Description

Technical Field

This invention relates to low bit rate coding and decoding of speech and in particular to an improved code excited linear predictive vocoder that provides high performance.

Background of the Invention

Code excited linear predictive coding (CELP) is a well-known technique. This coding technique synthesizes speech by utilizing encoded excitation information to excite a linear predictive coding (LPC) filter. This excitation is found by searching through a table of excitation vectors on a frame-by-frame basis. The table, also referred to as codebook, is made up of vectors whose components are consecutive excitation samples. Each vector contains the same number of excitation samples as there are speech samples in a frame.

A method of encoding speech comprising frames having a plurality of samples, including calculating an error value for each of a plurality of candidate excitation frames and selecting the one with the smallest error value, is disclosed by Trancoso and Atal in 'Efficient Procedures for finding the Optimum Innovation in Stochastic Encoders' in ICASSP 86, Proceedings of the IEEE-IECEJ-ASJ International Conference on Acoustics, Speech and Signal Processing, Tokyo 7th-11th April 1986, volume 4 of 4, pages 2375-2378.

It is known for such a codebook to be constructed as an overlapping table in which the excitation vectors are defined by shifting a window along a linear array of excitation samples. The analysis is performed by first doing an LPC analysis on a speech frame to obtain a LCP filter that is then excited by the various candidate vectors in the codebook. The best candidate vector is chosen on how well its corresponding synthesis output matches a frame of speech. After the best match has been found, information specifying the best codebook entry and the filter are transmitted to the synthesizer. The synthesizer has a similar codebook and accesses the appropriate entry in that codebook and uses it to excite an identical LPC filter. In addition, it utilizes the best candidate excitation vector to update the codebook so that the codebook adapts to the speech.

The problem with this technique is that the codebook adapts very slowly during speech transitions such as from unvoiced regions to voiced regions of speech. Voiced regions of speech are characterized in that a fundamental frequency is present in the speech. This problem is particularly noticeable for women since the fundamental frequencies that can be generated by women are higher than those for men.

Solution

The foregoing problem is solved according to the invention by apparatus and methods as set out in the claims.

Brief Description of the Drawings.

Some embodiments of the invention will now be described by way of example with reference to the accompanying drawings, in which:

FIG. 1 illustrates, in block diagram form, analyzer and synthesizer sections of a vocoder which embodies this invention;

FIG. 2 illustrates, in graphic form, the formation of excitation vectors from codebook 104 using virtual search technique which embodies this invention;

FIGS. 3 through 6 illustrate, in graphic form, the vector and matrix operation used in selecting the best candidate vector;

FIG. 7 illustrates, in greater detail, adaptive searcher 106 of FIG. 1;

FIG. 8 illustrates, in greater detail, virtual search control 708 of FIG 7; and

FIG. 9 illustrates, in greater detail, energy calculator 709 of FIG. 7.

Detailed Description

FIG. 1 illustrates, in block diagram form, a vocoder. Elements 101 through 112 represent the analyzer portion of the vocoder; whereas, elements 151 through 157 represent the synthesizer portion of the vocoder. The analyzer portion of FIG. 1 is responsive to incoming speech received on path 120 to digitally

sample the analog speech into digital samples and to group those digital samples into frames using well-known techniques. For each frame, the analyzer portion calculates the LPC coefficients representing the formant characteristics of the vocal tract and searches for entries from both the stochastic codebook 105 and adaptive codebook 104 that best approximate the speech for that frame along with scaling factors. The latter entries and scaling information define excitation information as determined by the analayzer portion. This excitation and coefficient information is then transmitted by encoder 109 via path 145 to the synthesizer portion of the vocoder illustrated in FIG. 1. Stochastic generator 153 and adaptive generator 154 are responsive to the codebook entries and scaling factors to reproduce the excitation information calculated in the analyzer portion of the vocoder and to utilize this excitation information to excite the LPC filter that is determined by the LPC coefficients received from the analyzer portion to reproduce the speech.

Consider now in greater detail the functions of the analyzer portion of FIG. 1 LPC analyzer 101 is responsive to the incoming speech to determine LPC coefficients using well-known techniques. These LPC coefficients are transmitted to target excitation calculator 102, spectral weighting calculator 103, encoder 109, LPC filter 110, and zero-input response filter 111. Encoder 109 is responsive to the LPC coefficients to transmit the latter coefficients via path 145 to decoder 151. Spectral weighting calculator 103 is responsive to the coefficients to calculate spectral weighting information in the form of a matrix that emphasizes those portions of speech that are known to have important speech content. This spectral weighting information is based on a finite impulse response LPC filter. The utilization of a finite impulse response filter will be shown to greatly reduce the number of calculations necessary for performing the computations performed in searchers 106 and 107. This spectral weighting information is utilized by the searchers in order to determine the best candidate for the excitation information from the codebooks 104 and 105.

Target excitation calculator 102 calculates the target excitation which searchers 106 and 107 attempt to approximate. This target excitation is calculated by convolving a whitening filter based on the LPC coefficients calculated by analyzer 101 with the incoming speech minus the effects of the excitation and LPC filter for the previous frame. The latter effects for the previous frames are calculated by filters 110 and 111. The reason that the excitation and LPC filter for the previous frame must be considered is that these factors produce a signal component in the present frame which is often referred to as the ringing of the LPC filter. As will be described later, filters 110 and 111 and responsive to the LPC coefficients and calculated excitation from the previous frame to determine this ringing signal and to transmit it via path 144 to subtracter 112, Subtracter 112 is responsive to the latter signal and the present speech to calculate a remainder signal representing the present speech minus the ringing signal. Calculator 102 is responsive to the remainder signal to calculate the target excitation information and to transmit the latter information via path 123 to searcher 106 and 107.

The latter searchers work sequentially to determine the calculated excitation also referred to as synthesis excitation which is transmitted in the form of codebook indices and scaling factors via encoder 109 and path 145 to the synthesizer portion of FIG. 1. Each searcher calculates a portion of the calculated excitation. First, adaptive searcher 106 calculates excitation information and transmits this via path 127 to stochastic searcher 107. Searcher 107 is responsive to the target excitation received via path 123 and the excitation information from adaptive searcher 106 to calculate the remaining portion of the calculated excitation that best approximates the target excitation calculated by calculator 102. Searcher 107 determines the remaining excitation to be calculated by subtracting the excitation determined by searcher 106 from the target excitation. The calculated or synthetic excitation determined by searchers 106 and 107 is transmitted via paths 127 and 126, respectively, to adder 108. Adder 108 adds the two excitation components together to arrive at a synthetic excitation for the present frame. The synthetic excitation is used by the synthesizer to produce the synthesized speech.

