQ and bit stream has beneficially rendered the LSF VQ relatively immune to bit errors.

[0079] The graph 450 shown in FIG. 7 demonstrates the error resilience of the preferred embodiment of the invention where the error sensitivity profile is shown rank-ordered, as compared to a bit-by-bit basis as shown in FIG. 6.

[0080] Graph 450 shows the bit error sensitivity profiles, of the same two 43-bit quantisers, 456 and 458. The distortion 452 is shown plotted against re-ordered bit position 454 on a rank-ordered bit basis. In particular, the graph highlights that the two quantisers have broadly similar error sensitivity profiles and that the addition of the mode bit has not increased error sensitivity.

[0081] It will be understood that the bit-error robust embedded split vector quantiser for wideband line spectral frequencies (lsfs) in narrowband tandemming described above provides at least the following advantages.

[0082] The invention provides for a single speech codec codebook that can quantise both wideband and narrowband signals in a near optimal manner to that of two independently-optimised speech codec codebooks. This provides for a reduced memory requirement of the codebook, in a memory constrained speech unit.

[0083] The inventive concepts described herein find particular use in speech processing units that are flexible enough to cope with a variety of bandwidth constrained speech input signals, such as future third generation cellular telecommunications systems

[0084] It will, of course, be appreciated that the above description has been given by way of example only and that modifications in detail may be made within the scope of the present invention. For example, whilst the preferred embodiment discusses the application of the present invention to a split vector quantiser codebook, it is envisaged by the inventors that other codebooks and speech coder techniques can benefit from the inventive concepts contained herein.

[0085] The skilled addressee will equally appreciate that any number of line spectral frequencies can be accommodated, in a LSF codebook arrangement. Indeed, the present invention may be implemented outside of the line spectral frequency area, such as in video encoding and decoding.

[0086] It is within the contemplation of the invention that alternative quantiser configurations can benefit from such inventive concepts. Furthermore, the inventive concepts can be applied to any LPC order, with any bit-division relationship.

[0087] It is also within the contemplation of the present invention, that the inventive concepts contained herein can be equally employed in any classified overlapping codebook arrangement, not necessarily limited to the overlapping arrangement between wideband and narrowband speech signals.

[0088] Thus, a bit-error robust speech codec has been provided that is complementary to popular, proven speech codecs such as embedded split VQ codecs. The bit-error robust speech codec accommodates wideband line spectral frequencies in narrowband tandemming, and alleviates at least some of the aforementioned disadvantages.

Claims

1. A speech coder (100, 200) for a speech communications unit, the speech coder comprising a set of at least two embedded vector codebooks capable of representing an input speech signal by a series of vectors, the speech coder (100, 200) **characterised by** the at least two embedded vector codebooks (256, 262) sharing at least some common vectors of the series of vectors and means for classifying (252) the input speech signal by selecting one of the at least two embedded vector codebooks (256, 262) in conjunction with at least a portion of said at least one other embedded vector codebook of the at least two embedded vector codebooks (256, 262) to represent the input speech signal.

2. The speech coder of claim 1, wherein indices which address individual vector entries within the at least two embedded vector codebooks (256, 262) are assigned such that distortion resulting from incorrect classification is minimised.

3. The speech coder of claim 1 or claim 2, further **characterised by** the speech codebook being a vector quantisation codebook, wherein the at least two embedded vector codebooks (256, 262) are classified as predominantly wideband or narrowband embedded codebooks having shared common vectors.

4. The speech coder of any one of preceding claims 1 to 3, the speech coder further **characterised by** the selected one of the embedded vector codebooks (256, 262) providing a coarse resolution and the at least a portion of at

least one other embedded vector codebook providing a fine resolution to represent the speech signal input to the speech coder.

5. The speech coder of any one of the preceding claims, further **characterised by** quantisation means (110, 210) operably coupled to the means for classifying (252) for quantising a line spectral frequency input speech signal.

6. The speech coder of any one of the preceding claims, further **characterised by** analysis means operably coupled to the speech vector codebook for analysing an incoming speech signal to determine a particular characteristic of the speech signal.

7. The speech coder of claim 6, wherein the classifying means includes means for generating or recovering at least one mode bit to identify a characteristic of the speech signal associated with wideband or narrowband speech.

8. The speech coder of claim 7, wherein the input speech signal comprises at least one core bit, the speech coder further **characterised by** positioning means to position said mode bit and said at least one core bit for both first and second embedded vector codebooks (256, 262) in the same vector space.

9. The speech coder according to any one of the preceding claims, further comprising appending means, the appending means adapted to append at least one zero (304, 306) to a vector representation (300) to represent vector positions for either said first or second embedded speech vector codebook (256, 262) in a combined embedded speech vector codebook.

10. The speech coder according to any one of the preceding claims, further comprising index reassignment means (350) for re-arranging the vector positions of the first and second codebook vectors in the combined embedded speech vector codebook to minimise perceptual distance between said embedded codebook vectors.

11. A speech communications unit adapted to include the speech coder of any one of preceding claims 1 to 10.

12. A method of generating a speech vector codebook in a speech communications unit, the method **characterised by** the steps of:

    representing speech signals by a series of vectors; and

    generating at least two embedded vector codebooks that share common vectors of the series of vectors, to generate said speech vector codebook, such that, in use, at least a portion of each of the at least two embedded vector codebooks is used to represent an input speech signal.

13. The method of generating a speech codebook according to claim 12, wherein the step of generating at least two embedded vector codebooks includes the steps of:

    generating a first embedded vector codebook of the at least two embedded vector codebooks to substantially represent narrowband speech signals; and

    generating a second embedded vector codebook of the at least two embedded vector codebooks to substantially represent wideband speech signals.

14. The method of generating a speech vector codebook according to claims 12 or 13, the method further **characterised by** the step of:

    quantising, by quantisation means, a line spectral frequency input speech signal.

15. The method of generating a speech vector codebook according to any one of claims 12 to 14, wherein the step of generating at least two embedded vector codebooks includes generating one of the embedded vector codebooks to provide a coarse resolution, and generating the at least one other embedded vector codebook to provide a fine resolution, to represent a speech signal input to the speech coder.

