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(54) **Method and apparatus for coding digital signals.**

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Description

The invention relates to a method and apparatus for coding of digital signals and is especially, but not exclusively, applicable to coding of voice frequency signals to reduce bit rates for reduced transmission or storage requirements.

In particular, the invention concerns coding systems in which the input signal is encoded in discrete blocks. For each block an excitation signal is derived which, when applied to a synthesis filter having suitable coefficients, will generate an approximation to the input signal.

In one type of coding system, specifically the vocoder, the excitation signal is a train of pulses having a period corresponding to the fundamental periodicity of the input signal. This is not entirely satisfactory because the decoded signal from the synthesis filter is not usually an accurate reproduction of the input signal.

Better accuracy can be achieved using so-called "waveform" coding, in which the derivation of the excitation signal takes into account the waveform of the input signal. One such waveform coding system is disclosed by B.S. Atal and J. R. Remde in a paper entitled "A New Model of LPC Excitation for Producing Natural-Sounding Speech at Low Bit Rates". Proceedings of the International Conference on Acoustics, Speech and Signal Processing, Paris, May 1982, pp 614—617. In their system, the excitation signal comprises a set of pulses having amplitudes and locations in the particular block determined according to the signal waveform. Each set of pulses is derived using a linear predictive coding (LPC) filter which has reflection coefficients derived from the input signal. Each excitation pulse is derived by calculating the filter response for every possible pulse position and selecting the one which gives least weighted error between such response and the input signal. The weighting of the error exploits the properties of human auditory perception. Naturally, the calculation of the excitation pulses can only begin when the LPC coefficients for the block are available. Consequently, there is a delay, equal to at least twice the period or frame over which the LPC coefficients are derived, between the synthesizer output and the encoder input.

To achieve adequate accuracy it is preferable, with such known LPC-based waveform coders, to derive the coefficients over a large period of time. Atal and Remde, for example, specify a frame period of about 20 milliseconds, equivalent to several blocks, which in their proposal, are 5 milliseconds long. In a practical environment, however, such as the telephone system, such a delay could be intolerable because of exacerbated echo problems.

According to one aspect of the present invention, apparatus for processing a digital signal comprises an encoder having

input means for providing a signal (S_n) in linear PCM format;

storage means for storing discrete blocks of said linear PCM signal (S_n) individually and successively, each block comprising a predetermined number of samples;

coefficient means for deriving from said signal (S_n) a plurality of sets of reflection coefficients, and

excitation signal generating means for generating an excitation signal (A) corresponding to said block, said excitation signal comprising a component representing amplitude and a component representing temporal location within said block for each of a set of excitation pulses smaller in number than the number of possible PCM samples in said block, characterised in that said plurality of sets of coefficients are updated continuously on a sample by sample basis during arrival of said predetermined number of samples in that block, and in that said excitation signal generating means generates the excitation signal (A) in response to the waveform of the signals (S_n) and the most recently derived set of said plurality of sets of coefficients which has been updated up to the last sample.

The apparatus may further comprise:

a decoder comprising:

means for synchronising the start of each block of the reconstructed output signal with the particular sample period to which the coefficients for that block correspond;

a decoder synthesis filter having an input to receive said excitation signal and coefficient means adjustable in response to the prediction coefficients generated by said encoder, whereby upon application of each set of excitation pulses and corresponding coefficients said synthesis filter produces an output signal substantially reproducing said linear PCM signal S_n.

The primary advantage of this arrangement is that the excitation signal calculations in the encoder are delayed by only the duration of the block. The delay therefore can be much shorter than in the previous known proposals.

The coefficients may be derived on a sample-by-sample or continuous basis and preferably are converted to LPC coefficients.

In preferred embodiments, the coefficients generating means comprises an adaptive lattice. The decoder synthesis filter may then comprise a lattice too, although other filters may also be used, for example, and preferably, a recursive filter with adaptable prediction coefficients.

According to a second aspect of the invention, a method of processing a digital signal comprises encoding the digital signal by:

providing a signal S_n in linear PCM format;

storing discrete blocks of said signal S_n individually and successively, each block comprising a predetermined number of samples;

deriving from said signal S_n a plurality of sets of reflection coefficients, and

generating an excitation signal (A) corresponding to said block, said excitation signal comprising a component representing amplitude and a component representing temporal location within said block for each of a set of excitation pulses less in number than the number of possible PCM samples in said block, characterised in that said plurality of sets of coefficients are updated continuously on a sample by sample basis during arrival of said predetermined number of samples in that block, and in that the excitation signal (A) is generated in response to the waveform of the signal (S_n) and the most recently derived set of the plurality of sets of coefficients which has been updated up to the last sample.

The method may further comprise decoding the output signal by:

deriving from said excitation signal said excitation pulses and applying them to the input of a decoder synthesis filter having adjustable predictor coefficients; and

adjusting said filter predictor coefficients, in response to said coefficient signal;

wherein each said set of excitation pulses is applied to said filter following corresponding adjustment of its coefficients, to produce an output signal substantially identical to the corresponding block of the linear PCM signal.

As mentioned previously, in the system proposed by Atal and Remde the first excitation pulse is derived by calculating the filter response for every possible pulse position in the block and selecting the one which gives the least weighted error when compared with the input signal. The procedure is then repeated, taking into account the contribution of the first excitation pulse, to find the next-best or second excitation pulse, and so on, until the complete set of excitation pulses have been selected. A disadvantage of this approach is that it requires a considerable number of operations, i.e. multiplications and additions for each set of excitation pulses.

In any or all of the aforementioned aspects of the invention the parameters of the set of excitation pulses may be derived by:

(i) computing the impulses response h_n of an encoder synthesis filter using the coefficients a_i appropriate to that block;

(ii) computing cross-correlation α_m between the impulse response h_n and the output of the encoder synthesis filter;

(iii) computing the general covariance $\emptyset$ (i, j) of the impulse response h_n in accordance with the general expression given in the equation 8 of the attached Figure 6. (It should be noted that in this specification covariance $\emptyset$ (i, j) is intended to encompass both covariance and the more limited auto-correlation used in signal processing applications.)

This has the advantage of mitigating the disadvantages described above.

Preferably, the encoder synthesis filter for which aid impulse response is computed is a modified synthesis filter i.e. a synthesis filter modified to take into account perceptual weighting additional to any preemphasis applied to the input signal. The modified synthesis filter may have coefficients $a_i'=\gamma^i \cdot a_i$, where a_i are the prediction coefficients of the synthesis filter in the decoder and γ is a constant between zero and unity, preferably 0.75.

At least for computing the parameters of the first excitation pulse, the covariance and the cross-correlation are used to derive the location of the maximum cross-correlation, which is the position of the first excitation pulse, and the corresponding amplitude of that pulse.

For computing the parameters of subsequent pulses of each set, the cross-correlation signal from the preceding pulse computation is used, having been stored in a buffer. The general covariance matrix vector corresponding to the pulse location is multiplied by the corresponding amplitude and subtracted from the stored preceding cross-correlation signal. The location of the maximum value of the difference α'_m and corresponding pulse amplitude, are derived as before to give the parameters of the instant excitation pulse.

More specifically the general covariance signal may be derived by:

computing the covariance matrix in accordance with equation 9 of Figure 6;

squaring the cross-correlation signal α_m (or the difference α'_m); and

dividing the squared cross-correlation signal by the diagonal $\emptyset_{(m,m)}$ of the covariance matrix by the squared cross-correlation signal.

The general covariance matrix vector is then the matrix row corresponding to that of the maximum cross-correlation value.

Alternatively, and preferably, the general covariance signal is the output of an auto-correlator. The auto-correlation signal is used directly in computing the amplitude of the first excitation pulse. For computing said subsequent pulses, the difference signal is derived by subtracting from the preceding value the product of the amplitude and the auto-correlation signal $R_h(i)$.

An embodiment of the invention will now be described by way of example only and with reference to the accompanying drawings, in which:

Figures 1 and 2 are schematic representations of an encoder and a decoder, which are shown included in a transmitter and receiver, respectively, interconnected by a communications channel.

Figure 3 is a more detailed diagram of the encoder/transmitter of Figure 1.

Figure 4 is a schematic representation of an adaptive lattice which is part of the encoder;

Figure 5 is a schematic representation, corresponding to Figure 3, of a modified encoder; and

Figure 6 is a table of equations referred to in this description.

Referring to Figure 1, the transmitter/encoder comprises input means 10, to which an analogue speech input signal is applied and which provides a corresponding signal S_n in linear PCM format. Each sample of the linear PCM formatted signal S_n comprises 16 bits. This signal S_n is applied to storage means 12 and coefficient signal generating means 14, respectively. The storage means 12 comprises a buffer which receives the signal S_n serially and stores it in blocks of 32 samples. Each block has a duration of 4 ms. The contents of the buffer 12 are accessed by excitation signal generating means 16 which uses them to generate an excitation signal (A), which is applied to a multiplexer 18. This excitation signal (A) comprises a series of bits representing the amplitude and location of each of a set of pulses which, applied to a synthesis filter having suitable coefficients, will regenerate the block of the PCM input signal. The excitation signal (A) also includes a gain factor, (G), which will be explained later.

The coefficient signal generating means 14 derives from the linear PCM signal S_n a coefficient signal representing sets of reflection coefficients K_1—K_8. These coefficients are updated on a sample-by-sample basis. The coefficient signal from the coefficient generating means 14 is applied to the multiplexer 18 and the excitation signal generating means 16, respectively.

Although the coefficient signal is updated continuously, only one set of coefficients K_1—K_8 is used in calculating each set of excitation pulses. In particular, for each 4 mS block the set of coefficients used are those pertaining to the last sample period of that block. In this specific example, therefore, the coefficients are subsampled or extracted every 32 samples coinciding with the end of each block.

The coefficient signal applied to the multiplexer 18 comprises a difference signal, specifically a series of bits which represent the difference between the instant coefficient values and those previously transmitted.