The output of adder 108 is also transmitted via path 128 to LPC filter 110 and adaptive codebook 104. The excitation information transmitted via path 128 is utilized to update adaptive codebook 104. The codebook indices and scaling factors are transmitted from searchers 106 and 107 to encoder 109 via paths 125 and 124, respectively.

Searcher 106 functions by accessing sets of excitation information stored in adaptive codebook 104 and utilizing each set of information to minimize an error criterion between the target excitation received via path 123 and the accessed set of excitation from codebook 104. A scaling factor is also calculated for each accessed set of information since the information stored in adaptive codebook 104 does not allow for the changes in dynamic range of human speech.

The error criterion used is the square of the difference between the original and synthetic speech. The synthetic speech is that which will be reproduced in the synthesizer portion of FIG. 1 on the output of LPC filter 117. The synthetic speech is calculated in terms of the synthetic excitation information obtained from codebook 104 and the ringing signal; and the speech signal is calculated from the target excitation and the

ringing signal. The excitation information for synthetic speech is utilized by performing a convolution of the LPC filter as determined by analyzer 102 utilizing the weighting information from calculator 103 expressed as a matrix. The error criterion is evaluated for each set of information obtained from codebook 104, and the set of excitation information giving the lowest error value is the set of information utilized for the present frame.

After searcher 106 has determined the set of excitation information to be utilized along with the scaling factor, the index into the codebook and the scaling factor are transmitted to encoder 109 via path 125, and the excitation information is also transmitted via path 127 to stochastic searcher 107. Stochastic searcher 107 subtracts the excitation information from adaptive searcher 106 from the target excitation received via path 123. Stochastic searcher 107 then performs operations similar to those performed by adaptive searcher 106.

The excitation information in adaptive codebook 104 is excitation information from previous frames. For each frame, the excitation information consists of the same number of samples as the sampled original speech. Advantageously, the excitation information may consist of 55 samples for a 4.8 Kbps transmission rate. The codebook is organized as a push down list so that the new set of samples are simply pushed into the codebook replacing the earliest samples presently in the codebook. When utilizing sets of excitation information out of codebook 104, searcher 106 does not treat these sets of information as disjoint sets of samples but rather treats the samples in the codebook as a linear array of excitation samples. For example, searcher 106 will form the first candidate set of information by utilizing sample 1 through samples 55 from codebook 104, and the second set of candidate information by using sample 2 through sample 56 from the codebook. This type of searching a codebook is often referred to as an overlapping codebook.

As this linear searching technique approaches the end of the samples in the codebook there is no longer a full set of information to be utilized. A set of information is also referred to as an excitation vector. At that point, the searcher performs a virtual search. A virtual search involves repeating accessed information from the table into a later portion of the set for which there are no samples in the table. This virtual search technique allows the adaptive searcher 106 to more quickly react to speech transitions such as from an unvoiced region of speech to a voiced region of speech. The reason is that in unvoiced speech regions the excitation is similar to white noise whereas in the voiced regions there is a fundamental frequency. Once a portion of the fundamental frequency has been identified from the codebooks, it is repeated.

FIG. 2 illustrates a portion of excitation samples such as would be stored in codebook 104 but where it is assumed for the sake of illustration that there are only 10 samples per excitation set. Line 201 illustrates that the contents of the codebook and lines 202, 203 and 204 illustrate excitation sets which have been formed utilizing the virtual search technique. The excitation set illustrated in line 202 is formed by searching the codebook starting at sample 205 on line 201. Starting at sample 205, there are only 9 samples in the table, hence, sample 208 is repeated as sample 209 to form the tenth sample of the excitation set illustrated in line 202. Sample 208 of line 202 corresponds to sample 205 of line 201. Line 203 illustrates the excitation set following that illustrated in line 202 which is formed by starting at sample 206 on line 201. Starting at sample 206 there are only 8 samples in the code book, hence, the first 2 samples of line 203 which are grouped as samples 210 are repeated at the end of the excitation set illustrated in line 203 as samples 211. It can be observed by one skilled in the art that if the significant peak illustrated in line 203 was a pitch peak then this pitch has been repeated in samples 210 and 211. Line 204 illustrates the third excitation set formed starting at sample 207 in the codebook. As can be seen, the 3 samples indicated as 212 are repeated at the end of the excitation set illustrated on line 204 as samples 213. It is important to realize that the initial pitch peak which is labeled as 207 in line 201 is a cumulation of the searches performed by searchers 106 and 107 from the previous frame since the contents of codebook 104 are updated at the end of each frame. The statistical searcher 107 would normally arrive first at a pitch peak such as 207 upon entering a voiced region from an unvoiced region.

Stochastic searcher 107 functions in a similar manner as adaptive searcher 106 with the exception that it uses as a target excitation the difference between the target excitation from target excitation calculator 102 and excitation representing the best match found by searcher 106. In addition, search 107 does not perform a virtual search.

A detailed explanation is now given of the analyzer portion of FIG. 1. This explanation is based on matrix and vector mathematics. Target excitation calculator 102 calculates a target excitation vector, t, in the following manner. A speech vector s can be expressed as

$$s = Ht + z .$$

The H matrix is the matrix representation of the all-pole LPC synthesis filter as defined by the LPC coefficients received from LPC analyzer 101 via path 121. The structure of the filter represented by H is described in greater detail later in this section and is part of the subject of this invention. The vector z represents the ringing of the all-pole filter from the excitation received during the previous frame. As was described earlier, vector z is derived from LPC filter 110 and zero-input response filter 111. Calculator 102 and subtracter 112 obtain the vector t representing the target excitation by subtracting vector z from vector s and processing the resulting signal vector through the all-zero LPC analysis filter also derived from the LPC coefficients generated by LPC analyzer 101 and transmitted via path 121. The target excitation vector t is obtained by performing a convolution operation of the all-zero LPC analysis filter, also referred to as a whitening filter, and the difference signal found by subtracting the ringing from the original speech. This convolution is performed using well-known signal processing techniques.