16. The method of generating a speech vector codebook according to any one of claims 12 to 15, the method further **characterised by** the step of:

analysing an incoming speech signal by analysis means in the speech encoder to determine a particular characteristic of the speech signal associated with wideband or narrowband signals; and

generating a mode bit (322) to represent the particular characteristic of the speech signal.

17. The method of generating a speech vector codebook according to claim 16, wherein the speech signals include core speech bits (324, 326), the method further **characterised by** the step of:

positioning the at least one mode bit and core speech bits for both embedded vector codebooks in substantially the same location of the speech vector codebook.

18. The method of generating a speech vector codebook according to any one of preceding claims 12 to 17, the method further **characterised by** the step of:

appending (304, 316) zeros to vector entries in either of the first or second embedded vector codebooks.

19. The method of generating a speech vector codebook according to any one of preceding claims 12 to 18, the method further **characterised by** the step of:

performing index re-assignment of the speech vector codebook to maintain a relatively small perceptual distance between respective vector positions thereby minimising distortion errors resulting from incorrect classification in a speech encoding or decoding process.

20. The method of generating a speech vector codebook according to any one of preceding claims 12 to 19, the method further **characterised by** the step of:

interlacing said vector entries at substantially evenly distributed positions within the combined speech vector codebook.

21. A speech communications unit adapted to incorporate a speech vector codebook generated in accordance with any one of method claims 12 to 20.

22. A method of encoding a speech signal, the method **characterised by** the steps of:

receiving a speech signal;

identifying a characteristic of the speech signal;

selecting one of at least two embedded vector codebooks, in conjunction with at least a portion of at least one other embedded vector codebook of the at least two embedded vector codebooks, to represent the input speech signal based on the identified characteristic, the embedded vector codebooks sharing at least some common vectors; and

transmitting information identifying said selected one embedded vector codebook as a representation of said received speech signal.

23. The method of encoding a speech signal according to claim 22, wherein the step of identifying a characteristic of the received speech signal encompasses identifying whether the speech signal is associated with wideband or narrowband speech.

24. The method of encoding a speech signal according to claim 22 or 23, wherein the step of selecting includes selecting one of the embedded vector codebooks to provide a coarse resolution, and the at least a portion of at least one other embedded vector codebook to provide a fine resolution, to represent the speech signal input to the speech coder.

25. The method of encoding a speech signal according to any one of claim 22 to 24, the method further **characterised by** the step of:

generating at least one mode bit (322) to identify said characteristic of the speech signal.

26. The method of encoding a speech signal according to any one of preceding claims 22 to 25, the method further **characterised by** the step of:

    appending (304, 316) at least one zero to a vector representation of the input speech signal to represent said vector position for either said first or second embedded speech vector codebook in the speech vector codebook.

27. A speech communications unit adapted to employ a speech encoding method in accordance with any one of method claims 22 to 26.

28. A method of decoding a speech signal, the method **characterised by** the steps of:

    receiving a speech signal;

    identifying a characteristic of the received speech signal; and

    selecting one of at least two embedded vector codebooks, in conjunction with at least a portion of at least one other embedded vector codebook of the at least two embedded vector codebooks, to represent the input speech signal based on the identified characteristic, the embedded vector codebooks sharing at least some common vectors.

29. The method of decoding a speech signal according to claim 27, wherein the step of identifying a characteristic of the received speech signal encompasses identifying whether the speech signal is associated with wideband or narrowband speech.

30. The method of decoding a speech signal according to claim 28 or 29, the method further **characterised by** the step of:

    recovering (322) at least one mode bit to identify said characteristic of the speech signal.

31. The method of decoding a speech signal according to any one of preceding claims 28 to 30, the method further **characterised by** the step of:

    removing at least one zero appended to a vector representation of the input speech signal to represent a vector position in either said first or second embedded speech vector codebook.

32. A speech communications unit adapted to employ a speech decoding method in accordance with any one of method claims 28 to 31.

Patentansprüche

1. Sprachkodierer (100, 200) für eine Sprachkommunikationseinheit, wobei der Sprachkodierer einen Satz von wenigstens zwei eingebetteten Vektor-Codebüchern umfasst, welche in der Lage sind, ein Spracheingangssignal durch eine Reihe von Vektoren abzubilden, und wobei der Sprachkodierer (100, 200) **gekennzeichnet ist durch** die wenigstens zwei eingebetteten Vektor-Codebücher (256, 262), welche sich wenigstens einige gemeinsame Vektoren aus der Reihe von Vektoren teilen, und **durch** Mittel (256) zum Klassifizieren des Spracheingangssignals **durch** Auswählen von einem der wenigstens zwei eingebetteten Vektor-Codebücher (256, 262) in Verbindung mit wenigstens einem Teil des wenigstens einen anderen eingebetteten Vektor-Codebuchs der wenigstens zwei eingebetteten Vektor-Codebücher (256, 262), um das Spracheingangssignal abzubilden.

2. Sprachkodierer gemäß Anspruch 1, wobei Indices, welche individuelle Vektoreinträge innerhalb der wenigstens zwei eingebetteten Vektor-Codebücher (256, 262) adressieren, so zugeordnet sind, dass eine aus einer inkorrekten Klassifizierung resultierende Verzerrung minimiert wird.

3. Sprachkodierer nach Anspruch 1 oder nach Anspruch 2, weiterhin **dadurch gekennzeichnet, dass** es sich bei

dem Sprachcodebuch um ein Vektor-quantisierungscodebuch handelt, wobei die wenigstens zwei eingebetteten Vektor-Codebücher (256, 262) als überwiegend breitband- oder schmalband-eingebettete Codebücher mit aufgeteilten gemeinsamen Vektoren klassifiziert sind.

4. Sprachkodierer nach einem der vorangegangenen Ansprüche 1 bis 3, wobei der Sprachkodierer weiterhin **dadurch gekennzeichnet ist, dass** das ausgewählte der eingebetteten Vektor-Codebücher (256, 262) eine grobe Auflösung ergibt und dass der wenigstens eine Teil von wenigstens einem anderen eingebetteten Codebuch eine hohe Auflösung ergibt, um das in den Sprachkodierer eingegebene Sprachsignal abzubilden.

5. Sprachkodierer nach einem der vorangegangenen Ansprüche, weiterhin **gekennzeichnet durch** Quantisierungsmittel (110, 210), welche betriebsfähig an die Mittel zum Klassifizieren (252) gekoppelt sind, zum Quantisieren eines Spracheingangssignals mit einer Linienspektralfrequenz.