The output of multiplexer 18 is transmitted via channel 20 to the receiver/decoder 22. There the excitation signal *A* and coefficient signal K_i are segregated by means of a demultiplexer 24. The three components of the excitation signal, comprising amplitude component A_m, location component *m*, and gain factor *G*, are applied to decoder input means 26 wherein they are decoded by amplitude decoder 21, location decoder 23, and gain factor or r.m.s. decoder 25, respectively. The outputs of the amplitude decoder 21 and gain or r.m.s. decoder 25 are multiplied together by multiplier 27, the output of which is applied to excitation pulse generator 29. The output of the location decoder 23 is also applied to excitation pulse generator 29, which produces therefrom a train of excitation pulses P, having the appropriate amplitudes and locations. In each 4 mS period only 8 pulses will be generated. It has been found that a mere 8 pulses are sufficient to reconstruct accurately each 32 sample block of the input signal.

The excitation pulses are applied to the input of a synthesis filter 28, which has adjustable predictor coefficients a_1—a_8.

The coefficient signal K_i from the demultiplexer 24 is applied to coefficient decoding means 30 which produces therefrom the reflection coefficients K_1—K_8.

These are converted by conversion means 32 into predictor coefficient values which are applied to the synthesis filter, as indicated by line 34, to effect corresponding adjustment of its predictor coefficients a_1—a_8.

The adjustment of the synthesis filter coefficients is synchronised to the beginning of the 4 millisecond period during which the corresponding set of excitation pulses are applied to the input of the synthesis filter 28.

The resulting output signal from the synthesis filter 28, following its excitation by the set of excitation pulses, is a close approximation to the corresponding 4 millisecond block of the linear PCM input signal S_n from which the set of excitation pulses were derived.

The apparatus will now be described in more detail with reference also to Figures 3 and 4. The input means 10 comprises sampling means 40 for sampling and digitizing the analogue voice signal into 8-bit μ-law samples, μ-law to linear conversion means 42, and pre-emphasis means 44 for emphasizing the high frequencies for perceptual reasons. It will be appreciated that the input means 10 could be modified readily to accommodate inputs other than analogue speech. For example, the A/D sampling means 40 could be omitted if the input signal were already in PCM form. Likewise, conversion from A-law could be used instead of μ-law if the signal required it, and other weighting could be applied.

The coefficient signal generating means 14 comprises an adaptive lattice 46 and coefficient encoder 48. The lattice 46 is shown in detail in Figure 4 and is of the kind disclosed by J. I. Makhoul and L. K. Cosell in a paper entitled "Adaptive Lattice Analysis of Speech". IEEE Transactions of Acoustics, Speech and Signal Processing, Vol. ASSP-29, No. 3, June 1981. The adaptive lattice illustrated in Figure 4 has eight stages. Its output is discarded, its prime purpose being to vary continually its coefficients K_1—K_8 in dependence upon the linear PCM input signal S_n applied to its input. The adaptive estimate of $K_m(n+1)$ for each stage m, where $1 \leqq m \leqq 8$, is derived generally in accordance with equation 1, (see Figure 6) where:

K_m is the reflection coefficient for stage *m* of the lattice

n is the instant sample or time slot

w is a weighting factor which weights the residual energy so that more recent values have greater significance

$f_m(n)$ is the "forward residual" at stage *m*

$g_m(n)$ is the "backward residual" at stage *m*

b is a constant (for example unity), which determines the mix between forward and backward residuals.

Preferably *w* is the impulse response of a recursive filter of finite order. Then the numerator and denominator of equation 1 can be determined in accordance with equations 2 and 3, respectively (see Figure 6). In those equations $\beta=0.988$, for example.

Referring again to Figure 3, the coefficient encoder 48 encodes the set of coefficient values (actually the difference between each set and the previously-transmitted set) every 4 milliseconds to provide the coefficient signal K_i which is applied to both the multiplexer 18 and the excitation signal generating means 16.

In the excitation signal generating means 16, the coefficient signal K_i is decoded by a decoder 52 which extracts the original set of reflection coefficients K_1—K_8. Conversion means 54 then transforms the reflection coefficients K_1—K_8 into the corresponding set of predictor coefficients a_1—a_8, using the recursive formulae specified as equations 4 and 5 in Figure 6.

The set of predictor coefficients a_1—a_8, shown as signal a_i in Figure 3, are applied to impulse response computation means 56, and "desired" signal computation means 58, respectively. The "desired" signal computation means 58 comprises an inverse filter 60, which is in all-zero filter, and a *modified* synthesis filter 62, which is an all-pole filter. The modification of the synthesis filter 62, respresented by the symbol γ, results in additional perceptual weighting being applied to the linear PCM signal S_n as it is encoded. The value of γ is between 0 and 1, preferably 0.75. More specifically, the predictor coefficients a_i of the modified synthesis filter 62 are obtained from the coefficients a_i by the relationship $a_i=\gamma^i a_i$; $1\leqq i\leqq 8$, i.e. as disclosed in the aforementioned disclosure by Messrs. Atal and Remde, but for a weighting filter rather than a synthesis filter.

The predictor coefficients of both filters 60 and 62, are updated every 4 milliseconds in response to the predictor coefficient signal a_i. The input to the inverse filter 60 is derived from the block of thirty-two 16-bit words stored in the storage means 12 at the instant the coefficient signals K_i, and hence a_i, are derived or sub-sampled.

The output of the inverse filter 60 is a linear prediction residual signal r_n, derived in accordance with equation 6 of Figure 6. The signal r_n, still comprising thirty-two 16-bit words, is applied by way of an adder 61 to the modified synthesis filter 62, which provides the "desired" signal d_n, in accordance with equation 7 of Figure 6.

The residual prediction signal r_n is also applied to RMS-computing means 63, which computes its root-mean-square value G which is then differentially encoded into a 3-bit word by RMS encoding means 65, the output of which is applied to multiplexer 18 and RMS-decoding means 67.

The adder 61 subtracts from the residual signal r_n the output of an encoder excitation pulse generator 69 which is connected to the adder 61 by a switch 71. The output of the pulse generator 69 is a set of excitation pulses P corresponding to those which will be generated in the decoder for that block of the input signal. The excitation pulse generator 69 computes the pulses using the location component *m* and amplitude component A_m derived from the excitation signal A, as will be described later. Closure of switch 71 is controlled by means of a counter 73 which counts the number of excitation pulses and closes the switch 71 after the last pulse in each block, reopening it before the first pulse of the next block.

The output of adder 61 is used by the modified synthesis filter 62 to compute the desired signal d_n twice. The first time uses the data from the inverse filter 60 directly and is the basis for the set of excitation pulses P used for the second computation. The value for d_n calculated the second time around, however, is discarded, the object of the second computation being merely to refresh the memory of the recursive modified synthesis filter.

The impulse response computation means 56 computes, for each 4 millisecond block, the impulse response h_n of the modified synthesis filter 62. The impulse response h_n comprises a corresponding block of thirty-two 16-bit words.

The impulse response h_n, and output d_n of the modified synthesis filter 62, are both applied to pulse-computation means, which computes the amplitude and location of each of the set of excitation pulses corresponding to that block or 4 millisecond period.

The impulse response h_n is applied to a cross-correlator 80, together with the "desired" signal d_n from the modified synthesis filter 62. The cross-correlator 80 correlates the two signals h_n and d_n in thirty-two steps over the entire 4 milliseconds block and produces a cross-correlation signal α_m, which is also a block of thirty-two 16-bit words. The cross-correlation signal α_m is applied by way of switch 82 to squaring means 84, which generates $(\alpha_m)^2$. The switch 82 is controlled by counter 73 so as to select the output of cross-correlator 82 for application to the squaring means 84 for computation of only the first excitation pulse. Thereafter switch 82 selects the output of an adder 86. One input of adder 86 is connected to the common or output of switch 82, by way of a buffer/delay 88, its other input being connected to the output of a multiplier 90.

The impulse response h_n is also applied to covariance computation means 92 which derives the covariance matrix (32×32 elements, each 16 bits), generally according to equation 8 (see Figure 6) and specifically, and preferably, equation 9 (see Figure 6).

One output of covariance means 92 comprises the diagonal terms $\emptyset(m,m)$ and is applied to one input of a divider 94, the output of squaring means 84 being applied to the second input of the divider 94. The output of the divider 94, representing the term $(\alpha_m)^2/\emptyset(m,m)$, is applied to pulse-maximum-locating means 98 which compares the values of all thirty-two samples thereof and selects the maximum. The temporal location of that maximum is m_1 the location of the first excitation pulse. This location signal is encoded by encoder 100 and supplied to the multiplexer 18.

The output of divider 94 is also supplied to amplitude-computing means 102, together with the location signal m, from the locating means 98, and the output of switch 82 which, for calculation of the first excitation pulse is α_m and for subsequent pulses of the set, α'_m.

The output of amplitude-computation means 102, derived by dividing $(\alpha_m^2/\emptyset(m,m)$ by α_m, is the amplitude component A_m (see equation 10 of Figure 6), which is applied to one input of a divider 104. The output of decoder 67, i.e. the gain factor G, is applied to the other input of the divider 104. The output of divider 104 is encoded by means of amplitude encoder 108 and applied to the multiplexer 18. The output of the encoder 108 is also decoded by decoder 110 and multiplied, by means of multiplier 112, by the output of the RMS decoder 67, i.e. the gain factor G, and applied to one input of multiplier 90. The other input of the multiplier 90 is connected to the output of row-selection means 114. As previously mentioned, the output of the multiplier 90 is applied to the negative input of adder 86.

Row-selection means 114 has one input derived from the covariance computation means 92, to receive the entire matrix $(O_{i,j})$, and a second input connected to the max-location means 98 to receive the location of the maximum value i.e. pulse position m. The row-selection means 114 determines to which row of the matrix the maximum pulse corresponds and supplies that row to the multiplier 90.

For derivation of the first excitation pulse, switch 82 selects the output α_m of cross-correlator 80 for application to squaring means 84. The squared cross-correlation signal α_m divided by the diagonal $\emptyset_{(m,m)}$ of the covariance matrix for the block. The maximum value of $\alpha_m^2/\emptyset_{(m,m)}$ is selected by location means 98 as the position m, of the first excitation pulse for that block. Amplitude computing means 102 derives the corresponding amplitude A_{m1}, for that first pulse according to equation 10 (Figure 6). Signal A_{m1} constitutes one element of the excitation signal component A_m, which comprises eight 4-bit words, one for each pulse. Before being encoded by amplitude encoding means 108, the amplitude A_{m1} is divided, by the divider 104, by the r.m.s./gain factor. This division process normalizes the amplitudes for different blocks of the signal.

