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Description

This invention relates to a method of synthesizing a given signal, typical of that received from sources in a real ocean environment, the method being of the kind comprising generating a noise signal, and so
 5 filtering the noise signal in accordance with linear prediction coefficients as to replicate the given signal, and to corresponding apparatus.

Acoustic trainers are required which produce signatures characteristic of signals received from sources in a real ocean environment. Traditionally, the broadband and harmonic spectral contents of targets and the broadband content of background noise have been emphasized for replication, as described in Chapter 7
 10 and 10 of "Principles of Underwater Sound for Engineers", 2nd Edition, 1975, published by MacGraw Hill, New York. Recently, active echoes and reverberation have been added to the trainer repertoire, as described in a report entitled "Multi Application Sonar Trainer" by Raytheon Company, 1st October 1982. An additional component of the acoustic environment which is required for purposeful training is the set of transient signatures. These include occasional and also continuous biologic emissions, hatch openings and
 15 closings, ice fractures in the arctic environment, undersea seismic disturbances, and the noise of submerged wrecks moving with currents - just to name a few. The synthesis of these transients has typically resided in an instructor controlled analog tape recorder. The disadvantage of this approach is the large number of tapes required and/or the problem of and time required for locating a particular sound amongst a number of sounds on a long tape. In addition, the control of the tape recorder and its connection are
 20 cumbersome.

In a paper entitled "The Application of Linear Prediction Techniques to Background Noise Generation for Sonar Trainers" published for the International Conference on Acoustics, Speech and Signal Processing, 1983, Boston, Massachusetts, by IEEE, Charles E. Schmid describes the generation of broad band background noise by passing white noise through an all-pole filter to create the desired spectrum.
 25 Background noise changes very slowly and in this known technique a pseudo-random noise generator supplied white noise to a linear prediction coefficient filter feeding a single stage deemphasis filter coupled through a digital to analog convertor to a low-pass filter which produced the analog output. The linear prediction coefficient filter used twelve reflection coefficients representing the best fit to the desired spectrum according to least square error criteria. The article refers to an experiment conducted to test the
 30 audibility of transitions between noise condition, such as shifting sea states or ship's speeds, and concludes that in general, immediate spectral level changes greater than 1dB were perceptible and should be changed gradually in time to maintain realism.

FR-A-2 510 288 also describes a linear prediction coefficient system of synthesizing submarine sonar sounds, in particular, the sounds of propellers, diesel engines, and other such sources from which the type
 35 of ship or engine can be recognised. FR-A-2 510 288 is particularly concerned with the transients that arise in the synthesized output when one set of linear prediction coefficients is replaced by another set of linear prediction coefficients, and discloses as a solution to this problem the use of an interposed set or sets of linear prediction coefficients derived from the autocorrelation function or functions corresponding to one or more transfer functions calculated from linear interpolation between the squares of the respective initial and
 40 final transfer functions corresponding to the two sets of linear prediction coefficients between which the change is to take place. Thereby a smooth transition in the simulated sound is obtained.

In contrast to the prior art, it is an object of the present invention to provide a method and apparatus for synthesizing a transient signal.

According to one aspect of the present invention there is provided a method of the kind defined
 45 hereinbefore at the beginning, characterised in that the noise signal is a pseudo-random noise signal which changes at a sample rate, the linear prediction coefficients are derived from the covariance matrix of a respective block of consecutive signal samples of the given signal and are changed at a block rate, the given signal being divided into a set of contiguous frames and each block of the samples being obtained from a respective frame, the gain of the filtering is derived from the covariance matrix coefficients and the
 50 linear prediction coefficients and is changed in synchronism with the coefficients, and the given signal is a transient signal.

According to another aspect of the invention there is provided apparatus for synthesizing a given signal, comprising means for generating a noise signal, and a filter for so filtering the noise signal in accordance with linear prediction coefficients as to replicate the given signal, characterised in that the noise signal
 55 generating means comprises a pseudo-random noise generator the output of which changes at a sample rate, the filter is a recursive filter including multiplying means for multiplying the noise signal with the said linear predictive coefficients (b_i) and a gain coefficient (A), and means for changing the gain and linear prediction coefficients (A , b_i) in synchronism at a block rate, the linear prediction coefficients (b_i) being

derived from the covariance matrix of a respective block of consecutive signal samples of the given signal, the gain coefficient (A) being derived from the covariance matrix coefficients (a_i) and the linear prediction coefficients (b_i), and the given signal being a transient signal.