6. Sprachkodierer nach einem der vorangegangenen Ansprüche, weiterhin **gekennzeichnet durch** Analysemittel, welche betriebsfähig an das Sprachvektor-Codebuch gekoppelt sind, zum Analysieren eines eingehenden Sprachsignals, um eine spezifische Eigenschaft des Sprachsignals zu bestimmen.

7. Sprachkodierer nach Anspruch 6, wobei die Klassifizierungsmittel Mittel zum Erzeugen oder Wiederherstellen von wenigstens einem Modusbit umfassen, zum Identifizieren einer Eigenschaft des Sprachsignals, die breitbandiger oder schmalbandiger Sprache zugeordnet ist.

8. Sprachkodierer nach Anspruch 7, wobei das Spracheingangssignal wenigstens ein Kernbit umfasst und wobei der Sprachkodierer weiterhin **gekennzeichnet ist durch** Positionierungsmittel zum Positionieren des Modusbits und des wenigstens einen Kernbits für sowohl das erste wie auch das zweite eingebettete Vektor-Codebuch (256, 262) in demselben Vektorraum.

9. Sprachkodierer nach einem der vorangegangenen Ansprüche, weiterhin umfassend Anhängemittel, wobei die Anhängemittel ausgebildet sind, wenigstens eine Null (304, 306) an eine Vektordarstellung (300) anzuhängen, um Vektorpositionen für entweder das erste oder das zweite eingebettete Sprachvektor-Codebuch (256, 262) in einem kombinierten eingebetteten Sprachvektor-Codebuch darzustellen.

10. Sprachkodierer gemäß einem der vorangegangenen Ansprüche, weiterhin umfassend Index-Neuzuordnungs-Mittel (350) zum Neuanordnen der Vektorpositionen der ersten und zweiten Codebuch-Vektoren in dem kombinierten eingebetteten Sprachvektor-Codebuch, um einen Wahrnehmungsabstand zwischen den eingebetteten Codebuch-Vektoren zu minimieren.

11. Sprachkommunikationseinheit, welche ausgebildet ist, den Sprachkodierer gemäß einem der vorangegangenen Ansprüche 1 bis 10 zu umfassen.

12. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs in einer Sprachkommunikationseinheit, wobei das Verfahren durch folgende Schritte **gekennzeichnet** ist:

Abbilden von Sprachsignalen durch eine Reihe von Vektoren; und
Erzeugen von wenigstens zwei eingebetteten Vektor-Codebüchern; welche sich gemeinsame Vektoren aus der Reihe von Vektoren teilen, um das Sprachvektor-Codebuch zu erzeugen, so dass - bei Verwendung - wenigstens ein Teil von jedem der wenigstens zwei eingebetteten Vektor-Codebücher verwendet wird, um ein Spracheingangssignal abzubilden.

13. Verfahren zum Erzeugen eines Sprachcodebuchs gemäß Anspruch 12, wobei der Schritt der Erzeugung von wenigstens zwei eingebetteten Vektor-Codebüchern die Schritte umfasst:

Erzeugen eines ersten eingebetteten Vektor-Codebuchs von den wenigstens zwei eingebetteten Vektor-Codebüchern, um im Wesentlichen Schmalband-Sprachsignale abzubilden beziehungsweise zu repräsentieren; und
Erzeugen eines zweiten eingebetteten Vektor-Codebuchs der wenigstens zwei eingebetteten Vektor-Codebücher, um im Wesentlichen Breitband-Sprachsignale abzubilden beziehungsweise zu repräsentieren.

14. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß den Ansprüchen 12 oder 13, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Quantisieren eines Spracheingangssignals mit einer Linienspektralfrequenz mithilfe von Quantisierungsmitteln.

15. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß einem der Ansprüche 12 bis 14, wobei der Schritt der Erzeugung von wenigstens zwei eingebetteten Vektor-Codebüchern umfasst:

Erzeugen eines der eingebetteten Vektor-Codebücher, um eine grobe Auflösung vorzusehen; und
Erzeugen des wenigstens einen anderen eingebetteten Vektor-Codebuchs, um eine hohe Auflösung vorzusehen, um ein in den Sprachkodierer eingegebenes Sprachsignal abzubilden.

16. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß einem der Ansprüche 12 bis 15, wobei das Verfahren weiterhin **gekennzeichnet ist durch** die Schritte:

Analysieren eines eingehenden Sprachsignals mithilfe von Analysemitteln in dem Sprachkodierer, um eine bestimmte Eigenschaft des Sprachsignals zu bestimmen, die Breitband- oder Schmalbandsignalen zugeordnet ist; und
Erzeugen eines Modusbits (322), um die bestimmte Eigenschaft des Sprachsignals darzustellen.

17. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß Anspruch 16, wobei das Sprachsignal Kernsprachbits (324, 326) umfasst und das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Positionieren des wenigstens einen Modusbits und der Kernsprachbits für beide eingebetteten Vektor-Codebücher an im Wesentlichen derselben Stelle des Sprachvektor-Codebuchs.

18. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß einem der vorangegangenen Ansprüche 12 bis 17, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Anhängen (304, 316) von Nullen an Vektoreinträge in entweder dem ersten oder dem zweiten eingebetteten Vektor-Codebuch.

19. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß einem der vorangegangenen Ansprüche 12 bis 18, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Durchführen einer Index-Neuzuordnung des Sprachvektor-Codebuchs, um einen relativ kleinen Wahrnehmungsabstand zwischen den jeweiligen Vektorpositionen aufrechtzuerhalten, wodurch Verzerrungsfehler, welche aus einer inkorrekten Klassifizierung in einem Sprachcodierungs- oder -decodierungsprozess resultieren, minimiert werden.

20. Verfahren zum Erzeugen eines Sprachvektor-Codebuchs gemäß einem der vorangegangenen Ansprüche 12 bis 19, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Verflechten der Vektoreinträge an im Wesentlichen gleichmäßig verteilten Positionen innerhalb des kombinierten Sprachvektor-Codebuchs.

21. Sprachkommunikationseinheit, welche ausgebildet ist, ein gemäß einem der Verfahrensansprüche 12 bis 20 erzeugtes Sprachvektor-Codebuch einzuschließen.