Adaptive searcher 106 searches adaptive codebook 104 to find a candidate excitation vector r that best matches the target excitation vector t. Vector r is also referred to as a set of excitation information. The error criterion used to determine the best match is the square of the difference between the original speech and the synthetic speech. The original speech is given by vector s and the synthetic speech is given by the vector y which is calculated by the following equation:

$$y = HL_i r_i + z,$$

where L_i is a scaling factor.
The error criterion can be written in the following form:

$$e = (Ht + z - HL_i r_i - z)^T (Ht + z - HL_i r_i - z). \quad (1)$$

In the error criterion, the H matrix is modified to emphasis those sections of the spectrum which are perceptually important. This is accomplished through well known pole-bandwidth widing technique. Equation 1 can be rewritten in the following form:

$$e = (t - L_i r_i)^T H^T H (t - L_i r_i). \quad (2)$$

Equation 2 can be further reduced as illustrated in the following:

$$e = t^T H^T Ht + L_i r_i^T H^T H L_i r_i - 2L_i r_i^T H^T Ht. \quad (3)$$

The first term of equation 3 is a constant with respect to any given frame and is dropped from the calculation of the error in determining which r_i vector is to be utilized from codebook 104. For each of the r_i excitation vectors in codebook 104, equation 3 must be solved and the error criterion, e, must be determined so as to chose the r_i vector which has the lowest value of e. Before equation 3 can be solved, the scaling factor, L_i must be determined. This is performed in a straight forward manner by taking the partial derivative with respect to L_i and setting it equal to zero, which yields the following equation:

$$L_i = \frac{r_i^T H^T Ht}{r_i^T H^T Hr_i}. \quad (4)$$

The numerator of equation 4 is normally referred to as the cross-correlation term and the denominator is referred to as the energy term. The energy term requires more computation than the cross-correlation term. The reason is that in the cross-correlation terms the product of the last three elements needs only to be calculated once per frame yielding a vector, and then for each new candidate vector, r_i, it is simply necessary to take the dot product between the candidate vector transposed and the constant vector resulting from the computation of the last three elements of the cross-correlation term.

The energy term involves first calculating Hr_i then taking the transpose of this and then taking the inner product between the transpose of Hr_i and Hr_i. This results in a large number of matrix and vector operations requiring a large number of calculations. The following technique reduces the number of calculations and enhances the resulting synthetic speech.

In part, the technique realizes this goal by utilizing a finite impulse response LPC filter rather than an infinite impulse response LPC filter as utilized in the prior art. The utilization of a finite impulse response

filter having a constant response length results in the H matrix having a different symmetry than in the prior art. The H matrix represents the operation of the finite impulse response filter in terms of matrix notation. Since the filter is a finite impulse response filter, the convolution of this filter and the excitation information represented by each candidate vector, r_i, results in each sample of the vector r_i generating a finite number of response samples which are designated as R number of samples. When the matrix vector operation of calculating Hr_i is performed which is a convolution operation, all of the R response points resulting from each sample in the candidate vector, r_i, are summed together to form a frame of synthetic speech.

The H matrix representing the finite impulse response filter is an N + R by N matrix, where N is the frame length in samples, and R is the length of the truncated impulse response in number of samples. Using this form of the H matrix, the response vector Hr has a length of N + R. This form of H matrix is illustrated in the following equation 5:

$$H = \begin{bmatrix} h_0 & 0 & . & . & 0 \\ h_1 & h_0 & . & . & . \\ . & . & . & . & . \\ h_R & h_{R-1} & . & . & . \\ 0 & h_R & . & . & h_0 \\ . & 0 & . & . & h_1 \\ . & . & . & . & . \\ . & . & . & h_R & h_{R-1} \\ 0 & 0 & . & 0 & h_R \end{bmatrix} . \quad (5)$$

Consider the product of the transpose of the H matrix and the H matrix itself as in equation 6:

$$A = H^T H . \quad (6)$$

Equation 6 results in a matrix A which is N by N square, symmetric, and Toeplitz as illustrated in the following equation 7.

$$A = \begin{bmatrix} A_0 & A_1 & A_2 & A_3 & A_4 \\ A_1 & A_0 & A_1 & A_2 & A_3 \\ A_2 & A_1 & A_0 & A_1 & A_2 \\ A_3 & A_2 & A_1 & A_0 & A_1 \\ A_4 & A_3 & A_2 & A_1 & A_0 \end{bmatrix} . \quad (7)$$

Equation 7 illustrates the A matrix which results from H^TH operation when N is five. One skilled in the art would observe from equation 5 that depending on the value of R that certain of the elements in matrix A would be 0. For example, if R = 2 then elements A_2, A_3 and A_4 would be 0.

FIG. 3 illustrates what the energy term would be for the first candidate vector r_1 assuming that this vector contains 5 samples which means that N equals 5. The samples X_0 through X_4 are the first 5 samples stored in adaptive codebook 104. The calculation of the energy term of equation 4 for the second candidate vector r_2 is illustrated in FIG. 4. The latter figure illustrates that only the candidate vector has changed and that it has only changed by the deletion of the X_0 sample and the addition of the X_5 sample.

The calculation of the energy term illustrated in FIG. 3 results in a scalar value. This scalar value for r_1 differs from that for candidate vector r_2 as illustrated in FIG. 4 only by the addition of the X_5 sample and the deletion of the X_0 sample. Because of the symmetry and Toeplitz nature introduced into the A matrix due to the utilization of a finite impulse response filter, the scalar value for FIG. 4 can be easily calculated in the following manner. First, the contribution due to the X_0 sample is eliminated by realizing that its contribution is easily determinable as illustrated in FIG. 5. This contribution can be removed since it is simply based on

the multiplication and summation operations involving term 501 with terms 502 and the operations involving terms 504 with terms 503. Similarly, FIG. 6 illustrates that the addition of term X_5 can be added into the scalar value by realizing that its contribution is due to the operations involving term 601 with terms 602 and the operations involving terms 604 with the terms 603. By subtracting the contribution of the terms indicated in FIG. 5 and adding the effect of the terms illustrated in FIG. 6, the energy term for FIG. 4 can be recursively calculated from the energy term of FIG. 3.

This method of recursive calculation is independent of the size of the vector r_1 or the A matrix. These recursive calculations allow the candidate vectors contained within adaptive codebook 104 or codebook 105 to be compared with each other but only requiring the additional operations illustrated by FIGS. 5 and 6 as each new excitation vector is taken from the codebook.