The counter 73, which counts the excitation pulses located by location means 98, is arranged so that it resets the switch 82 to disconnect the cross-correlator 80 once the parameters, i.e. amplitude and location, of the first excitation pulse have been generated. Thus, the switch 82 applies to the squaring means 84, and amplitude-computing means 102, the output of adder 86, which is the different α'_m between the previous cross-correlation signal, stored in buffer/delay 88, and the output of multiplier 90.

The output of multiplier 90 is the product of the maximum amplitude vector and the row of the covariance matrix which contains the maximum value. The parameters of the second excitation pulse are computed in the same way as the first, using α'_m instead of α_m. The cycle repeates for the third and each subsequent pulse until a total set of eight have been generated, whereupon counter 73 restores switch 82 to its position, selecting the cross-correlator output, and momentarily closes switch 71 to refresh the memory of synthesis filter 62.

A second embodiment of the invention is illustrated in Figure 5, wherein parts corresponding to the first embodiment have the same reference numerals. In the encoder of Figure 5 the computation of the covariance, in the general sense, is simplified by using an auto-correlator 120 in place of covariance computation means 92 and row-selector 114. The squarer 84 and divider 94 and also omitted. Other modifications are that the output of switch 82 is taken to location means 122 as well as amplitude computing means 102. The locating means 122 differs from locating means 98 in Figure 3 because, although it computes the same output, it uses a different input, namely α_m direct rather than the signal $\alpha_m^2/\emptyset_{(m,m)}$. In the embodiment of Figure 5, the auto-correlator 120 supplies the first auto-correlation value, directly to the amplitude-computing means 132, as indicated by the line 124. Amplitude-computing means 132 differs from amplitude-computing means 102 of Figure 3 because, although it supplies the same output, it uses different input signals.

The impulse response signal h_n is applied to the auto-correlator 120, which auto-correlates it in thirty-two steps over the entire 4 millisecond period to provide an auto-correlation signal $R_h(i)$ which then comprises a block of thirty-two 16-bit words.

As in the first embodiment, the impulse response signal h_n, is applied to the cross-correlator 80, together with the desired signal d_n from the modified synthesis filter 62. The cross-correlator again cross-correlates the two signals h_n and d_n in thirty-two steps over the 4 millisecond period and produces a cross-correlation signal α_m, which is also a block of thirty-two 16-bit words.

To determine the location of the first excitation pulse, cross correlation signal α_m is applied by way of one pole of a switch 82 to pulse-location means 122. The pulse-location means 122 compares the absolute values of the thirty-two 16 bit words of the cross-correlation signal α_m and selects the maximum. The location of this maximum within the block is coded as a binary word m_1 which represents the location of the first excitation pulse. This binary word is applied to encoder 100, as are binary words for locations of subsequent excitation pulses. Encoder 100 encodes them to produce location component *m* of the excitation signal A and is applied to the multiplexer 18.

As before, the position of the maximum value from pulse-location means 122 is applied to amplitude-computing means 132, which receives also, direct from the autocorrelator 102 (as indicated by line 124) the value $R_h(O)$, which is the first of the thirty-two words of the auto-correlation signal R_h. The amplitude-computing means 132 derives the ratio $\alpha_{mi}/R_h(O)$ to provide a signal A_{m1} which represents the amplitude of the first excitation pulse. Signal A_{m1} is divided by gain factor G, by means of divider 104, and is then supplied to the multiplexer 18 as a 4-bit binary word by means of a pulse-amplitude encoder 118. The amplitude signal A_{m1} constitutes one element of the excitation signal component A_m, which comprises eight 4-bit words, one for each excitation pulse.

In this embodiment, the output of the autocorrelator 120, i.e. $R_h(i)$, is applied directly to multiplier 90. The output of multiplier 90 is applied, as before, to the adder 86.

The second and subsequent excitation pulses are generated in a similar manner to the first with switch 82 operated to select the output of adder 86 for application to the locating means 98, i.e. disconnecting the cross-correlation signal α_m. As before the signal α'_m is applied to the maximum locating means 122 is then the difference between α_m, the previous input to means 122, and the output of multiplier 90. The cycle is repeated, with counter 73 being incremented each time, until eight excitation pulses have been computed, whereupon switch 82 resets to apply the cross-correlation signal for the next block to the pulse-maximum locating means 122.

Claims

1. Apparatus for processing a digital signal comprising an encoder having:

input means (10) for providing a signal (S_n) in linear PCM format;

storage means (12) for storing discrete blocks of said linear PCM signal (S_n) individually and successively, each block comprising a predetermined number of samples;

coefficient generating means (14, 46, 48) for deriving from said signal (S_n) a plurality of sets of reflection coefficients, and

excitation signal generating means (15, 52, 54) for generating an excitation signal (A) corresponding to said block, said excitation signal comprising a component representing amplitude and a component representing temporal location within said block for each of a set of excitation pulses smaller in number than the number of possible PCM samples in said block, characterised in that said plurality of sets of coefficients are updated continuously on a sample by sample basis during arrival of said predetermined number of samples in that block, and in that said excitation signal generating means (16, 52, 54) generates the excitation signal (A) in response to the waveform of the signals (S_n) and the most recently derived set of said plurality of sets of coefficients which has been updated up to the last sample.

2. Apparatus as claimed in claim 1, further characterized by:

a decoder comprising:

a decoder synthesis filter (28) having adjustable predictor coefficients;

decoder input means (26) responsive to said excitation signal for providing said excitation pulses and applying them to the input of said synthesis filter (28);

coefficient means (30, 32) responsive to said coefficient signal for adjusting said filter predictor coefficients,

whereby application to said decoder synthesis filter of each said set of excitation pulses, following adjustment of its coefficients to corresponding values, produces an output signal substantially identical to the corresponding block of linear PCM signal S_n.

3. Apparatus as claimed in claim 2, wherein said encoder includes means (18) for multiplexing said excitation signal (A) and said coefficient signal, and said decoder includes means (24) for demultiplexing said excitation signal and said coefficient signal.

4. Apparatus as claimed in claims 1 or 2 wherein said coefficient generating means (14; 46, 48) is arranged to derive said coefficients on a continuous basis.

5. Apparatus as claimed in claim 4 wherein the coefficient generating means (14; 46, 48) comprises an adaptive lattice (46).

6. Apparatus as claimed in any preceding claim wherein said reflection coefficients are converted to linear prediction coefficients.

7. Apparatus as claimed in claim 6, wherein the excitation signal generating means (16) comprises:

filter means (62) for generating from the linear PCM signal a desired signal d_n;

impulse response computer means (56) for computing the impulse response h_n of the filter means (62) using the coefficients a_i appropriate to that block;

cross-correlator means (80) for computing cross-correlation a_m between the impulse response h_n and the output d_n of the filter means (62);

covariance means (92) for computing the general covariance O(i,j) of the impulse response h_n, in accordance with the general expression given in equation 8 of the attached Figure 6;

location means (98) responsive to the covariance means (92) and the cross-correlation means (80) for deriving the location of the maximum correlation, and generating therefrom an element of the said component representing temporal location, such element being the position of the first excitation pulse, and

means (84, 86, 88, 90) responsive to the covariance means, the locating means and cross-correlation means for generating the corresponding amplitude of that pulse.

8. Apparatus as claimed in claim 7, further comprising;

buffer means (88) for storing the output of the cross-correlator means, for each excitation pulse;

substraction means (86) for substracting from the buffer means output a signal representing the product of a covariance vector corresponding to the instant pulse and the corresponding amplitude parameter; and

switching means (82) operative to select the output of the cross-correlator (80) for application to said locating means (98) for computation of said first pulse and to select the output of the substraction means for application to said location means for computation of subsequent pulses in the same set.

9. Apparatus as claimed in claim 8, wherein said covariance means (92) comprises:

matrix means for computing the covariance matrix in accordance with equation 9 of Figure 6;

squaring means for deriving the square of said cross-correlation signal; and

divider means responsive to the matrix means and the squaring means for dividing the diagonal of the covariance vector by the output of the squaring means; the output of the divider means being applied to said locating means for determination of the maximum thereof.

10. Apparatus as claimed in claim 9, further comprising:

row selection means response to the matrix means and the output of the locating means for selecting as said covariance vector the row of the covariance matrix corresponding to the location represented by such locating means output, such row being applied to multiplier means for providing said product.

11. Apparatus as claimed in claim 7, wherein the filter means (62) comprises a synthesis filter having coefficients which are modified relative to those of the decoder synthesis filter.

12. Apparatus as claimed in claim 11, wherein the filter means coefficients a'_i are related to the decoder synthesis coefficients a_i by the expression $a'_i = y^i \cdot a_i$ where $1 \leqq i \leqq 8$ and $\emptyset \leqq y \leqq 1$.

13. Apparatus as claimed in claim 8, wherein said covariance means (92) comprises:

auto-correlator means for providing the auto-correlation $R_h(i)$ of the impulse response h_n, said subtraction means being arranged to subtract from the buffer output the product of the auto-correlation $r_h(i)$ and said amplitude parameter.

14. Apparatus as claimed in claim 13, wherein means for computing the amplitude is responsive to the output of the auto-correlator, at least for computing the amplitude of the first excitation pulse.

15. A method of processing a digital signal comprising encoding the digital signal by:

providing a signal S_n in linear PCM format;

storing discrete blocks of said signal S_n individually and successively, each block comprising a predetermined number of samples;

deriving from said signal S_n a plurality of sets of reflection coefficients, and

generating an excitation signal (A) corresponding to said block, said excitation signal comprising a component representing amplitude and a component representing temporal location within said block for each of a set of excitation pulses less in number than the number of possible PCM samples in said block, characterised in that said plurality of sets of coefficients are updated continuously on a sample by sample basis during arrival of said predetermined number of samples in that block, and in that the excitation signal (A) is generated in response to the waveform of the signal (S_n) and the most recently derived set of the plurality of sets of coefficients which has been updated up to the last sample.