22. Verfahren zum Codieren eines Sprachsignals, wobei das Verfahren **gekennzeichnet ist durch** die Schritte:

Empfangen eines Sprachsignals;
Identifizieren einer Eigenschaft des Sprachsignals;
Auswählen eines von wenigstens zwei eingebetteten Vektor-Codebüchern in Verbindung mit wenigstens einem Teil von wenigstens einem anderen eingebetteten Vektor-Codebuch der wenigstens zwei eingebetteten Vektor-Codebücher, um das Spracheingangssignal, basierend auf der identifizierten Eigenschaft abzubilden, wobei die eingebetteten Vektor-Codebücher wenigstens einige der gemeinsamen Vektoren miteinander teilen; und

Übertragen von Informationen, welche das ausgewählte eingebettete Vektor-Codebuch identifizieren, als eine Darstellung des empfangenen Sprachsignals.

23. Verfahren zum Codieren eines Sprachsignals gemäß Anspruch 22, wobei der Schritt der Identifizierung einer Eigenschaft des empfangenen Sprachsignals eine Identifizierung, ob das Sprachsignal breitbandiger oder schmalbandiger Sprache zugeordnet ist, einschließt.

24. Verfahren zum Codieren eines Sprachsignals gemäß der Ansprüche 22 oder 23, wobei der Schritt des Auswählens umfasst:

Auswählen von einem der eingebetteten Vektor-Codebücher zum Bereitstellen einer groben Auflösung und Auswählen von dem wenigstens einen Teil des wenigstens einen anderen eingebetteten Vektor-Codebuchs zum Bereitstellen einer hohen Auflösung, um das in den Sprachkodierer eingegebene Sprachsignal abzubilden.

25. Verfahren zum Codieren eines Sprachsignals gemäß einem der Ansprüche 22 bis 24, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Erzeugen von wenigstens einem Modusbit (322) zum Identifizieren der Eigenschaft des Sprachsignals.

26. Verfahren zum Codieren eines Sprachsignals gemäß einem der vorangegangenen Ansprüche 22 bis 25, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Anhängen (304, 316) von wenigstens einer Null an eine Vektorabbildung des Spracheingangssignals, um die Vektorposition für entweder das erste oder das zweite eingebettete Sprachvektor-Codebuch in dem Sprachvektor-Codebuch abzubilden.

27. Sprachkommunikationseinheit, welche ausgebildet ist, ein Sprachcodierungsverfahren gemäß einem der Verfahrensansprüche 22 bis 26 zu benutzen.

28. Verfahren zum Decodieren eines Sprachsignals, wobei das Verfahren **gekennzeichnet ist durch** die Schritte:

Empfangen eines Sprachsignals;
Identifizieren einer Eigenschaft des empfangenen Sprachsignals; und
Auswählen von einem von wenigstens zwei eingebetteten Vektor-Codebücher in Verbindung mit wenigstens einem Teil von wenigstens einem anderen eingebetteten Vektor-Codebuch der wenigstens zwei eingebetteten Vektor-Codebücher, um das Spracheingangssignal, basierend auf der identifizierten Eigenschaft darzustellen, wobei die eingebetteten Vektor-Codebücher sich wenigstens einige gemeinsame Vektoren teilen.

29. Verfahren zum Decodieren eines Sprachsignals gemäß Anspruch 27, wobei der Schritt der Identifizierung einer Eigenschaft des empfangenen Sprachsignals eine Identifizierung umfasst, ob das Sprachsignal einer breitbandigen oder schmalbandigen Sprache zugeordnet ist.

30. Verfahren zum Decodieren eines Sprachsignals gemäß Anspruch 28 oder 29, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Wiederherstellen (322) des wenigstens einen Modusbits zum Identifizieren der Eigenschaft des Sprachsignals.

31. Verfahren zum Decodieren eines Sprachsignals gemäß einem der vorangegangenen Ansprüche 28 bis 30, wobei das Verfahren weiterhin **gekennzeichnet ist durch** den Schritt:

Entfernen von wenigstens einer Null, welche an eine Vektorabbildung des Spracheingangssignals angehangen wurde, um eine Vektorposition in entweder dem ersten oder dem zweiten eingebetteten Sprachvektor-Codebuch darzustellen.

32. Sprachkommunikationseinheit, welche ausgebildet ist, ein Sprachdecodierungsverfahren gemäß einem der Verfahrensansprüche 28 bis 31 zu benutzen.

Revendications

1. Codeur vocal (100, 200) pour unité de communications vocales, le codeur vocal comprenant un ensemble d'au moins deux livres de codes vectoriels intégrés, susceptible de représenter un signal vocal d'entrée par une série de vecteurs, le codeur vocal (100, 200) étant **caractérisé par le fait que** les deux ou plus de deux livres de codes vectoriels intégrés (256, 262) se partagent au moins certains vecteurs communs de la série de vecteurs, et par un moyen servant à classer (252) le signal vocal d'entrée en sélectionnant l'un des deux ou plus de deux livres de codes vectoriels intégrés (256, 262) en liaison avec au moins une partie dudit autre ou desdits autres livres de codes vectoriels intégrés des deux ou plus de deux livres de codes vectoriels intégrés (256, 262) pour représenter le signal vocal d'entrée.

2. Codeur vocal selon la revendication 1, où des indices, qui adressent des points d'entrée de vecteurs individuels à l'intérieur des deux ou plus de deux livres de codes vectoriels intégrés (256, 262), sont affectés de façon que la distorsion résultant d'un classement incorrect soit minimisée.

3. Codeur vocal selon la revendication 1 ou 2, **caractérisé en outre par le fait que** le livre de code vocal est un livre de code de quantification vectorielle, où les deux ou plus de deux livres de codes vectoriels intégrés (256, 262) sont classés en tant que livres de codes intégrés présentant de façon prédominante une bande large ou une bande étroite et ayant des vecteurs partagés en commun.

4. Codeur vocal selon l'une quelconque des revendications 1 à 3, le codeur vocal étant en outre **caractérisé en ce que** le livre sélectionné parmi les livres de codes vectoriels intégrés (256, 262) fournit une résolution grossière et ladite ou lesdites parties dudit ou desdits autres livres de codes vectoriels intégrés fournissent une résolution fine de façon à représenter le signal vocal appliqué en entrée au codeur vocal.