In general terms, these recursive calculations can be mathematically expressed in the following manner. First, a set of masking matrices is defined as I_k where the last one appears in the kth row.

$$I_k = \begin{bmatrix} 1 & 0 & . & . & . & 0 \\ 0 & 1 & . & . & . & . \\ . & . & . & . & . & . \\ . & . & . & 1 & 0 & . \\ . & . & . & 0 & 0 & . \\ . & . & . & . & . & . \\ 0 & . & . & . & . & 0 \end{bmatrix}. \qquad (8)$$

In addition, the unity matrix is defined as I as follows:

$$I = \begin{bmatrix} 1 & 0 & . & . & . & . \\ 0 & 1 & 0 & . & . & . \\ . & 0 & 1 & 0 & . & . \\ . & . & 0 & 1 & 0 & . \\ . & . & . & 0 & 1 & 0 \\ 0 & . & . & . & 0 & 1 \end{bmatrix}. \qquad (9)$$

Further, a shifting matrix is defined as follows:

$$S = \begin{bmatrix} 0 & 1 & 0 & . & . & 0 \\ 0 & 0 & 1 & . & . & . \\ . & . & . & . & . & . \\ . & . & . & . & 0 & 1 \\ 0 & . & . & . & 0 & 0 \end{bmatrix}. \qquad (10)$$

For Toeplitz matrices, the following well known theorem holds:

$$S^T AS = (I-I_1)\, A\, (I-I_1). \qquad (11)$$

Since A or H^TH is Toeplitz, the recursive calculation for the energy term can be expressed using the following nomenclature. First, define the energy term associated with the r_{j+1} vector as E_{j+1} as follows:

$$E_{j+1} = r_{j+1}^T H^T H r_{j+1} \quad . \quad (12)$$

In addition, vector r_{j+1} can be expressed as a shifted version of r_j combined with a vector containing the new sample of r_{j+1} as follows:

$$r_{j+1} = Sr_j + (I-I_{N-1})\, r_{j+1} \, . \quad (13)$$

Utilizing the theorem of equation 11 to eliminate the shift matrix S allows equation 12 to be rewritten in the following form:

$$E_{j+1} = E_j + 2\left[r_{j+1}^T (I-I_{N-1}) H^T H S r_j - r_j^T (I-I_1) H^T H I_1 r_j\right]$$
$$- r_j^T I_1 H^T H I_1 r_j + r_{j+1}^T (I-I_{N-1}) H^T H (I-I_{N-1})\, r_{j+1} \quad .(14)$$

It can be observed from equation 14, that since the I and S matrices contain predominantly zeros with a certain number of ones that the number of calculations necessary to evaluate equation 14 is greatly reduced from that necessary to evaluate equation 3. A detailed analysis indicates that the calculation of equation 14 requires only 2Q + 4 floating point operations, where Q is the smaller of the number R or the number N. This is a large reduction in the number of calculations from that required for equation 3. This reduction in calculation is accomplished by utilizing a finite impulse response filter rather than an infinite impulse response filter and by the Toeplitz nature of the H^tH matrix.

Equation 14 properly computes the energy term during the normal search of codebook 104. However, once the virtual searching commences, equation 14 no longer would correctly calculate the energy term since the virtual samples as illustrated by samples 213 on line 204 of FIG. 2 are changing at twice the rate. In addition, the samples of the normal search illustrated by samples 214 of FIG. 2 are also changing in the middle of the excitation vector. This situation is resolved in a recursive manner by allowing the actual samples in the codebook, such as samples 214, to be designated by the vector w_i and those of the virtual section, such as samples 213 of FIG. 2, to be denoted by the vector v_i. In addition, the virtual samples are restricted to less than half of the total excitation vector. The energy term can be rewritten from equation 14 utilizing these conditions as follows:

$$E_i = w_i^T H^T H w_i + 2v_i^T H^T H w_i + v_i^T H^T H v_i \, . \quad (15)$$

The first and third terms of equation 15 can be computationally reduced in the following manner. The recursion for the first term of equation 15 can be written as:

$$w_{j+1}^T H^T H w_{j+1} = w_j^T H^T H w_j - 2w_j^T (I-I_1) H^T H I_1 w_j - w_j^T I_1 H^T H I_1 w_j \quad ;(16)$$

and the relationship between v_j and v_{j+1} can be written as follows:

$$v_{j+1} = S^2 (I-I_{p+1})\, v_j + (I-I_{N-2})\, v_{j+1} \, . \quad (17)$$

This allows the third term of equation 15 to be reduced by using the following:

$$H^T H v_{j+1} = S^2 H^T H v_j + H^T H S^2 (I_p - I_{p+1})\, v_j + (I-I_{N-2})\, H^T H S^2 (I-I_{p+1}) v_j + H^T H (I-I_{N-2}) v_{j+1} . \quad (18)$$

The variable p is the number of samples that actually exists in the codebook 104 that are presently used in the existing excitation vector. An example of the number of samples is that given by samples 214 in FIG. 2. The second term of equation 15 can also be reduced by equation 18 since $v_i^T H^T H$ is simply the purpose of $H^T H v_i$ in matrix arithmetic.

The rate at which searching is done through the actual codebook samples and the virtual samples is different. In the above illustrated example, the virtual samples are searched at twice the rate of actual

samples.

FIG. 7 illustrates adaptive searcher 106 of FIG. 1 in greater detail. As previously described, adaptive searcher 106 performs two types of search operations: virtual and sequential. During the sequential search operation, searcher 106 accesses a complete candidate excitation vector from adaptive codebook 104; whereas, during a virtual search, adaptive searcher 106 accesses a partial candidate excitation vector from codebook 104 and repeats the first portion of the candidate vector accessed from codebook 104 into the latter portion of the candidate excitation vector as illustrated in FIG. 2. The virtual search operations are performed by blocks 708 through 712, and the sequential search operations are performed by blocks 702 through 706. Search determinator 701 determines whether a virtual or a sequential search is to be performed. Candidate selector 714 determines whether the codebook has been completely searched; and if the codebook has not been completely searched, selector 714 returns control back to search determinator 701.

Search determinator 701 is responsive to the spectral weighting matrix received via path 122 and the target excitation vector received path 123 to control the complete search codebook 104. The first group of candidate vectors are filled entirely from the codebook 104 and the necessary calculations are performed by blocks 702 through 706, and the second group of candidate excitation vectors are handled by blocks 708 through 712 with portions of vectors being repeated.

If the first group of candidate excitation vectors is being accessed from codebook 104, search determinator communicates the target excitation vector, spectral weighting matrix, and index of the candidate excitation vector to be accessed to sequential search control 702 via path 727. The latter control is responsive to the candidate vector index to access codebook 104. The sequential search control 702 then transfers the target excitation vector, the spectral weighting matrix, index, and the candidate excitation vector to blocks 703 and 704 via path 728.

Block 704 is responsive to the first candidate excitation vector received via path 728 to calculate a temporary vector equal to the H^THt term of equation 3 and transfers this temporary vector and information received via path 728 to cross-correlation calculator 705 via path 729. After the first candidate vector, block 704 just communicates information received on path 728 to path 729. Calculator 705 calculates the cross-correlation term of equation 3.

Energy calculator 703 is responsive to the information on path 728 to calculate the energy term of equation 3 by performing the operations indicated by equation 14. Calculator 703 transfers this value to error calculator 706 via path 733.