A preferred embodiment of this invention provides a computer controlled synthesis system providing transient audio signals. The system is not cumbersome and is easy to use in the selection of different stored transient sounds. The digital synthesizer has denser packaging (smaller volume) for storing a large repertoire of audio sound signals, and is more reliable than the instructor controlled analog tape recorder. The method of synthesis utilizes linear prediction coding techniques to derive time-varying-filter coefficients. These coefficients are stored in digital form and are used to program a recursive filter which is driven by white noise. The resulting signatures are then an inherent part of the trainer and are generated under complete computer control. The system approximates the desired transient signature by the storage of sets of coefficients of a recursive digital filter, which coefficients are updated periodically thereby resulting in an output from the recursive filter which is a close approximation of the actual transient signature. It is assumed that an autoregressive model will provide an adequate description of the desired transient signature. The signature which is desired to be synthesized is most easily obtained from a recording of the signature which is later to be synthesized. Because the spectral content of the transient signature is time varying, the auto-regressive model is nonstationary and must be updated periodically. Therefore, the transient signature is synthesized by considering the signal to be comprised of a serial sequence of blocks of the signal. Each block of the signal has its amplitude sampled to provide 1024 samples of digital data. The autocorrelation function of the 1024 amplitude samples provides the 12 most significant autocorrelation values and a gain value which is obtained through the normalization of the autocorrelation values. A matrix equation is obtained relating the autocorrelation values of the actual signal to the unknown coefficients of a recursive filter. The system of equations of the matrix is solved for each block of data and the filter coefficients are stored. Through the synthesizer signature the coefficients are periodically updated in real-time by a control processor. The coefficients are recovered from memory in real-time and provided to the recursive filter circuitry whose output is provided to a digital to analog converter to produce audible sound replicating the original transient signature. There is thus provided the synthesis of audio signals from stored digital data and more particularly a synthesizer in which the stored data are sets of the coefficients of a recursive filter and where each set is applied to the filter for a fixed time period, the periods totaling the duration of the synthesized audio signal.

Brief Description of the Drawings

The aforementioned aspects and other features, objects and advantages of the method and apparatus of this invention will be apparent from the following detailed description taken in conjunction with the accompanying drawings wherein:

FIG. 1 shows the waveform of the original transient audio signal which is reproduced by the synthesizer of this invention;

FIG. 2 shows a flow diagram of a time-varying recursive filter;

FIG. 3 is an analog representation of an embodiment of the synthesizer of this invention; and

FIG. 4 is a digital implementation of a preferred embodiment of the synthesizer of this invention.

Description of the Preferred Embodiment

The transient analog signal 10 of figure 1 which is to be simulated by the apparatus of this invention is operated upon by first partitioning the analog signal into a sequence of frames 11. The time duration of each frame is determined by examining the power spectral characteristics of the signal over a multiple of frame durations and then choosing the maximum duration over which those spectral characteristics are essentially constant. The greater the frequency extent of the power distribution, the shorter is the time period for that frame. The signal within each frame is then periodically sampled at a rate T_c exceeding the Nyquist rate and stored in digital form. The set of samples 12 which is stored for each signal frame is a block of digital data. One block of data results for each signal frame. The number of sample points per block is determined by the frame duration and by the highest frequency contained in the data signal waveform which is to be synthesized. At least two and preferably four data points are obtained within a signal frame for each cycle of the highest frequency component within that frame. The sample data points contained within each block are autocorrelated and a selected number of the autocorrelation coefficients are

determined. The number of autocorrelations values which are used is equal to the least number required to reduce the autocorrelation coefficient recursive prediction residual to an acceptable fraction of the zero lag autocorrelation value. An acceptable fraction is commonly 0.01. The autocorrelation values are normalized by a factor R which provides unity value of the autocorrelation coefficient at zero displacement. For the waveforms and block times utilized in embodiments of this invention, it is found that twelve autocorrelation coefficients are adequate to produce synthesized audio signals which are indistinguishable from the original signal from which the autocorrelation functions were obtained. The autocorrelation function coefficients have been designated by the letters $a_0, a_1, \dots, a_m$ with the subscript indicating the relative lag displacement in the autocorrelated data block. The unity value coefficient a_0 is the normalized autocorrelation coefficient at zero relative displacement.