5. Codeur vocal selon l'une quelconque des revendications précédentes, **caractérisé en outre par** un moyen de quantification (110, 210) qui est couplé de façon fonctionnelle au moyen de classement (252) afin de quantifier un signal vocal d'entrée de fréquence spectrale linéaire.

6. Codeur vocal selon l'une quelconque des revendications précédentes, **caractérisé en outre par** un moyen d'analyse fonctionnellement couplé au livre de code vectoriel vocal afin d'analyser un signal vocal entrant pour déterminer une caractéristique particulière du signal vocal.

7. Codeur vocal selon la revendication 6, où le moyen de classement comporte un moyen servant à produire ou récupérer au moins un bit de mode afin d'identifier une caractéristique du signal vocal associé à un signal vocal à large bande ou à bande étroite.

8. Codeur vocal selon la revendication 7, où le signal vocal d'entrée comprend au moins un bit de coeur, le codeur vocal étant en outre **caractérisé par** un moyen de positionnement afin de positionner dans le même espace vectoriel ledit bit de mode et ledit ou lesdits bits de coeur se rapportant aux premier et deuxième livres de codes vectoriels intégrés (256, 262) tous les deux.

9. Codeur vocal selon l'une quelconque des revendications précédentes, comprenant en outre un moyen d'ajout, le moyen d'ajout étant destiné à ajouter au moins un zéro (304, 306) à une représentation vectorielle (300) afin de représenter des positions vectorielles pour l'un ou l'autre desdits premier et deuxième livres de codes vectoriels vocaux intégrés (256, 262) en un livre de code vectoriel vocal intégré combiné.

10. Codeur vocal selon l'une quelconque des revendications précédentes, comprenant en outre un moyen (350) de réaffectation d'indices destiné à réarranger les positions vectorielles des vecteurs des premier et deuxième livres de codes dans le livre de code vectoriel vocal intégré combiné afin de minimiser la distance perceptuelle existant entre lesdits vecteurs des livres de codes intégrés.

11. Unité de communications vocales conçue pour incorporer le codeur vocal défini dans l'une quelconque des revendications 1 à 10.

12. Procédé de production d'un livre de code vectoriel vocal dans une unité de communications vocales, le procédé étant **caractérisé par** les opérations suivantes :

représenter des signaux vocaux par une série de vecteurs ; et
produire au moins deux livres de codes vectoriels intégrés qui partagent des vecteurs communs de la série de vecteurs, afin de produire ledit livre de code vectoriel vocal, de façon que, en utilisation, au moins une partie de chaque livre des deux ou plus de deux livres de codes vectoriels intégrés est utilisée pour représenter un signal vocal d'entrée.

13. Procédé de production d'un livre de code vocal selon la revendication 12, où l'opération de production d'au moins deux livres de codes vectoriels intégrés comporte les opérations suivantes :

produire un premier livre de code vectoriel intégré des deux ou plus de deux livres de codes vectoriels intégrés afin de représenter sensiblement des signaux vocaux à bande étroite ; et
produire un deuxième livre de code vectoriel intégré des deux ou plus de deux livres de codes vectoriels intégrés afin de représenter sensiblement des signaux vocaux à bande large.

14. Procédé de production d'un livre de code vectoriel intégré selon la revendication 12 ou 13, le procédé étant en outre **caractérisé par** l'opération suivante :

quantifier, avec un moyen de quantification, un signal vocal d'entrée de fréquence spectrale linéaire.

15. Procédé de production d'un livre de code vectoriel intégré selon l'une quelconque des revendications 12 à 14, où l'opération de production d'au moins deux livres de codes vectoriels intégrés comporte la production d'un des livres de codes vectoriels intégrés afin de produire une résolution grossière et la production d'au moins un autre livre de code vectoriel intégré afin de produire une résolution fine, pour représenter le signal vocal appliqué en entrée au codeur vocal.

16. Procédé de production d'un livre de code vectoriel vocal selon l'une quelconque des revendications 12 à 15, le procédé étant en outre **caractérisé par** les opérations suivantes :

analyser un signal vocal entrant avec un moyen d'analyse du codeur vocal afin de déterminer une caractéristique particulière du signal vocal associé à des signaux à bande large ou à bande étroite ; et
produire un bit de mode (322) afin de représenter la caractéristique particulière du signal vocal.

17. Procédé de production d'un livre de code vectoriel vocal selon la revendication 16, où les signaux vocaux comportent des bits vocaux de coeur (324, 326), le procédé étant en outre **caractérisé par** l'opération suivante :

positionner le ou les bits de mode et les bits vocaux de coeur afin que les livres de codes vectoriels intégrés soient tous deux sensiblement au même emplacement du livre de code vectoriel vocal.

18. Procédé de production d'un livre de code vectoriel vocal selon l'une quelconque des revendications 12 à 17, le procédé étant en outre **caractérisé par** l'opération suivante :

ajouter (304, 316) des zéros aux points d'entrée vectoriels dans le premier ou le deuxième livre de code vectoriel intégré.

19. Procédé de production d'un livre de code vectoriel vocal selon l'une quelconque des revendications 12 à 18, le procédé étant en outre **caractérisé par** l'opération suivante :

effectuer une réaffectation des indices du livre de code vectoriel vocal afin de maintenir une distance perceptuelle relativement petite entre des positions vectorielles respectives, pour ainsi minimiser les erreurs de distorsion qui résultent d'un classement incorrect dans un processus de codage ou de décodage de signaux vocaux.

20. Procédé de production d'un livre de code vectoriel vocal selon l'une quelconque des revendications 12 à 19, le procédé étant en outre **caractérisé par** l'opération suivante :

entrelacer lesdits points d'entrée vectoriels en des positions qui sont sensiblement réparties de façon uniforme à l'intérieur du livre de code vectoriel vocal combiné.

21. Unité de communications vocales conçue pour incorporer un livre de code vectoriel vocal produit selon l'une quelconque des revendications de procédé 12 à 20.