Error calculator 706 is responsive to the information received via paths 730 and 733 to calculate the error value by adding the energy value and the cross-correlation value and to transfer that error value along with the candidate number, scaling factor, and candidate value to candidate selector 714 via path 730.

Candidate selector 714 is responsive to the information received via path 732 to retain the information of the candidate whose error value is the lowest and to return control to search determinator 701 via path 731 when actuated via path 732.

When search determinator 701 determines that the second group of candidate vectors is to be accessed from codebook 104, it transfers the target excitation vector, spectral weighting matrix, and candidate excitation vector index to virtual search control 708 via path 720. The latter search control accesses codebook 104 and transfers the accessed code excitation vector and information received via path 720 to blocks 709 and 710 via path 721. Blocks 710, 711 and 712, via paths 722 and 723, perform the same type of operations as performed by blocks 704, 705 and 706. Block 709 performs the operation of evaluating the energy term of equation 3 as does block 703; however, block 709 utilizes equation 15 rather than equation 14 as utilized by energy calculator 703.

For each candidate vector index, scaling factor, candidate vector, and error value received via path 724, candidate selector 714 retains the candidate vector, scaling factor, and the index of the vector having the lowest error value. After all of the candidate vectors have been processed, candidate selector 714 then transfers the index and scaling factor of the selected candidate vector which has the lowest error value to encoder 109 via path 125 and the selected excitation vector via path 127 to adder 108 and stochastic searcher 107 via path 127.

FIG. 8 illustrates, in greater detail, virtual search control 708. Adaptive codebook accessor 801 is responsive to the candidate index received via path 720 to access codebook 104 and to transfer the accessed candidate excitation vector and information received via path 720 to sample repeater 802 via path 803. Sample repeater 802 is responsive to the candidate vector to repeat the first portion of the candidate vector into the last portion of the candidate vector in order to obtain a complete candidate excitation vector which is then transferred via path 721 to blocks 709 and 710 of FIG. 7.

FIG. 9 illustrates, in greater detail, the operation of energy calculator 709 in performing the operations

indicated by equation 18. Actual energy component calculator 901 performs the operations required by the first term of equation 18 and transfers the results to adder 905 via path 911. Temporary virtual vector calculator 902 calculates the term H^THv_i in accordance with equation 18 and transfers the results along with the information received via path 721 to calculators 903 and 904 via path 910. In response to the information on path 910, mixed energy component calculator 903 performs the operations required by the second term of equation 15 and transfers the results to adder 905 via path 913. In response to the information on path 910, virtual energy component calculator 904 performs the operations required by the third term of equation 15. Adder 905 is responsive to information on paths 911, 912, and 913 to calculate the energy value and to communicate that value on path 726.

Stochastic searcher 107 comprises blocks similar to blocks 701 through 706 and 714 as illustrated in FIG. 7. However, the equivalent search determinator 701 would form a second target excitation vector by subtracting the selected candidate excitation vector received via path 127 from the target excitation received via path 123. In addition, the determinator would always transfer control to the equivalent control 702.

Claims

1. A method of encoding speech based on determining sets of filter coefficients and corresponding excitation frames, said speech comprising frames each having a plurality of samples, comprising the steps of
determining (101) a set of filter coefficients of a filter in response to a present one of said frames of speech;
forming (102) a first excitation frame in response to the said present one of said frames of speech;
calculating (104, 106) an error value for each one of a plurality of candidate excitation frames stored in an adaptive code book in response to the said first excitation frame including forming virtual candidate excitation frames by repeating a first portion of each of a group of said candidate excitation frames at a second portion of said each of said group of said candidate excitation frames;
communicating (109) said filter coefficients and information defining the location of the candidate excitation frame selected as having the smallest error value in said adaptive code book to a decoder for reproduction of speech for the present speech frame, the said location defining information enabling the decoder to identify and itself form a virtual candidate excitation frame when said selected candidate excitation frame is a virtual excitation frame for the said present speech frame.

2. The method of Claim 1 further characterised in that said step of calculating an error value comprises the steps of: storing (104) an array of samples in said adaptive code book; shifting (801) a window equal to the number of samples in said present speech frame to form each of said candidate excitation frames thereby creating candidate excitation frames of said group for each of which there are not samples in said array to fill the second portion of each of said excitation frames of said group; and repeating (802) said first portion of each of said group of said candidate excitation frames in said second portion in each of said candidate excitation frames to complete each of said group of candidate excitation frames.

3. The method of Claim 2 further characterised in that candidate excitation frames other than those contained in said group of said candidate excitation frames are filled entirely with samples accessed sequentially from said adaptive code book.

4. The method of Claim 3 characterised by further comprising the steps of:
calculating a temporary excitation frame from said first excitation frame and the selected excitation frame;
calculating (101) a set of filter coefficients in response to a present one of said speech frames;
calculating (103) a spectral weighting matrix of a Toeplitz form to model a finite impulse response filter based on said filter coefficients for said present speech frame;
calculating (705) a cross-correlation value in response to said temporary excitation frame and said spectral weighting matrix and each of a plurality of other candidate excitation frames stored in a stochastic code book;
recursively calculating (703) an energy value for each of said other candidate excitation frames in response to said temporary excitation frame and said spectral weighting matrix and each of said other candidate excitation frames;

indicated by equation 18. Actual energy component calculator 901 performs the operations required by the first term of equation 18 and transfers the results to adder 905 via path 911. Temporary virtual vector calculator 902 calculates the term H^THv_i in accordance with equation 18 and transfers the results along with the information received via path 721 to calculators 903 and 904 via path 910. In response to the information on path 910, mixed energy component calculator 903 performs the operations required by the second term of equation 15 and transfers the results to adder 905 via path 913. In response to the information on path 910, virtual energy component calculator 904 performs the operations required by the third term of equation 15. Adder 905 is responsive to information on paths 911, 912, and 913 to calculate the energy value and to communicate that value on path 726.

Stochastic searcher 107 comprises blocks similar to blocks 701 through 706 and 714 as illustrated in FIG. 7. However, the equivalent search determinator 701 would form a second target excitation vector by subtracting the selected candidate excitation vector received via path 127 from the target excitation received via path 123. In addition, the determinator would always transfer control to the equivalent control 702.