The covariance matrix is next employed to determine the values for the multiplication factors "b" applied to the output of each of the delay units of the recursive filter as shown in FIG. 2. The covariance matrix is given below where "a" with subscripts are the normalized autocorrelation coefficient values. "b" with subscripts are the linear prediction coefficients obtained from the covariance matrix and are the multiplying factors which are applied to the multipliers of the recursive filter in addition to the gain factor A.

$$\begin{bmatrix} a_0 & a_1 & \cdot & \cdot & \cdot & a_{m-1} \\ a_1 & a_0 & \cdot & \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\ \cdot & \cdot & \cdot & \cdot & \cdot & \cdot \\ a_{m-1} & \cdot & \cdot & \cdot & \cdot & a_0 \end{bmatrix} \cdot \begin{bmatrix} b_1 \\ b_2 \\ \cdot \\ \cdot \\ \cdot \\ b_m \end{bmatrix} = \begin{bmatrix} a_1 \\ a_2 \\ \cdot \\ \cdot \\ \cdot \\ a_m \end{bmatrix} ; a_0 = 1$$

$m = \text{number of coefficient}$

$$A^2 = R(1 + \sum_{k=1}^m a_k \cdot b_k)$$

The transient signatures are generated by feeding "white" (uncorrelated) noise samples $S(n)$ through a time-varying recursive digital filter. The transfer function of this filter, $G(Z)$, is given as

$$G(Z) = -A/(1 - b_1 z^{-1} - b_2 z^{-2} \dots - b_m z^{-m})$$

where A is the filter gain, and the b's are the multiplier coefficients. A flow diagram of the recursive filter 20 is shown in FIG. 2. The random noise signal $S(n)$ produces an amplitude modulated signal which changes its amplitude from one level to another at the same rate as that at which the original analog signal was sampled. The random noise signal is multiplied in multiplier 22 by the factor A, where A is determined from the normalization of the autocorrelation function as explained earlier and is constant during each block. The output of multiplier 22 is applied to an adder 23 which provides the output $y(n)$ ($n = \text{total number of samples of the signal being simulated}$) of the recursive filter 20 and also provides $y(n)$ to the first delay stage 24' having a delay which is equal to the intersample interval. The output $y(n-1)$ of the delay unit 24' is transferred to a second delay unit 24'' and also is provided to a multiplier 25' which multiplies the output $y(n-1)$ by the coefficient b_1 obtained from the covariance matrix. The output of multiplier 25' is provided as an input to the adder 23. The process of delaying the earlier sampled values $y(n-2), \dots, y(n-m)$ continues in the remaining delay units 24 whose outputs are respectively multiplied by the coefficients $b_2, b_3, b_4 \dots b_m$ (constant during each block) in multipliers 25 whose outputs are in turn applied as inputs to adder 23. The transfer function of the recursive filter 20 of FIG. 2 is that given by the transfer function of the preceding equation.

Referring now to FIG. 3, there is shown an analog representation of a circuit for the implementation of the synthesizer of this invention. The analog synthesizer 50 comprises a pseudo-random noise generator 51 which produces an analog output signal having a value between zero and one which changes with every clock pulse input, the clock pulses having a period T_c which is the same period as that at which the original signal 10 was sampled. The clock pulses are provided by clock pulse generator 52 which also provides the clock pulses to the counter 53 having a modulo F where F is the number of samples of the analog signal in

one block time. The counter 53 provides block pulses whose period T_f is equal to the period of the clock pulses T_c multiplied by the number of samples F , $T_f = T_c \cdot F$. At the end of each block a pulse having period T_f is applied to the memory 54 to produce a new set of analog numbers $A_1, -b_1, \dots, -b_m$. The memory 54 is represented as a multi-pole switch having $n + 1$ poles with the switch arms 55 moving by one switch position in response to each energization of the switch coil 56 by the pulse T_f . Each set of coefficients appears at a selected position of switch arms 25. As shown in FIG. 2, the initial position of the switch arm provides the set of coefficients $A, -b_1, \dots, -b_m$. The second switch arm position which would exist as a result of one pulse T_f would provide a different set of coefficients for the second block time; namely, $A', -b'_1, \dots, b'_m$. The last set of coefficients corresponding to the last block of the sampled input signal is provided by the memory 54 as $A^k, b_1^k, \dots, -b_m^k$, where k is the number of blocks.