22. Procédé de codage d'un signal vocal, le procédé étant **caractérisé par** les opérations suivantes :

recevoir un signal vocal ;
identifier une caractéristique du signal vocal ;
sélectionner l'un d'au moins deux livres de codes vectoriels intégrés en liaison avec au moins une partie de l'autre ou des autres livres de codes vectoriels intégrés desdits deux ou plus de deux livres de codes vectoriels intégrés, afin de représenter le signal vocal d'entrée sur la base de la caractéristique identifiée, les livres de codes vectoriels intégrés se partageant au moins certains vecteurs communs ; et
transmettre des informations identifiant ledit livre de code vectoriel intégré qui a été sélectionné au titre d'une représentation dudit signal vocal reçu.

23. Procédé de codage d'un signal vocal selon la revendication 22, où l'opération d'identification d'une caractéristique du signal vocal reçu recouvre l'identification du fait que le signal vocal est associé à un signal vocal à large bande ou un signal vocal à bande étroite.

24. Procédé de codage d'un signal vocal selon la revendication 22 ou 23, où l'opération de sélection comporte la sélection de l'un des livres de codes vectoriels intégrés afin de produire une résolution grossière, et de la partie ou des parties dudit ou desdits autres livres de codes vectoriels intégrés afin de produire une résolution fine, pour ainsi représenter le signal vocal appliqué en entrée au codeur vocal.

25. Procédé de codage d'un signal vocal selon l'une quelconque des revendications 22 à 24, le procédé étant en outre **caractérisé par** l'opération suivante :

produire au moins un bit de mode (322) afin d'identifier la caractéristique du signal vocal.

26. Procédé de codage d'un signal vocal selon l'une quelconque des revendications 22 à 25, le procédé étant en outre **caractérisé par** l'opération suivante :

ajouter (304, 316) au moins un zéro à une représentation vectorielle du signal vocal d'entrée afin de représenter ladite position vectorielle relative auxdits premier ou deuxième livres de codes vectoriels vocaux intégrés se trouvant dans ledit livre de code vectoriel vocal.

27. Unité de communications vocales conçue pour employer un procédé de codage vocal selon l'une quelconque des revendications 22 à 26.

28. Procédé de décodage d'un signal vocal, le procédé étant **caractérisé par** les opérations suivantes :

recevoir un signal vocal ;
identifier une caractéristique du signal vocal reçu ; et
sélectionner l'un d'au moins deux livres de codes vectoriels intégrés en liaison avec au moins une partie d'au moins un autre livre de code vectoriel intégré desdits deux ou plus de deux livres de codes vectoriels intégrés, afin de représenter le signal vocal d'entrée sur la base de la caractéristique identifiée, les livres de codes vectoriels intégrés se partageant au moins quelques vecteurs communs.

29. Procédé de décodage d'un signal vocal selon la revendication 27, où l'opération d'identification d'une caractéristique du signal vocal reçu recouvre l'identification du fait que le signal vocal est associé à un signal vocal à large bande ou à bande étroite.

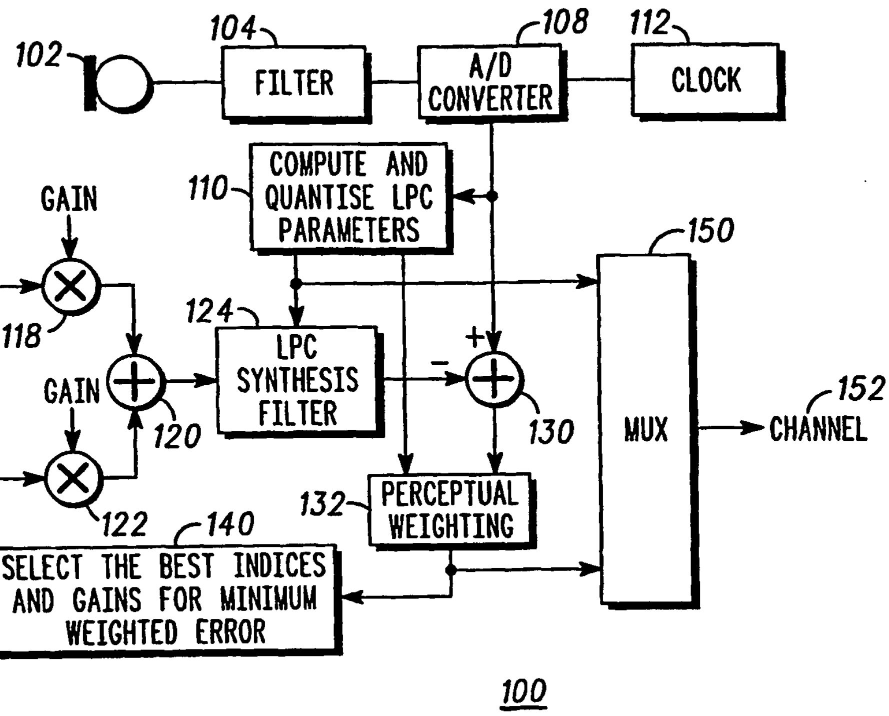
30. Procédé de décodage d'un signal vocal selon la revendication 28 ou 29, le procédé étant en outre **caractérisé par** l'opération suivante :

récupérer (322) au moins un bit de mode afin d'identifier ladite caractéristique du signal vocal.

31. Procédé de décodage d'un signal vocal selon l'une quelconque des revendications 28 à 30, le procédé étant **caractérisé en outre par** l'opération suivante :