Claims

1. A method of encoding speech based on determining sets of filter coefficients and corresponding excitation frames, said speech comprising frames each having a plurality of samples, comprising the steps of
determining (101) a set of filter coefficients of a filter in response to a present one of said frames of speech;
forming (102) a first excitation frame in response to the said present one of said frames of speech;
calculating (104, 106) an error value for each one of a plurality of candidate excitation frames stored in an adaptive code book in response to the said first excitation frame including forming virtual candidate excitation frames by repeating a first portion of each of a group of said candidate excitation frames at a second portion of said each of said group of said candidate excitation frames;
communicating (109) said filter coefficients and information defining the location of the candidate excitation frame selected as having the smallest error value in said adaptive code book to a decoder for reproduction of speech for the present speech frame, the said location defining information enabling the decoder to identify and itself form a virtual candidate excitation frame when said selected candidate excitation frame is a virtual excitation frame for the said present speech frame.

2. The method of Claim 1 further characterised in that said step of calculating an error value comprises the steps of: storing (104) an array of samples in said adaptive code book; shifting (801) a window equal to the number of samples in said present speech frame to form each of said candidate excitation frames thereby creating candidate excitation frames of said group for each of which there are not samples in said array to fill the second portion of each of said excitation frames of said group; and repeating (802) said first portion of each of said group of said candidate excitation frames in said second portion in each of said candidate excitation frames to complete each of said group of candidate excitation frames.

3. The method of Claim 2 further characterised in that candidate excitation frames other than those contained in said group of said candidate excitation frames are filled entirely with samples accessed sequentially from said adaptive code book.

4. The method of Claim 3 characterised by further comprising the steps of:
calculating a temporary excitation frame from said first excitation frame and the selected excitation frame;
calculating (101) a set of filter coefficients in response to a present one of said speech frames;
calculating (103) a spectral weighting matrix of a Toeplitz form to model a finite impulse response filter based on said filter coefficients for said present speech frame;
calculating (705) a cross-correlation value in response to said temporary excitation frame and said spectral weighting matrix and each of a plurality of other candidate excitation frames stored in a stochastic code book;
recursively calculating (703) an energy value for each of said other candidate excitation frames in response to said temporary excitation frame and said spectral weighting matrix and each of said other candidate excitation frames;

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calculating (706) an error value for each of said other candidate excitation frames in response to each of said cross-correlation and energy values for each of said other candidate excitation frames and selecting (714) the other candidate excitation frame having the smallest error value; and in that said communicating step further communicates the location of the selected other candidate excitation frame in said stochastic code book for reproduction of said speech for said present speech frame.

5. An apparatus for encoding speech based on determining sets of filter coefficients and corresponding excitation frames, said speech comprising frames each having a plurality of samples, comprising
means (101) for determining a set of filter coefficients of a filter in response to a present one of said frames of speech;
means (102) for forming a first excitation frame in response to the said present one of said frames of speech;
means (104, 106) for calculating an error value for each one of a plurality of candidate excitation frames stored in an adaptive code book in response to the said first excitation frame, including forming virtual candidate excitation frames by repeating a first portion of each of a group of said candidate excitation frames at a second portion of said each of said group of said candidate excitation frames;
means (109) for communicating said filter coefficients and information defining the location of the candidate excitation frame selected as having the smallest error value in said adaptive code book to a decoder for reproduction of speech for the present speech frame, the said location defining information enabling the decoder to identify and itself form a virtual candidate excitation frame when said selected candidate excitation frame is a virtual excitation frame for the said present speech frame.

6. The apparatus of Claim 5 wherein the calculating means includes means (104) for storing said candidate excitation frames in said adaptive code book as a linear array of samples;
means (801) for shifting a window equal to number of samples in each candidate excitation frame to form each candidate excitation frame thereby creating candidate excitation frames of said group of said candidate excitation frames for each of which there are not samples in said array to fill the second portion of each of said candidate excitation frames of said group of said excitation frames; and
means (802) for repeating said first portion of each of said group of candidate excitation frames in said second portion of each of said group of said candidate excitation frames to complete each of said group of said candidate excitation frames.

7. The apparatus of Claim 6 further including means (103) for calculating information representing a finite impulse response filter from said set of filter coefficients;
means (708, 709, 710, 711, 712) for recursively calculating an error value for each of said plurality of candidate excitation frames stored in said adaptive code book in response to the finite impulse response filter information and each of said candidate excitation frames and said first excitation frame; and
means (714) for selecting said best one of said candidate excitation frame that has the smallest error value.

Patentansprüche

1. Verfahren zur Kodierung von Sprache unter Bestimmung von Sätzen von Filterkoeffizienten und entsprechenden Erregungsrahmen, wobei die Sprache Rahmen mit je einer Vielzahl von Abtastwerten umfaßt, mit den Schritten:
Bestimmen (101) eines Satzes von Filterkoeffizienten eines Filters unter Ansprechen auf einen augenblicklichen Sprachrahmen,
Bilden (102) eines ersten Erregungsrahmens unter Ansprechen auf den augenblicklichen Sprachrahmen,
Berechnen (104, 106) eines Fehlerwertes für jeden von einer Vielzahl von in einem adaptiven Kodebuch gespeicherten Anwärter-Erregungsrahmen unter Ansprechen auf den ersten Erregungsrahmen einschließlich einer Bildung virtueller Anwärter-Erregungsrahmen durch Wiederholen eines ersten Teils jeder Gruppe der Anwärter-Erregungsrahmen bei einem zweiten Teil jeder Gruppe der Anwärter-Erregungsrahmen,
Übertragen (109) der Filterkoeffizienten und von Informationen, die den Ort des Anwärter-Erregungsrahmens, der als der Rahmen mit dem kleinsten Fehlerwert in dem adaptiven Kodebuch gewählt worden ist, definieren, an einen Dekoder zur Reproduktion der Sprache für den augenblicklichen Sprachrah-

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men, wobei die den Ort definierenden Informationen den Dekoder in die Lage versetzen, einen virtuellen Anwärter-Erregungsrahmen zu identifizieren und selbst zu bilden, wenn der gewählte Anwärter-Erregungsrahmen ein virtueller Erregungsrahmen für den augenblicklichen Sprachrahmen ist.

2. Verfahren nach Anspruch 1,
dadurch gekennzeichnet,
daß der Schritt zur Berechnung eines Fehlerwertes folgende Schritte umfaßt:
Speichern (104) einer Anordnung von Abtastwerten in dem adaptiven Kodebuch,
Verschieben (801) eines Fensters gleich der Anzahl von Abtastwerten in dem augenblicklichen Sprachrahmen zur Bildung jedes der Anwärter-Erregungsrahmen und dadurch Erzeugen von Anwärter-Erregungsrahmen der Gruppe, für jeden von denen keine Abtastwerte in der Anordnung vorhanden sind, die den zweiten Teil jedes Erregungsrahmens der Gruppe füllen, und
Wiederholen (802) des ersten Teils jeder Gruppe der Anwärter-Erregungsrahmen im zweiten Teil jedes Anwärter-Erregungsrahmens, um jeden Rahmen der Gruppe von Anwärter-Erregungsrahmen zu vervollständigen.