16. A method as claimed in claim 15, further comprising the steps of:

decoding the output signal by;

deriving from said excitation signal said excitation pulses and applying them to the input of a synthesis filter having adjustable predictor coefficient signal;

adjusting said filter predictor coefficients, in response to said coefficient signal;

wherein each said set of excitation pulses is applied to said filter following corresponding adjustment of its coefficients, to produce an output signal substantially identical to the corresponding block of the linear PCM signal S_n.

17. A method as claimed in claim 16, wherein said encoding includes multiplexing said excitation signal and said coefficient signal, and said decoding including demultiplexing said excitation signal and said coefficient signal.

18. A method as claimed in claims 15 and 16 wherein said coefficients are derived on a continuous basis.

19. A method as claimed in claim 18, wherein the coefficients are derived using an adaptive lattice.

20. A method as claimed in any of claims 15 to 19 wherein said reflection coefficients are converted to linear prediction coefficients.

21. A method as claimed in claim 20 wherein the generation of said excitation signal comprises the steps of:

generating from the linear PCM signal a desired signal d_n;

computing the impulse response h_n of encoder filter means using the coefficients a_i appropriate to that block;

computing cross-correlation α_m between the impulse response h_n and the desired signal d_n;

computing the general covariance O(i,j) of the impulse response h_n, in accordance with the general expression given in equation 8 of the attached Figure 6;

in response to the covariance and the cross-correlation deriving the location of the maximum correlation, and generating therefrom an element of said component representing temporal location, such element being the position of the first excitation pulse; and

responsive to the covariance, the maximum location and cross-correlation generating the corresponding amplitude of that pulse in accordance with equation 10 of Figure 6.

22. A method as claimed in claim 21, further comprising:

storing the output of the cross-correlator means, for each excitation pulse;

subtracting from the said stored output a signal representing the product of a covariance vector corresponding to the instant pulse and the corresponding amplitude parameter; and

selecting the cross-correlation signal for computation of said first pulse and the product of the subtraction means for computation of subsequent pulses in the same set.

23. A method as claimed in claim 22, further comprising:

computing the covariance matrix in accordance with equation 9 of Figure 6;

deriving the square of said cross-correlation signal; and

in response to the covariance matrix and the squared signal dividing the diagonal of the covariance vector by the squared signal; the result of such division being used in determination of the maximum thereof.

24. A method as claimed in claim 23 further comprising:

responsive to the covariance matrix and the pulse location selecting as said covariance vector the row of the covariance matrix corresponding to such location, such row being applied to multiplier means for providing said product.

25. A method as claimed in claim 22, wherein said covariance is derived by:

providing the auto-correlation $R_h(i)$ of the impulse response h_n, said subtraction subtracting from the stored signal the product of the auto-correlation $R_h(i)$ and said amplitude parameter.

26. A method as claimed in claim 25, wherein computation of the amplitude is responsive to the auto-correlation signal, at least for computing the amplitude of the first excitation pulse.

27. A method as claimed in claim 21, wherein the encoder filter means comprises a synthesis filter having coefficients modified relative to those of the decoder synthesis filter.

28. A method as claimed in claim 27, wherein the encoder filter means has coefficients a'_i related to the coefficients a_i of the decoder synthesis filter by the expression

$$a'_i=\gamma \cdot a_i \text{ where } 1 \leqq i \leqq 8 \text{ and } \emptyset \leqq \gamma \leqq 1.$$

Patentansprüche

1. Vorrichtung zum Bearbeiten eines Digitalsignals, welche einen Kodierer enthält mit:

Eingangsmitteln (10) zur Schaffung eines Signales (S_n) in linearem PCM-Format;

Speichermitteln (12) zum einzelnen aufeinanderfolgenden Speichern diskreter Blöcke des linearen PCM-Signals (S_n), wobei jeder Block eine vorbestimmte Anzahl von Abtastungen umfaßt;

Koeffizienten-Generatormitteln (14, 46, 48) zum Ableiten einer Vielzahl von Reihen von Reflexionskoeffizienten aus dem Signal (S_n), und

Anregungssignal-Generatormitteln (15, 52, 54) zur Erzeugung eines dem Block entsprechenden Anregungssignals (A), wobei das Anregungssignal eines Komponente umfaßt, welche die Amplitude repräsentiert, und eine Komponente, welche den zeitlichen Ort für jeden aus einer Reihe von Anregungsimpulsen innerhalb des Blockes repräsentiert, die eine kleinere Zahl bilden als die Anzahl möglicher PCM-Abtastungen in dem Block,

dadurch gekennzeichnet, daß die Vielzahl von Koeffizientenreihen kontinuierlich auf Grundlage Abtastung um Abtastung aufgefrischt werden während der Ankunft der vorbestimmten Anzahl von Abtastungen in dem Block, und daß das Anregungssignal-Generatormittel (16, 52, 54) das Anregungssignal (A) in Reaktion auf die Wellenform der Signale (S_n) und die zuletzt abgeleitete Reihe aus der Vielzahl von Koeffizientenreihen erzeugt, die bis zu der letzten Abtastung aufgefrischt worden sind.

2. Vorrichtung nach Anspruch 1, weiter gekennzeichnet durch:

einen Dekoder mit:

einem Dekodier-Synthesefilter (28) mit einstellbaren Ziel-Koeffizienten;

von dem Erregungssignal abhängige Dekoder-Eingangsmitteln (26), um die Anregungsimpulse zu schaffen und sie an den Eingang des Synthesefilters (28) anzulegen;

auf das Koeffizientensignal zur Einstellung der Filter-Zielkoeffizienten reagierende Koeffizientenmittel (30, 32),

wodurch Anwendung jeder Reihe von Anregungsimpulsen auf das Dekodier-Synthesefilter nach Einstellung seiner Koeffizienten auf entsprechende Werte ein im wesentlichen mit dem entsprechenden Block des linearen PCM-Signals S_n identisches Ausgangssignal erzeugt.

3. Vorrichtung nach Anspruch 2, bei der der Kodierer Mittel (18) zum Multiplexen des Anregungssignales (A) und des Koeffizientensignales enthält, und der Dekoder Mittel (24) zum Demultiplexen des Anregungssignales und des Koeffizientensignales enthält.

4. Vorrichtung nach Anspruch 1 oder 2, bei der das Koeffizienten-Generatormittel (14; 46, 48) zum Ableiten der Koeffizienten auf einer kontinuierlichen Grundlage ausgelegt ist.

5. Vorrichtung nach Anspruch 4, bei der das Koeffizienten-Generatormittel (14; 46, 48) ein adaptives Gitter (46) umfaßt.

6. Vorrichtung nach einem der vorangehenden Ansprüche, bei der die Reflexions-Koeffizienten in lineare Ziel-Koeffizienten gewandelt werden.

7. Vorrichtung nach Anspruch 6, bei der das Anregungssignal-Generatormittel (16) umfaßt:

Filtermittel (62) zur Erzeugung eines gewünschten Signales d_n aus dem linearen PCM-Signal,

auf Impulse ansprechendes Computermittel (56) zum Errechnen der Impuls-Reaktion h_n des Filtermittels (62) unter Benutzung der für den Block zuständigen Koeffizienten a_i;

Quer-Korrelatormittel (80) zum Errechnen der Quer-korrelation a_m der Impulsreaktion h_n mit dem Ausgangssignal d_n des Filtermittels (62);

Kovarianz-Mittel (92) zum Errechnen der allgemeinen Kovarianz O(i, j) der Impulsreaktion h_n gemäß dem in Gleichung 8 der beigefügten Figur 6 gegebenen allgemeinen Ausdruck;

Lokalisierungsmittel (98) in Abhängigkeit von den Kovarianz-Mitteln (92) und den Querkorrelationsmitteln (80) zum Ableiten der Lokalisierung der maximalen Korrelation und zum Erzeugen eines Elementes der die zeitliche Lokalisierung repräsentierenden Komponente daraus, wobei dieses Element die Lage des ersten Anregungsimpulses ist, und

von den Kovarianz-Mitteln, den Lokalisierungs-Mitteln und den Querkorrelations-Mitteln abhängige Mittel (84, 86, 88, 90) zur Erzeugung der entsprechenden Amplitude jenes Impulses.

8. Vorrichtung nach Anspruch 7, die weiter enthält:

Puffermittel (88) zum Speichern des Ausgangssignales der Querkorrelatormittel für jeden Anregungsimpuls;

Subtraktionsmittel (86) zum Subtrahieren eines das Produkt eines dem augenblicklichen Impuls entsprechenden Kovarianzvektors mit dem entsprechenden Amplitudenparameter repräsentierenden Signales von dem Ausgangssignal des Puffermittels; und

Schaltmittel (82), die zum Auswählen des Ausgangssignales des Quer-Korrelators (80) zum Anlegen an das Lokalisierungsmittel (98) wirksam sind zur Errechnung des ersten Impulses und zum Auswählen des Ausgangssignales des Subtraktionsmittels zum Anlegen an das Lokalisierungsmittel zur Errechnung der folgenden Impulse der gleichen Reihe.

9. Vorrichtung nach Anspruch 8, bei der das Kovarianzmittel (92) umfaßt:

Matrix-Mittel zum Errechnen der Kovarianz-Matrix entsprechend Gleichung 9 der Figur 6;

Quadrierungsmittel zum Ableiten des Quadrates des Querkorrelationssignales; und

Dividiermittel in Abhängigkeit von den Matrix-Mitteln und dem Quadrierungsmittel zum Dividieren der Diagonale des Kovarianz-Vektors durch das Ausgangssignal des Quadrierungsmittels, wobei das Ausgangssignal des Dividiermittels an das Lokalisiermittel zur Bestimmung von dessen Maximum angelegt wird.

10. Vorrichtung nach Anspruch 9, die weiter enthält:

Zeilenwahl-Mittel in Abhängigkeit von den Matrix-Mitteln und dem Ausgangssignal der Lokalisierungsmittel zum Auswählen der Zeile der Kovarianz-Matrix, die dem durch solches Lokalisierungsmittel-Ausgangssignal dargestellten Ort entspricht, als den Kovarianzvektor, wobei diese Zeile zur Schaffung des Produktes dem Multiplikator-Mittel angelegt wird.

11. Vorrichtung nach Anspruch 7, bei dem das Filtermittel (62) ein Synthesefilter-Mittel umfaßt mit Koeffizienten, die relativ zu denen des Dekodier-Synthesefilters modifiziert sind.