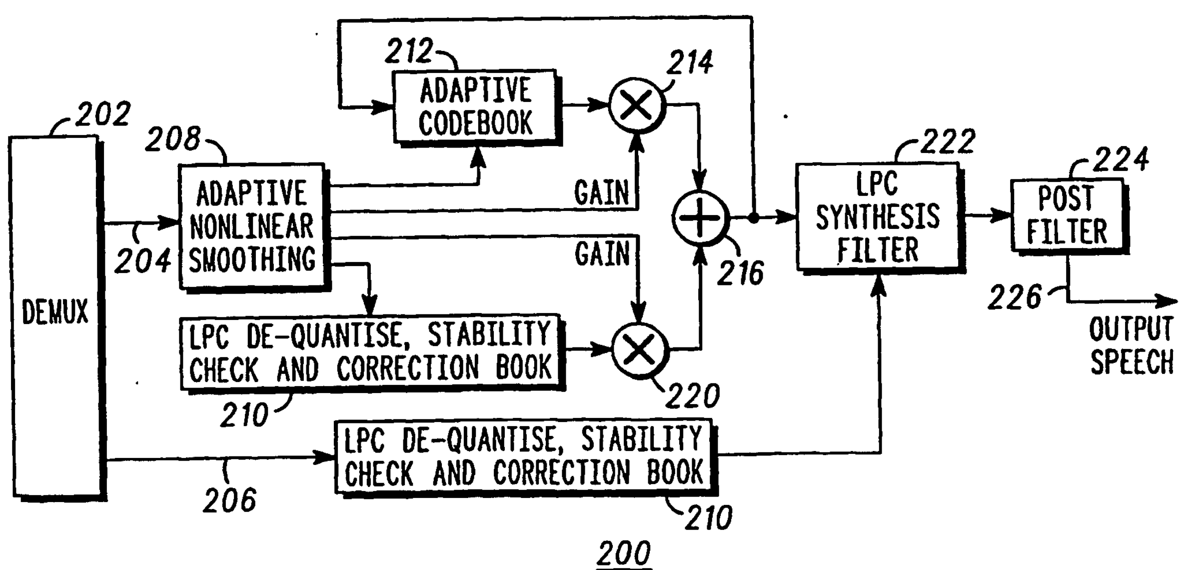
retirer au moins un zéro ajouté à une représentation vectorielle du signal vocal d'entrée afin de représenter la position vectorielle dans ledit premier ou ledit deuxième livre de code vectoriel vocal intégré.

32. Unité de communications vocales conçue pour employer un procédé de décodage vocal tel que défini dans l'une quelconque des revendications de procédé 28 à 31.



102
104
FILTER
108
A/D CONVERTER
112
CLOCK
110
COMPUTE AND QUANTISE LPC PARAMETERS
GAIN
116
ADAPTIVE CODEBOOK
118
150
124
LPC SYNTHESIS FILTER
GAIN
120
+
-
130
MUX
152
CHANNEL
114
SCHOCHASTIC CODEBOOK
122
140
132
PERCEPTUAL WEIGHTING
SELECT THE BEST INDICES AND GAINS FOR MINIMUM WEIGHTED ERROR
100

FIG. 1



212
ADAPTIVE CODEBOOK
214
202
208
ADAPTIVE NONLINEAR SMOOTHING
GAIN
222
LPC SYNTHESIS FILTER
224
POST FILTER
204
GAIN
216
226
OUTPUT SPEECH
DEMUX
LPC DE-QUANTISE, STABILITY CHECK AND CORRECTION BOOK
220
210
LPC DE-QUANTISE, STABILITY CHECK AND CORRECTION BOOK
206
210
200