3. Verfahren nach Anspruch 2,
dadurch gekennzeichnet,
daß Anwärter-Erregungsrahmen, die nicht in der Gruppe von Anwärter-Erregungsrahmen enthalten sind, vollständig mit Abtastwerten gefüllt werden, auf die sequentiell aus dem adaptiven Kodebuch zugegriffen wird.

4. Verfahren nach Anspruch 3,
dadurch gekennzeichnet,
daß ferner folgende Schritte vorgesehen sind:
Berechnen eines zeitweiligen Erregungsrahmens aus dem ersten Erregungsrahmen und dem gewählten Erregungsrahmen,
Berechnen (101) eines Satzes von Filterkoeffizienten unter Ansprechen auf einen augenblicklichen Sprachrahmen,
Berechnen (103) einer spektralen Bewertungsmatrix einer Toeplitz-Form zur Nachbildung eines Filters mit endlichem Impulsansprechen auf der Grundlage der Filterkoeffizienten für den augenblicklichen Sprachrahmen,
Berechnen (705) eines Kreuzkorrelationswertes unter Ansprechen auf den zeitweiligen Erregungsrahmen und die spektrale Bewertungsmatrix sowie jeden von einer Vielzahl von weiteren Anwärter-Erregungsrahmen, die in einer stochastischen Kodebuch gespeichert sind,
rekursives Berechnen (703) eines Energiewertes für jeden der weiteren Anwärter-Erregungsrahmen unter Ansprechen auf den zeitweiligen Erregungsrahmen und die spektrale Bewertungsmatrix sowie jeden der weiteren Anwärter-Erregungsrahmen,
Berechnen (706) eines Fehlerwertes für jeden der weiteren Anwärter-Erregungsrahmen unter Ansprechen auf jeden der Kreuzkorrelations- und Energiewerte für jeden der weiteren Anwärter-Erregungsrahmen und
Auswählen (714) desjenigen weiteren Anwärter-Erregungsrahmens, der den kleinsten Fehlerwert besitzt, und
daß bei dem Übertragungsschritt ferner der Ort des gewählten weiteren Anwärter-Erregungsrahmens in dem stochastischen Kodebuch zur Wiederherstellung der Sprache in dem augenblicklichen Sprachrahmen übertragen wird.

5. Vorrichtung zur Kodierung von Sprache unter Bestimmung von Sätzen von Filterkoeffizienten und entsprechenden Erregungsrahmen, wobei die Sprache Rahmen mit je einer Vielzahl von Abtastwerten umfaßt, mit
einer Einrichtung (101) zur Bestimmung eines Satzes von Filterkoeffizienten eines Filters unter Ansprechen auf einen augenblicklichen Sprachrahmen,
einer Einrichtung (102) zur Bildung eines ersten Erregungsrahmens unter Ansprechen auf den augenblicklichen Sprachrahmen,
eine Einrichtung (104, 106) zur Berechnung eines Fehlerwertes für jeden von einer Vielzahl von in einem adaptiven Kodebuch gespeicherten Anwärter-Erregungsrahmen unter Ansprechen auf den ersten Erregungsrahmen einschließlich Bildung virtueller Anwärter-Erregungsrahmen durch Wiederholen eines ersten Teils jedes von einer Gruppe der Anwärter-Erregungsrahmen bei einem zweiten Teil jedes der

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Gruppe von Anwärter-Erregungsrahmen, und
einer Einrichtung (109) zur Übertragung der Filterkoeffizienten und von Informationen, die den Ort des Anwärter-Erregungsrahmens definieren, der als der Rahmen mit dem kleinsten Fehlerwert in dem adaptiven Kodebuch gewählt worden ist, an einen Dekoder zur Wiederherstellung der Sprache für den augenblicklichen Sprachrahmen, wobei die den Ort definierenden Informationen den Dekoder in die Lage versetzen, einen virtuellen Anwärter-Erregungsrahmen zu identifizieren und selbst zu bilden, wenn der gewählte Anwärter-Erregungsrahmen ein virtueller Erregungsrahmen für den augenblicklichen Sprachrahmen ist.

6. Vorrichtung nach Anspruch 5,
bei der die Berechnungseinrichtung eine Einrichtung (104) zur Speicherung der Anwärter-Erregungsrahmen als lineare Anordnung von Abtastwerten in dem adaptiven Kodebuch enthält, ferner
eine Einrichtung (801) zum Verschieben eines Fensters gleich der Anzahl von Abtastwerten in jedem Anwärter-Erregungsrahmen zwecks Bildung jedes Anwärter-Erregungsrahmens und damit Erzeugen von Anwärter-Erregungsrahmen der Gruppe von Anwärter-Erregungsrahmen, für jeden von denen kleine Abtastwerte in der Anordnung zur Auffüllung des zweiten Teils jedes Anwärter-Erregungsrahmens der Gruppe von Erregungsrahmen vorhanden sind, und
eine Einrichtung (802) zum Wiederholen des ersten Teils jedes Rahmens der Gruppe von Anwärter-Erregungsrahmen im zweiten Teil jedes Rahmens der Gruppe von Anwärter-Erregungsrahmen zur Vervollständigung jedes Rahmens der Gruppe von Anwärter-Erregungsrahmen.

7. Vorrichtung nach Anspruch 6 mit ferner einer Einrichtung (103) zur Berechnung von ein Filter mit endlichem Impulsansprechen darstellenden Informationen aus dem Satz von Filterkoeffizienten,
einer Einrichtung (708, 709, 710, 711, 712) zur rekursiven Berechnung eines Fehlerwertes für jeden der Vielzahl von in dem adaptiven Kodebuch gespeicherten Anwärter-Erregungsrahmen unter Ansprechen auf die das Filter mit endlichem Impulsansprechen definierenden Informationen und jeden der Anwärter-Erregungsrahmen sowie den ersten Erregungsrahmen, und
einer Einrichtung (714) zur Auswahl des besten Anwärter-Erregungsrahmens, der den kleinsten Fehlerwert besitzt.