12. Vorrichtung nach Anspruch 11, bei der die Filtermittel-Koeffizienten a'_i auf die Dekodier-Synthesekoeffizienten a_i durch den Ausdruck $a'_i = y^i \cdot a_i$ bezogen sind, wobei $1 \leqq i \leqq 8$ und $0 \leqq y \leqq 1$.

13. Vorrichtung nach Anspruch 8, bei der das Kovarianz-Mittel (92) umfaßt:

Auto-Korrelator-Mittel zum Schaffen der Auto-Korrelation $R_h(i)$ der Impuls-Reaktion h_n, wobei das Subtraktionsmittel ausgelegt ist, von dem Puffer-Ausgangssignal das Produkt der Autokorrelation $r_h(i)$ mit dem Amplitudenparameter zu subtrahieren.

14. Vorrichtung nach Anspruch 13, bei der das Mittel zum Errechnen der Amplitude auf das Ausgangssignal des Autokorrelators reagiert, mindestens zum Errechnen der Amplitude des ersten Anregungsimpules.

15. Verfahren zur Bearbeitung eines Digitalsignales mit Kodierung des Digitalsignales durch:

Schaffung eines Signals S_n in linearem PCM-Format;

einzelnes und aufeinanderfolgendes Speichern diskreter Blöcke des Signales S_n, wobei jeder Block eine vorbestimmte Anzahl von Abtastungen umfaßt;

Ableiten einer Vielzahl von Reihen von Reflexionskoeffizienten aus dem Signal S_n, und

Erzeugen eines Anregungssignales (A) entsprechend dem Block, wobei das Anregungssignal eine die Amplitude repräsentierende Komponente und eine den zeitlichen Ort innerhalb des Blockes repräsentierende Komponente für jeden Anregungsimpuls einer Reihe von Anregungsimpulsen umfaßt mit einer Anzahl kleiner als der Anzahl der möglichen PCM-Abtastungen in dem Block,

dadurch gekennzeichnet,

daß die Vielzahl von Reihen von Koeffizienten kontinuierlich auf Basis Abtastung um Abtastung während der Ankunft der vorbestimmten Anzahl von Abtastungen in dem Block aufgefrischt werden, und daß das Anregungssignal (A) in Abhängigkeit von den Wellenform des Signals (S_n) und der zuletzt abgeleiteten Reihe aus der Vielzahl von Reihen von Koeffizienten erzeugt wird, die bis zur letzten Abtastung aufgefrischt wurde.

16. Verfahren nach Anspruch 15, das weiter die Schritte umfaßt:

Dekodieren des Ausgangssignales durch:

Ableiten der Anregungsimpulse von dem Anregungssignal und Anlegen derselben an den Eingang eines Synthesefilters mit einstellbarem Ziel-Koeffizienten-Signal;

Einstellen der Filter-Ziel-Koeffizienten in Abhängigkeit von dem Koeffizienten-Signal;

wobei jede Reihe von Anregungsimpulsen nach entsprechender Einstellung der Koeffizienten an das Filter angelegt wird zur Erzeugung eines im wesentlichen mit dem entsprechenden Block des linearen PCM-Signals S_n identischen Ausgangssignals.

17. Verfahren nach Anspruch 16, bei dem das Kodieren das Multiplexen des Anregungssignales und des Koeffizienten-Signales einschließt und das Dekodieren das Demultiplexen des Anregungssignales und des Koeffizientensignales einschließt.

18. Verfahren nach Ansprüchen 15 und 16, bei dem die Koeffizienten auf kontinuierlicher Grundlage abgeleitet werden.

19. Verfahren nach Anspruch 18, bei dem die

Koeffizienten unter Benutzung eines adaptiven Gitters abgeleitet werden.

20. Verfahren nach einem der Ansprüche 15 bis 19, bei dem die Reflexionskoeffizienten in lineare Ziel-Koeffizienten gewandelt werden.

21. Verfahren nach Anspruch 20, bei dem die Erzeugung des Anregungssignales folgende Schritte umfaßt:

Erzeugen eines gewünschten Signals d_n aus dem linearen PCM-Signal;

Errechnen der Impulsreaktion h_n des Kodier-Filtermittels unter Benutzung der diesem Block angemessenen Koeffizienten a_i;

Errechnen der Querkorrelation α_m zwischen der Impulsreaktion h_n und dem gewünschten Signal d_n;

Errechnen der allgemeinen Kovarianz O(i, j) der Impulsreaktion h_n entsprechend dem allgemeinen Ausdruck Gleichung 8 der beigefügten Figur 6;

Ableiten des Ortes der maximalen Korrelation in Abhängigkeit von der Kovarianz und der Querkorrelation und Erzeugen eines Elementes der den zeitlichen Ort repräsentierenden Komponente daraus, wobei dieses Element die Position des ersten Anregungsimpulses ist; und

Erzeugen der entsprechenden Amplitude dieses Impulses entsprechend Gleichung 10 aus Figur 6 in Abhängigkeit von der Kovarianz, dem Maximum-Ort und der Querkorrelation.

22. Verfahren nach Anspruch 2, das weiter enthält:

Speichern des Ausgangssignales des Quer-Korrelatormittels für jeden Anregungsimpuls;

Subtrahieren eines das Produkt eines dem augenblicklichen Impuls entsprechenden Kovarianzvektors mit dem entsprechenden Amplitudenparameter darstellenden Signales von dem gespeicherten Ausgangssignal; und

Auswählen des Quer-Korrelationssignales zur Berechnung des ersten Impulses und des Produktes des Subtraktionsmittels zur Berechnung nachfolgender Impulse in der gleichen Reihe.

23. Verfahren nach Anspruch 22, das weiter enthält:

Errechnen der Kovarianz-Matrix nach Gleichung 9 der Figur 6;

Ableiten des Quadrates des Korrelationssignales; und

Dividieren der Diagonale des Kovarianzvektors durch das quadrierte Signal in Abhängigkeit von der Kovarianzmatrix und dem quadriertem Signal; wobei das Ergebnis einer solchen Division bei der Bestimmung des Maximums derselben benutzt wird.

24. Verfahren nach Anspruch 23, das weiter umfaßt:

in Abhängigkeit von der Kovarianzmatrix und dem Impulsort, Auswählen der Zeile der Kovarianzmatrix als den Kovarianzvektor, die diesem Ort entspricht, wobei eine solche Zeile an das Multiplikatormittel zur Schaffung des Produktes angelegt wird.

25. Verfahren nach Anspruch 22, bei dem die Kovarianz abgeleitet wird durch:

Schaffen der Autokorrelation $R_h(i)$ der Impulsreaktion h_n, wobei die Subtraktion das Produkt der Auto-Korrelation $R_h(i)$ mit dem Amplitudenparameter von dem gespeicherten Signal subtrahiert.

26. Verfahren nach Anspruch 25, bei dem die Berechnung der Amplitude in Abhängigkeit von dem Auto-Korrelationssignal geschieht, mindestens zur Errechnung der Amplitude des ersten Anregungsimpulses.

27. Verfahren nach Anspruch 21, bei dem das Kodierfiltermittel ein Synthesefilter mit relativ zu denen des Dekodier-Synthesefilters modifizierten Koeffizienten umfaßt.

28. Verfahren nach Anspruch 27, bei dem das Kodierfiltermittel Koeffizienten a'_i besitzt, die auf die Koeffizienten a_i des Dekodier-Synthesefilters durch den Ausdruck

$$a'_i=\gamma \cdot a_i$$

bezogen sind, wobei

$$1 \leqq i \leqq 8 \text{ und } 0 \leqq \gamma \leqq 1.$$

Revendications

1. Appareil pour traiter un signal numérique, comprenant un codeur ayant:

—un moyen d'entrée (10) pour fournir un signal (S_n) en format MIC linéaire;

—un moyen de stockage (12) pour stocker des blocs discrets du signal MIC linéaire (S_n) individuellement et successivement, chaque bloc comprenant un nombre prédéterminé d'échantillons;

—un moyen générateur de coefficient (14, 46, 48) pour obtenir à partir du signal (S_n) une multitude d'ensembles de coefficients de réflexion; et

un moyen générateur de signal d'excitation (15, 52, 54) pour produire un signal d'excitation (A) correspondant au bloc, le signal d'excitation comprenant une composante représentant l'amplitude et une composante représentant l'emplacement temporel à l'intérieur du bloc pour chaque ensemble d'impulsions d'excitation d'un nombre plus petit que le nombre d'échantillons MIC possibles dans le bloc,

caractérisé en ce que la multitude d'ensembles de coefficients est mise à jour de façon continue sur la base échantillon par échantillon pendant l'arrivée du nombre prédéterminé d'échantillons dans ce bloc, et en ce que le moyen générateur de signal d'excitation (16, 52, 54) produit le signal d'excitation (A) en réponse à la forme d'onde des signaux (S_n) et l'ensemble obtenu le plus récemment de la multitude d'ensembles de coefficients qui a été mise à jour jusqu'au dernier échantillon.

2. Appareil selon la revendication 1, caractérisé en outre par:

—un décodeur comprenant:

—un filtre (28) de synthèse de décodeur ayant des coefficients de prédiction ajustables;

—un moyen (26) d'entrée de décodeur répondant au signal d'excitation pour fournir les impulsions d'excitation est les appliquer à l'entrée du filtre de synthèse (28);

—un moyen de coefficient (30, 32) répondant au signal de coefficient pour ajuster le coefficient de prédiction du filtre;

—d'où il résulte que l'application au filtre de synthèse du décodeur de chaque ensemble d'impulsions d'excitation, à la suite du réglage de ses coefficients à des valeurs correspondantes, produit un signal de sortie sensiblement identique au bloc correspondant au signal MIC linéaire S_n.

3. Appareil selon la revendication 2, dans lequel le codeur comprend un moyen (18) pour multiplexer le signal d'excitation (A) et le signal de coefficient, et le décodeur comprend un moyen (24) pour démultiplexer le signal d'excitation et le signal de coefficient.

4. Appareil selon la revendication 1 ou la revendication 2, dans lequel le moyen générateur de coefficient (14; 46, 48) est disposé de manière à donner les coefficients sur une base continue.