FIG. 2

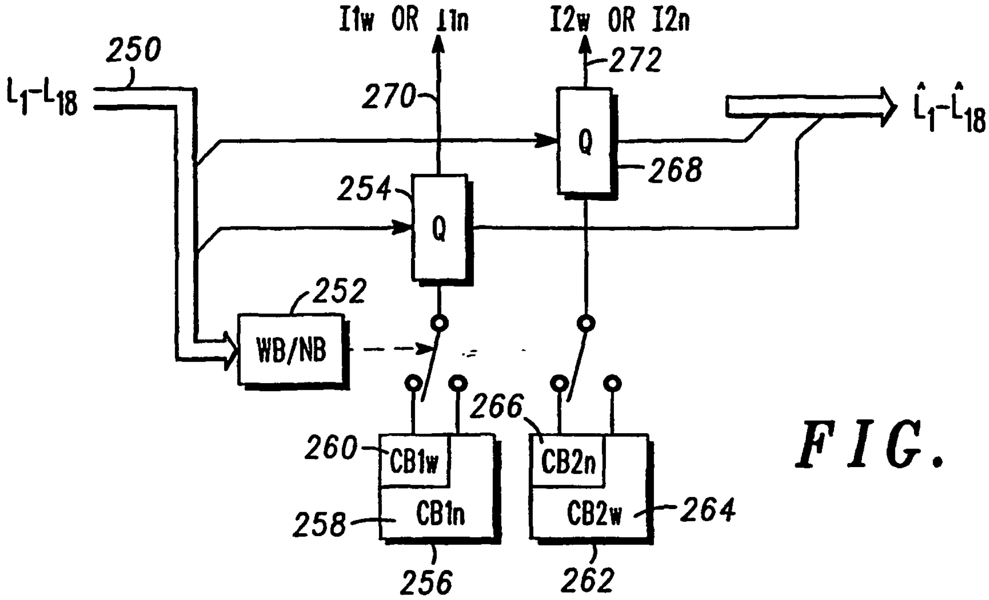


FIG. 3

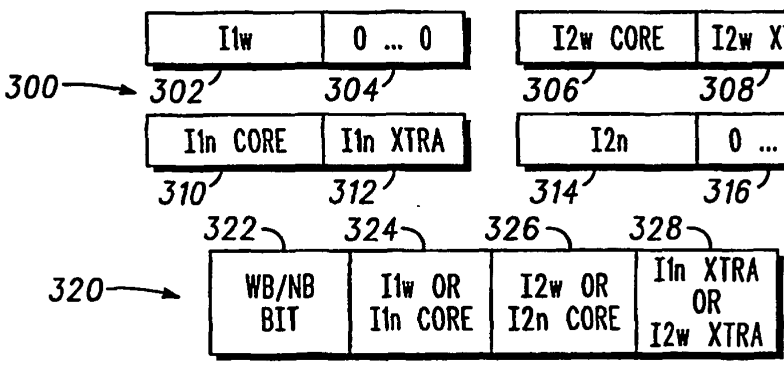


FIG. 4

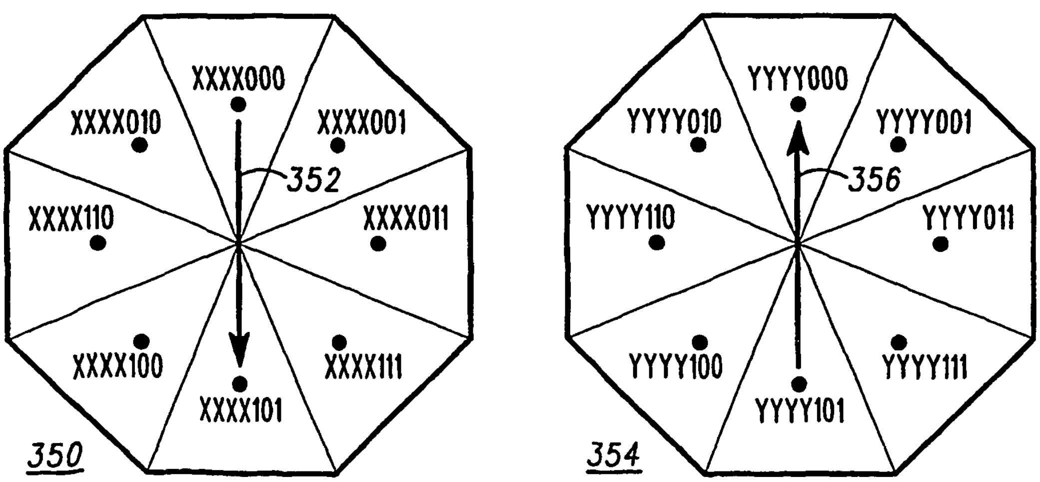


FIG. 5

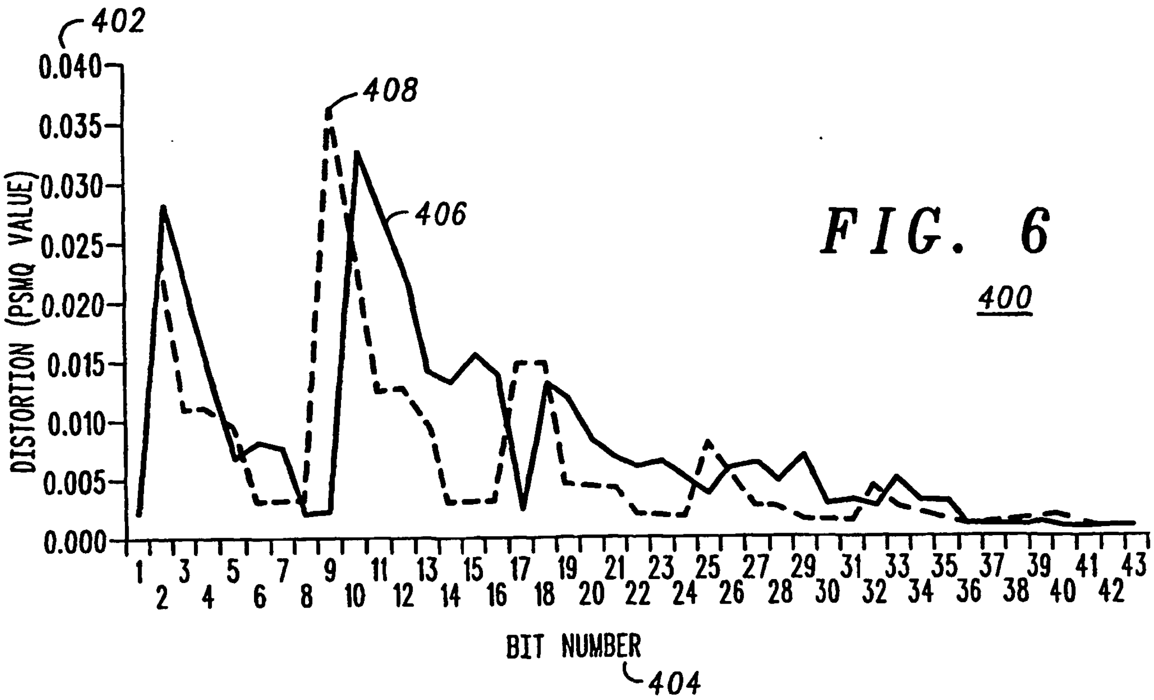


FIG. 6

400

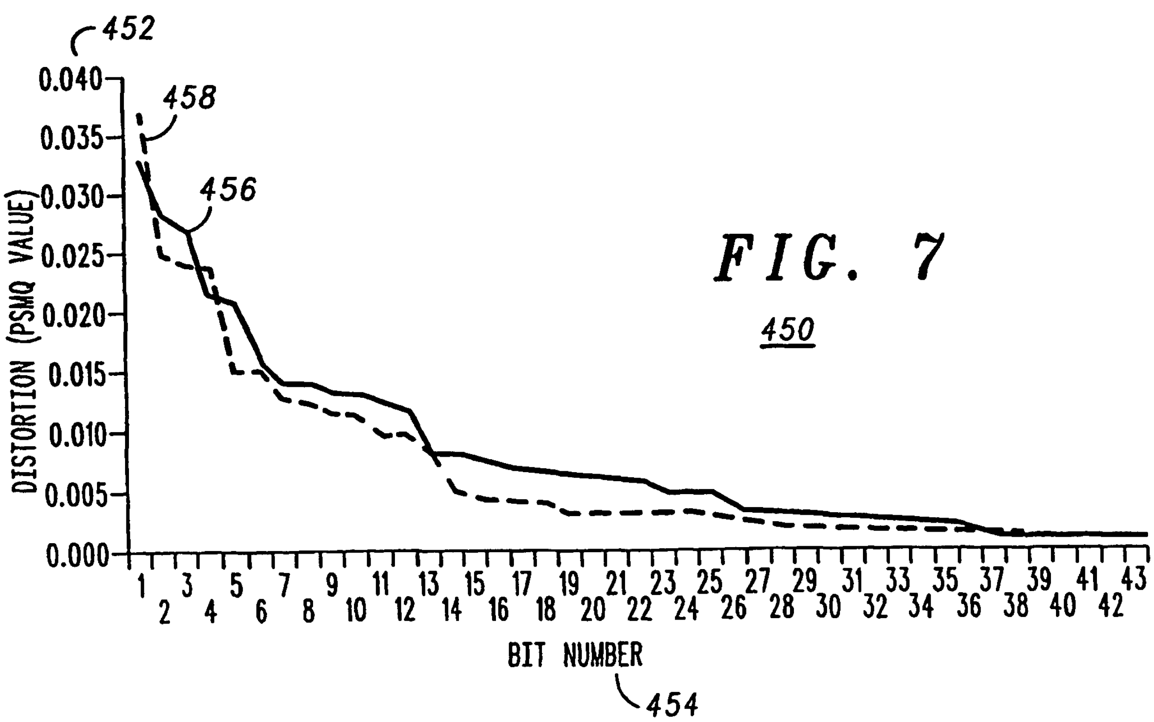


FIG. 7

450