Revendications

1. Un procédé de codage de la parole basé sur la détermination d'ensembles de coefficients de filtre et de trames d'excitation correspondantes, la parole considérée consistant en trames ayant chacun un ensemble d'échantillons, comprenant les étapes suivantes :
on détermine (101) un ensemble de coefficients d'un filtre sous la dépendance d'une trame présente parmi les trames de parole;
on forme (102) une première trame d'excitation sous la dépendance de la trame présente parmi les trames de parole;
on calcule (104, 106) une valeur d'erreur pour chaque trame parmi un ensemble de trames d'excitation candidates qui sont enregistrées dans un répertoire de codes adaptatif, sous la dépendance de la première trame d'excitation, cette opération comprenant la formation de trames d'excitation candidates virtuelles par la répétition d'une première partie de chaque trame d'un groupe de trames d'excitation candidates, dans une seconde partie de chaque trame du groupe de trames d'excitation candidates;
on transmet (109) à un décodeur, pour la reproduction de la parole pour la trame de parole présente, les coefficients de filtre et l'information définissant la position de la trame d'excitation candidate qui est sélectionnée comme ayant la plus faible valeur d'erreur dans le répertoire de codes adaptatif, l'information qui définit la position permettant au décodeur d'identifier et de former lui-même une trame d'excitation candidate virtuelle lorsque la trame d'excitation candidate sélectionnée est une trame d'excitation virtuelle pour la trame de parole présente.

2. Le procédé de la revendication 1, caractérisé en outre en ce que l'étape de calcul d'une valeur d'erreur comprend les étapes suivantes : on enregistre (104) un tableau d'échantillons dans le répertoire de codes adaptatif; on décale (801) une fenêtre égale au nombre d'échantillons dans la trame de parole présente, pour former chacune des trames d'excitation candidates, de façon à créer ainsi des trames d'excitation candidates du groupe pour chacune desquelles le tableau ne contient pas d'échantillons permettant de remplir la seconde partie de chacune des trames d'excitation du groupe; et on répète

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(802) la première partie de chaque trame du groupe de trames d'excitation candidates, dans la seconde partie de chacune des trames d'excitation candidates, pour compléter chaque trame du groupe de trames d'excitation candidates.

3. Le procédé de la revendication 2, caractérisé en outre en ce que des trames d'excitation candidates autres que celles qui sont contenues dans le groupe précité de trames d'excitation candidates, sont entièrement remplies avec des échantillons auxquels on accède séquentiellement dans le répertoire de codes adaptatif.

4. Le procédé de la revendication 3, caractérisé en ce qu'il comprend en outre les étapes suivantes :

on calcule une trame d'excitation temporaire à partir de la première trame d'excitation et de la trame d'excitation sélectionnée;

on calcule (101) un ensemble de coefficients de filtre sous la dépendance de l'une présente des trames de parole;

on calcule (103) une matrice de pondération spectrale d'une forme Toeplitz pour modéliser un filtre à réponse impulsionnelle finie basé sur les coefficients de filtre pour la trame de parole présente;

on calcule (705) une valeur d'intercorrélation sous la dépendance de la trame d'excitation temporaire et de la matrice de pondération spectrale et de chaque trame parmi un ensemble d'autres trames d'excitation candidates qui sont enregistrées dans un répertoire de codes stochastiques;

on calcule de façon récursive (703) une valeur d'énergie pour chacune des autres trames d'excitation candidates, sous la dépendance de la trame d'excitation temporaire et de la matrice de pondération spectrale, et de chacune des autres trames d'excitation candidates;

on calcule (706) une valeur d'erreur pour chacune des autres trames d'excitation candidates, sous la dépendance de chacune des valeurs d'intercorrélation et d'énergie pour chacune des autres trames d'excitation candidates; et

on sélectionne (714) l'autre trame d'excitation candidate ayant la plus faible valeur d'erreur; et en ce que

l'étape de transmission comprend en outre la transmission de la position de l'autre trame d'excitation sélectionnée dans le répertoire de codes stochastiques, pour la reproduction de la parole pour la trame de parole présente.

5. Un appareil pour le codage de la parole basé sur la détermination d'ensembles de coefficients de filtre et de trames d'excitation correspondantes, la parole considérée comprenant des trames ayant chacun un ensemble d'échantillons, comprenant :

des moyens (101) destinés à déterminer un ensemble de coefficients d'un filtre sous la dépendance de l'une présente des trames de parole;

des moyens (102) destinés à former une première trame d'excitation sous la dépendance de la trame présente parmi les trames de parole;

des moyens (104, 106) destinés à calculer une valeur d'erreur pour chaque trame parmi un ensemble de trames d'excitation candidates qui sont enregistrées dans un répertoire de codes adaptatif, sous la dépendance de la première trame d'excitation, cette opération comprenant la formation de trames d'excitation candidates virtuelles par la répétition d'une première partie de chaque trame d'un groupe des trames d'excitation candidates, dans une seconde partie de chaque trame de ce groupe de trames d'excitation candidates;

des moyens (109) destinés à transmettre à un décodeur, pour la reproduction de la parole pour la trame de parole présente, les coefficients de filtre et l'information définissant la position de la trame d'excitation candidate sélectionnée comme ayant la plus faible valeur d'erreur dans le répertoire de codes adaptatif, l'information qui définit la position permettant au décodeur d'identifier et de former lui-même une trame d'excitation candidate virtuelle lorsque la trame d'excitation candidate sélectionnée est une trame d'excitation virtuelle pour la trame de parole présente.

6. L'appareil de la revendication 5, dans lequel les moyens de calcul comprennent des moyens (104) qui sont destinés à enregistrer les trames d'excitation candidates dans le répertoire de codes adaptatif sous la forme d'un tableau linéaire d'échantillons;

des moyens (801) destinés à décaler une fenêtre égale au nombre d'échantillons dans chaque trame d'excitation candidate, pour former chaque trame d'excitation candidate et pour créer ainsi des trames d'excitation candidates du groupe de trames d'excitation candidates pour chacune desquelles le tableau ne contient pas d'échantillons permettant de remplir la seconde partie de chacune des trames

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d'excitation candidates du groupe de trames d'excitation candidates; et

des moyens (802) destinés à répéter la première partie de chaque trame du groupe de trames d'excitation candidates, dans la seconde partie de chaque trame du groupe de trames d'excitation candidates, pour compléter chaque groupe de trames d'excitation candidates.

7. L'appareil de la revendication 6, comprenant en outre des moyens (103) destinés à calculer une information représentant un filtre à réponse impulsionnelle finie, à partir de l'ensemble de coefficients de filtre,

des moyens (708, 709, 710, 711, 712) destinés à calculer de façon récursive une valeur d'erreur pour chaque trame de l'ensemble de trames d'excitation candidates qui sont enregistrées dans le répertoire de codes adaptatif, sous la dépendance de l'information de filtre à réponse impulsionnelle finie, et de chacune des trames d'excitation candidates et de la première trame d'excitation; et

des moyens (714) destinés à sélectionner la meilleure trame d'excitation candidate qui a la plus faible valeur d'erreur.

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