5. Appareil selon la revendication 4, dans lequel le moyen générateur de coefficient (14; 46, 48) comprend un réseau adaptatif (46).

6. Appareil selon l'une quelconque des revendications précédentes, dans lequel les coefficients de réflexion sont transformés en coefficients de prédiction linéaire.

7. Appareil selon la revendication 6, dans lequel le moyen générateur de signal d'excitation (16) comprend:

—un moyen de filtre (62) pour produire à partir du signal MIC linéaire un signal désiré d_n;

—un moyen (56) de calcul de réponse impulsionnelle pour calculer la réponse impulsionnelle h_n du moyen de filtre (62) en utilisant les coefficients a_i appropriés à ce bloc;

—un moyen de corrélateur croisé (80) pour calculer une corrélation croisée a_m entre la réponse impulsionnelle h_n et la sortie d_n du moyen de filtre (62);

—un moyen de covariance (92) pour calculer la covalence générale O(i, j) de la réponse impulsionnelle h_n, en conformité avec l'expression générale donnée dans l'équation 8 de la figure 6 annexée;

—un moyen de positionnement (98) répondant au moyen de covariance (92) et au moyen de corrélation croisée (80) pour donner l'emplacement de la corrélation maximum, et produire à partir de celui-ci un élément de ladite composante représentant l'emplacement temporel, un tel élément étant la position de la première impulsion d'excitation; et

—un moyen (84, 86, 88, 90) répondant au moyen de covariance, au moyen de positionnement et au moyen de corrélation croisée pour produire l'amplitude correspondante de cette impulsion.

8. Appareil selon la revendication 7, comprenant en outre:

—un moyen de tampon (88) pour stocker la sortie du moyen de corrélation croisée, pour chaque impulsion d'excitation;

—un moyen de soustraction (86) pour soustraire de la sortie du moyen de tampon un signal représentant le produit d'un vecteur de covariance correspondant à l'impulsion instantanée et du paramètre correspondant sur l'amplitude; et

—un moyen de commutation (82) pouvant fonctionner pour sélectionner la sortie du corrélateur croisé (80) pour application au moyen de positionnement (98) afin de calculer la première impulsion et pour sélectionner la sortie du moyen de soustraction pour application au moyen de positionnement pour le calcul des impulsions ultérieures de même ensemble.

9. Appareil selon la revendication 8, dans lequel le moyen de covariance (92) comprend:

—un moyen de matrice pour calculer la matrice de covariance en conformité avec l'équation 9 et la figure 6;

—un moyen de mise au carré pour obtenir le carré du signal de corrélation croisée; et

—un moyen de diviseur répondant au moyen de matrice ou au moyen de mise au carré pour diviser la diagonale de vecteur de covariance par la sortie du moyen de mise au carré; la sortie du moyen de diviseur étant appliquée au moyen de positionnement pour déterminer son maximum.

10. Appareil selon la revendication 9, comprenant en outre:

—un moyen de sélection de rangée répondant au moyen de matrice et à la sortie du moyen de positionnement pour sélectionner comme vecteur de covariance la rangée de la matrice de covariance correspondant à l'emplacement représenté par une telle sortie du moyen de positionnement, la rangée étant appliquée à un moyen de multiplicateur pour fournir le produit.

11. Appareil selon la revendication 7, dans lequel le moyen de filtre (62) comprend un filtre de synthèse ayant des coefficients qui sont modifiés par rapport à ceux du filtre de synthèse du décodeur.

12. Appareil selon la revendication 11, dans lequel les coefficients du moyen de filtre a'_i sont liés aux coefficients de synthèse du décodeur a_i par l'expression $a'_i = y^i \cdot a_i$ où $1 \leqq i \leqq 8$ et $\emptyset \leqq y \leqq 1$.

13. Appareil selon la revendication 8, dans lequel le moyen de covariance (92) comprend:

—un moyen d'auto-corrélateur pour fournir l'auto-corrélation $R_h(i)$ de la réponse impulsionnelle h_n, le moyen de soustraction étant agencé de manière à soustraire de la sortie du tampon le produit de l'auto-corrélation $r_h(i)$ et du paramètre sur l'amplitude.

14. Appareil selon la revendication 13, dans lequel le moyen pour calculer l'amplitude répond à la sortie de l'auto-corrélateur, au moins pour calculer l'amplitude de la première impulsion d'excitation.

15. Procédé de traitement d'un signal numérique, comprenant le codage du signal numérique en procédant aux étapes suivantes:

—fournir un signal S_n sous format MIC linéaire;

—stocker des blocs discrets du signal S_n individuellement et successivement, chaque bloc comprenant un nombre prédéterminé d'échantillons;

—obtenir à partir du signal S_n une multitude d'ensembles de coefficients de réflexion; et

—générer un signal d'excitation (A) correspondant au bloc, le signal d'excitation comprenant une composante représentant l'amplitude et une composante représentant l'emplacement temporel à l'intérieur du bloc pour chacune d'un ensemble d'impulsions d'excitation d'un nombre inférieur au nombre des échantillons MIC possibles dans le bloc;

caractérisé en ce que la multitude d'ensembles de coefficients est mise à jour continuellement sur la base échantillon par échantillon pendant l'arrivée du nombre prédéterminé d'échantillons dans ce bloc, et en ce que le signal d'excitation (A) est produit en réponse à la forme d'onde du signal (S_n) et l'ensemble obtenu le plus récemment de la multitude d'ensembles de coefficients qui a été mis à jour jusqu'au dernier échantillon.

16. Procédé selon la revendication 15, comprenant en outre les étapes consistant à:

—décoder le signal de sortie en:

—obtenant à partir du signal d'excitation les impulsions d'excitation et en les appliquant à l'entrée d'un filtre de synthèse ayant un signal de coefficient de prédiction ajustable;

—adjustant les coefficients de prédiction du filtre, en réponse au signal de coefficient;

où chaque ensemble d'impulsions d'excitation est appliqué au filtre à la suite du réglage correspondant de ses coefficients, afin de produire un signal de sortie sensiblement identique au bloc correspondant du signal MIC linéaire S_n.

17. Procédé selon la revendication 16, dans lequel le codage comprend le multiplexage du signal d'excitation et du signal de coefficient, et le décodage comporte le démultiplexage du signal d'excitation et du signal de coefficient.

18. Procédé selon la revendication 15 et la revendication 16, dans lequel les coefficients sont obtenus sur une base continue.

19. Procédé selon la revendication 18, dans lequel les coefficients sont obtenus en utilisant un réseau adaptatif.

20. Procédé selon l'une quelconque des revendications 15 à 19, dans lequel les coefficients de réflexion sont transformés en coefficients de prédiction linéaire.

21. Procédé selon la revendication 20, dans lequel la génération du signal d'excitation comprend les étapes consist ant à:

—générer à partir du signal MIC linéaire un signal désiré d_n;

—calculer la réponse impulsionnelle h_n du moyen de filtre du codeur en utilisant les coefficients a_i concernant le bloc;

—calculer la corrélation croisée α_m entre la réponse impulsionnelle h_n et le signal désiré d_n;

—calculer la covariance générale O(i, j) de la réponse impulsionnelle h_n, en conformité avec l'expression générale donnée dans l'équation 8 de la figure 6 annexée;

—en réponse à la covariance et à la corrélation croisée, obtenir l'emplacement de la corrélation maximum, et générer à partir de celui-ci un élément de la composante représentant l'emplacement temporel, un tel élément étant la position de la première impulsion d'excitation; et

—en réponse à la covariance, l'emplacement maximum et la corrélation croisée, générer l'amplitude correspondante de cette impulsion en conformité avec l'équation 10 de la figure 6.

22. Procédé selon la revendication 21, comprenant en outre les étapes consistant à:

—stocker la sortie du moyen de corrélateur croisé, pour chaque impulsion d'excitation;

—soustraire de la sortie stockée un signal représentant le produit d'un vecteur de covariance correspondant à l'impulsion instantanée et du paramètre de l'amplitude correspondant, et

—sélectionner le signal de corrélation croisée pour le calcul de la première impulsion et du produit du moyen de soustraction pour calculer les impulsions ultérieures du même ensemble.

23. Procédé selon la revendication 22, comprenant en outre les étapes consistant à:

—calculer la matrice de covariance en conformité avec l'équation 9 de la figure 6;

—obtenir le carré du signal de corrélation croisée; et

—en réponse à la matrice de covariance et au signal mis au carré, diviser la diagonale du vecteur de covariance par le signal mis au carré, le résultat de cette division étant utilisé dans la détermination de son maximum.

24. Procédé selon la revendication 23, comprenant en outre l'étape consistant à:

—en réponse à la matrice de covariance et à l'emplacement de l'impulsion sélectionner comme vecteur de covariante la rangée de la matrice de covariance correspondant à un tel emplacement, cette rangée étant appliquée au moyen de multiplicateur pour fournir le produit.

25. Procédé selon la revendication 22, dans lequel la covariance est obtenue en:

—fournissant l'auto-corrélation $R_h(i)$ de la réponse impulsionnelle h_n, la soustraction procédant à la soustraction du signal stocké du produit de l'autor-corrélation $R_h(i)$ et du paramètre d'amplitude.

26. Procédé selon la revendication 25, dans lequel le calcul de l'amplitude répond au signal d'autocorrélation, au moins pour calculer l'amplitude de la première impulsion d'excitation.

27. Procédé selon la revendication 21, dans lequel le moyen de filtre du codeur comprend un filtre de synthèse ayant des coefficients modifiés par rapport à ceux du filtre de synthèse du décodeur.

28. Procédé selon la revendication 27, dans lequel le moyen de filtre du codeur a des coefficients a'_i liés aux coefficients a_i du filtre de synthèse du décodeur par l'expression

$$a'_i=\gamma \cdot a_i \text{ où } 1\leqq i\leqq 8 \text{ et } \emptyset\leqq\gamma\leqq 1.$$

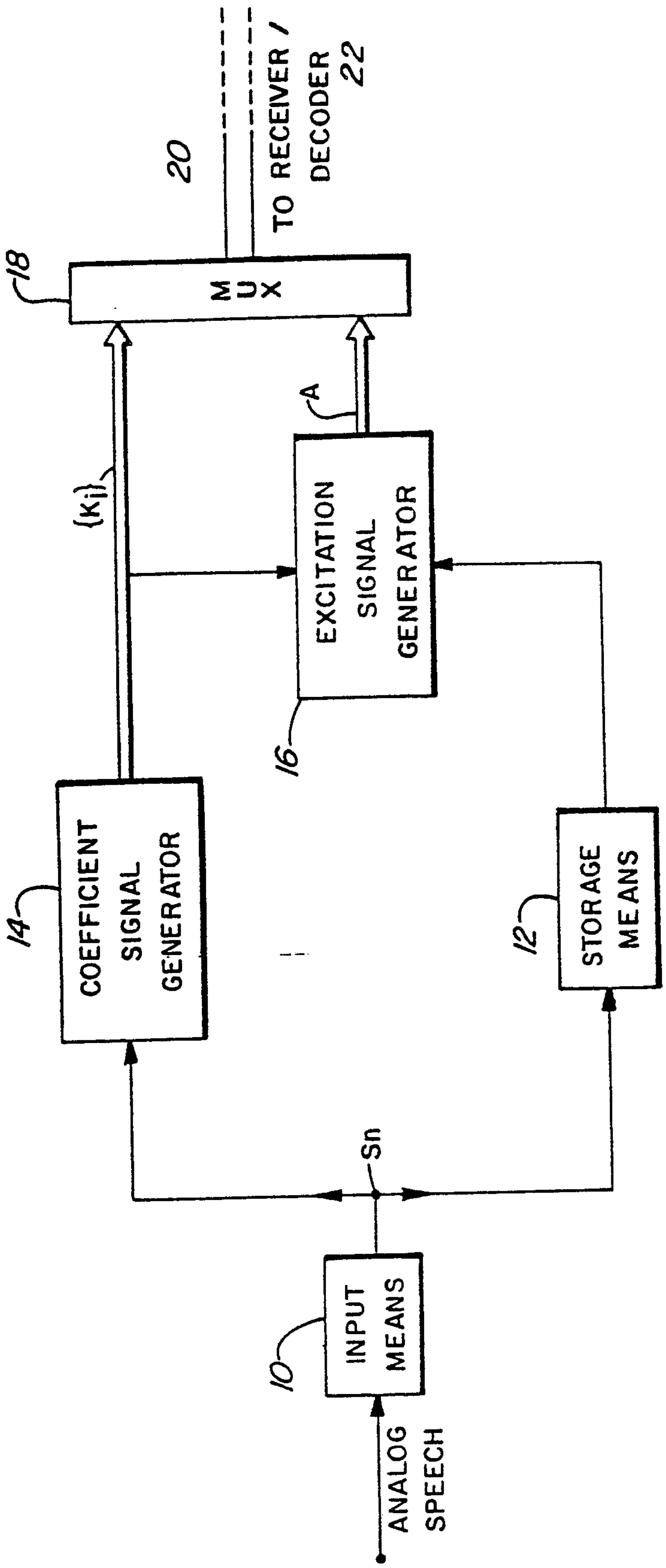
ANALOG SPEECH
10
INPUT MEANS
Sn
14
COEFFICIENT SIGNAL GENERATOR
12
STORAGE MEANS
{Ki}
16
EXCITATION SIGNAL GENERATOR
A
18
MUX
20
TO RECEIVER / DECODER
22

FIG. 1

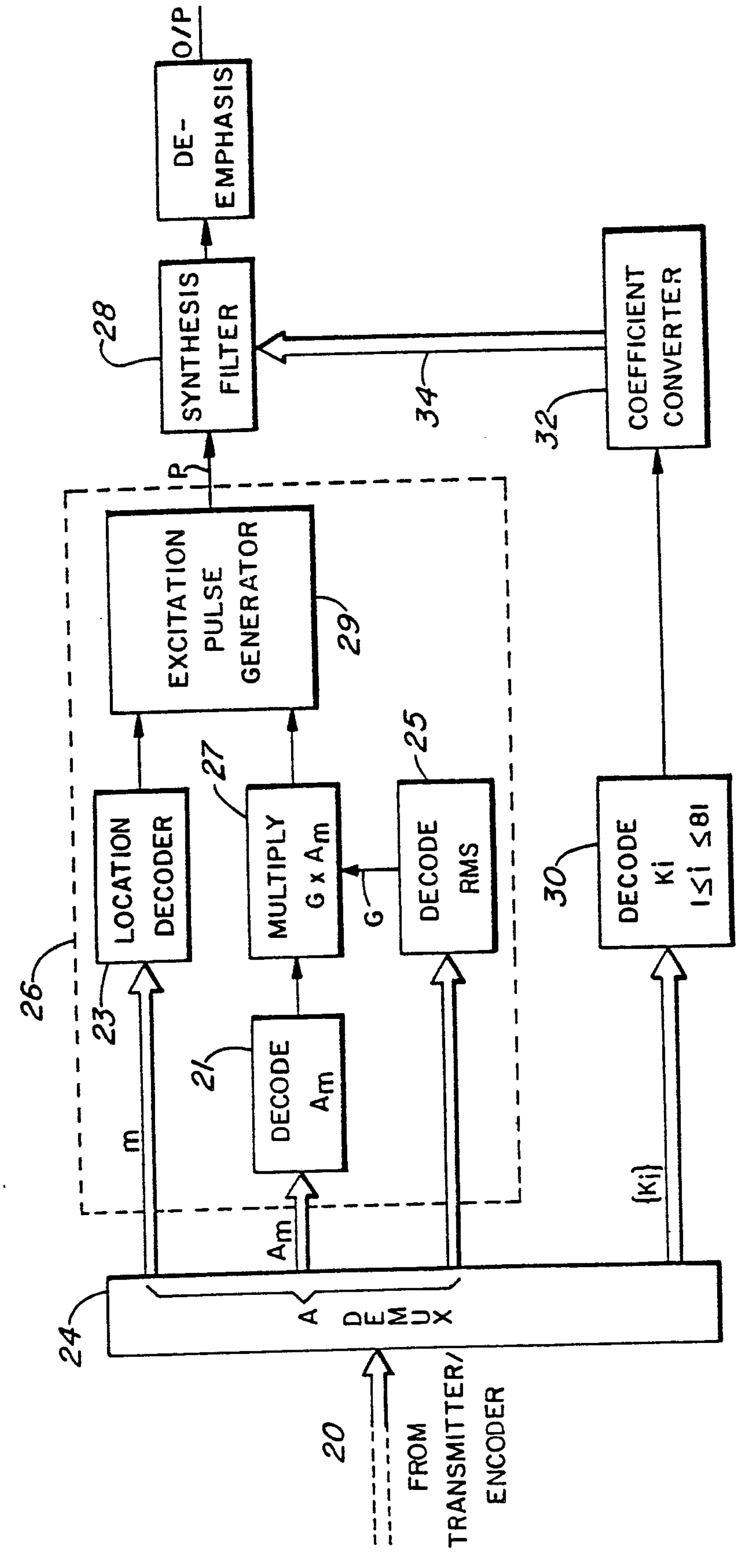
O/P
DE-EMPHASIS
SYNTHESIS FILTER
28
COEFFICIENT CONVERTER
32
34
P
EXCITATION PULSE GENERATOR
29
LOCATION DECODER
23
26
MULTIPLY G x Am
27
DECODE RMS
25
G
DECODE Am
21
m
Am
DECODE Ki 1≤i≤8
30
{Ki}
DEMUX
24
20
FROM TRANSMITTER/ ENCODER

FIG. 2

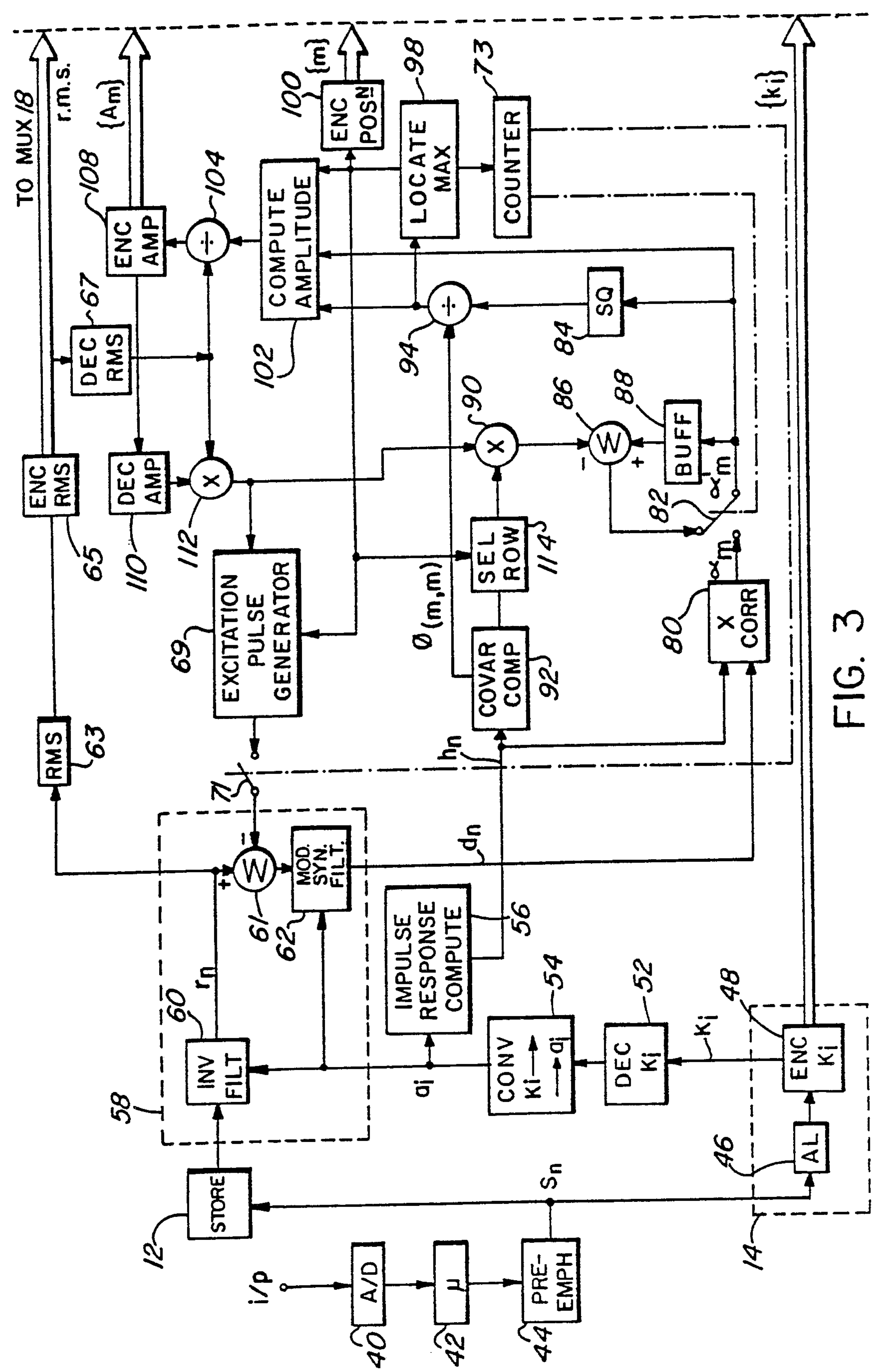
TO MUX 18
r.m.s.
{Am}
{m}
{ki}
ENC AMP
108
104
100
ENC POSN
98
LOCATE MAX
73
COUNTER
COMPUTE AMPLITUDE
102
DEC RMS
67
SQ
84
94
ENC RMS
65
DEC AMP
110
112
90
86
88
BUFF
82
αm
EXCITATION PULSE GENERATOR
69
SEL ROW
114
Ø(m,m)
COVAR COMP
92
X CORR
80
RMS
63
71
hn
dn
MOD. SYN. FILT.
61
62
IMPULSE RESPONSE COMPUTE
56
INV FILT
60
rn
58
CONV Ki→ai
54
DEC Ki
52
Ki
ENC Ki
48
AL
46
ai
STORE
12
Sn
14
i/p
A/D
40
μ
42
PRE-EMPH
44

FIG. 3

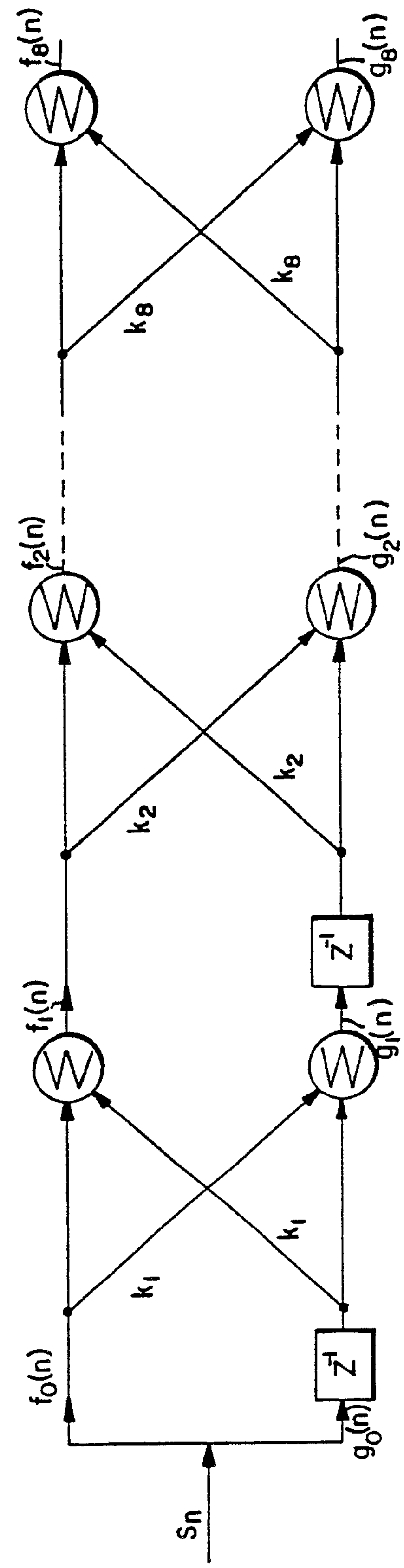

FIG. 4

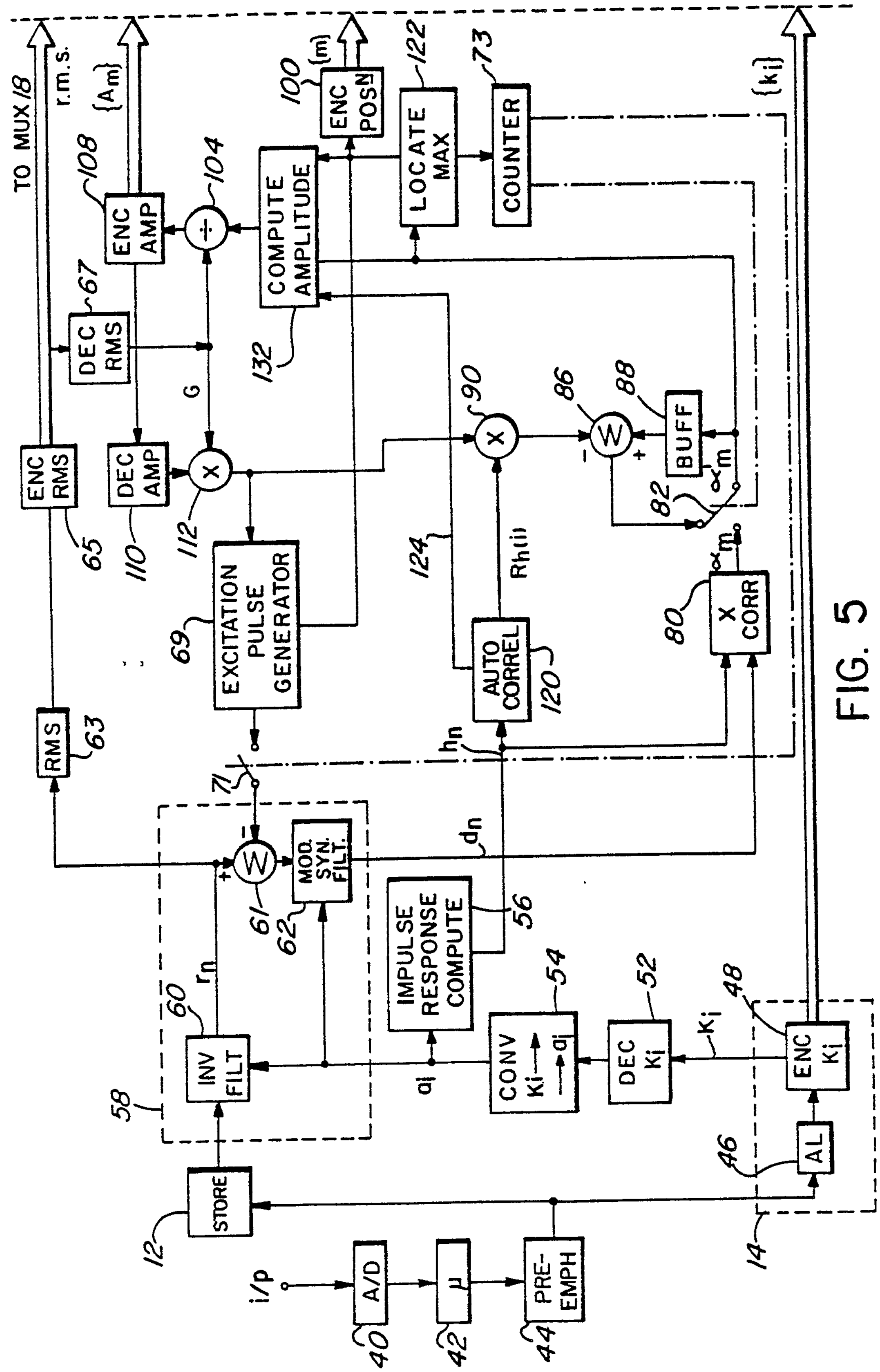
TO MUX 18
r.m.s.
{Am}
{m}
{ki}
ENC AMP
ENC POSN
LOCATE MAX
COUNTER
COMPUTE AMPLITUDE
DEC RMS
ENC RMS
DEC AMP
G
BUFF
EXCITATION PULSE GENERATOR
AUTO CORREL
X CORR
Rh(i)
hn
dn
rn
ai
Ki
RMS
MOD. SYN. FILT.
IMPULSE RESPONSE COMPUTE
INV FILT
CONV Ki → ai
DEC Ki
ENC Ki
AL
STORE
A/D
μ
PRE-EMPH
i/p
αm

FIG. 5

$$K_m(n+1) = -\frac{\Sigma_{k=-}^{n} \quad W(n-k)\, f_{m-1}(k) g_{m-1}(k-1)}{\Sigma_{k=-}^{n} \quad W(n-k)\left[b f_{m-1}^{2}(k)+(1-b) g_{m-1}^{2}(k-1)\right.}$$

$$= -\frac{C_m(n)}{D_m(n)} \text{ ---------- } (1)$$

$$C_m(n) = \beta . C_m(n-1) - f_{m-1}(n g_{m-1}(n-1) \text{ ---------- } (2)$$

$$D_m(n) = \beta D_m(n-1) - \left[b f_{m-1}^{2}(n) - (1-b) g_{m-1}^{2}(n-1)\right. \text{ ---------- } (3)$$

$$a_m^{(m)} = K_m \text{ ---------- } (4)$$

$$a_j^{(m)} = a_j^{(m-1)} - K_m a_{m-j}^{(m-1)}, \quad 1-j--1 \text{ ---------- } (5)$$

$$r_n = s_n - \sum_{i=1}^{8} a_i s_{n-i} \text{ ---------- } (6)$$

$$d_n = r_n - \sum_{i=1}^{8} a_i' d_{n-i} \text{ ---------- } (7)$$

$$\phi(i,j) = \sum_{n} h_{n-i} h_{n-j} \text{ ---------- } (8)$$

$$\phi(i-1,j-1) = \phi(i,j) - h_{N-i} h_{N-j} \text{ ---------- } (9)$$

$$\hat{A}_m = \frac{\alpha_m}{\phi(m,m)} \text{ ---------- } (10)$$

FIG. 6