The output of the pseudo-random noise generator 51 is provided to a multiplier 57 whose other input during a block time is the amplitude coefficient A . The output of multiplier 57 is provided at one input of the summing circuit 58. The output of the summing circuit 58 is provided as the input to a delay unit 59_1 whose output is provided to delay unit 59_2 and to multiplier 57_1 . The other input to multiplier 57_1 is the coefficient $-b_1$ provided by the memory 54 during the first time block. The output of multiplier 57_1 is provided at another input to the summing circuit 58. The time delay provided by delay 59_1 is equal to the interpulse period T_c of the clock pulses provided by generator 52. The circuit 50 has a cascade of delay elements $59_2, \dots, 59_m$ connected serially to the delay 59_1 . The output of the summer circuit is the desired simulated signal which corresponds to the original signal which is being simulated. This simulated signal is designated as $y(n)$. The output of each delay unit $59_1, 59_2, \dots, 59_m$ is correspondingly $y(n-1), y(n-2), \dots, y(n-m)$. The circuit 50 will, therefore, provide an output $y(n)$ in accordance with the equation presented earlier which sounds like the original audio signal which was sampled to provide coefficients b in the manner described earlier and stored in memory 54.

Referring now to the block diagram of FIG. 4 showing an embodiment of the invention, the coefficients which have been computed in the manner detailed in the preceding paragraphs are stored in sequential addresses of a RAM or ROM coefficient memory 31. In the example of the embodiment of this invention, it will be assumed that seven coefficients b_1 through b_7 of FIG. 2 together with the gain factor A are adequate for the synthesis and are stored in the first eight addresses 0, ..., 7 of memory 31. Addresses 8, ..., 15 will contain the coefficients $A', -b'_1, \dots, -b'_7$. Successive groups of eight addresses of RAM 31 have successive groups of coefficients $A, -b_1$ through $-b_7$, (one group for each block of signal data) with the total number of groups of coefficients equaling the number of blocks of the original audio signal which is to be simulated by the synthesizer 30 of FIG. 4. The address generator 32 comprises a counter 33 of modulo m ($m = 8$ for the group of eight addresses), a counter 34 of modulo L ($L =$ number of sample points per block), a block counter 341 responsive to the L th count of counter 34 and a buffer register 35. Counter 33 has clock input pulses provided by clock pulse generator 61 having a period T_c/m equal to the sampling period of the original audio signal divided by the number of coefficients " m " in a group ($m = 8$ in this example). Counter 34 has input clock pulses having a period T_c obtained from the modulo eight output line 44 of counter 33. Counter 34 is of modulo L , where L is the number of samples per block of input signal. The output pulse of counter 34 at the count of L increments by one the block counter 341. The output count of counter 341 is provided to the more significant bits (MSB) of buffer register 35 which provide the block address to the memory 31. The output count on line 33' of counter 33 is provided as the least significant bits (LSB) of register 35. Therefore, the output address of address generator 32 on line 321 will initially produce (through adder 65 and register 66) the sequential addresses 0, ..., 7 to the memory 31 repetitively for the number of samples L in the block, followed by the addresses 8, ..., 15, repeated L times, etc. Therefore, the memory 31 output will be a group of sequential coefficients $A, -b_1, \dots, -b_m$ at a period $T_c/8$ (for addresses 1 through 8) repeated L times because of the modulo L of counter 34. Block counter 341 which is incremented by one changes the MSB of register 35 so that the addresses 8, ..., 15 of memory 31 provide the next group of coefficients $A', -b'_1, \dots, -b'_m$ repeated L times also. This process of providing successive groups of coefficients to synthesize blocks of a signal continues until the memory 31 addresses contain no coefficients.

The pseudo-random noise generator 36 produces a 16-bit word for every 16-bit coefficient provided by memory 31. The word produced by noise generator 36 is stored in a 16-bit register 37. The memory 31 also produces the coefficients as 16-bit digital words and stores the words in register 38. Registers 37 and 38 provide digital inputs to multiplier 39 which provides a 32-bit output word to adder 40. Adder 40 provides an input to accumulator register 41 whose output is provided as a second input to adder 40 and whose output is provided also as an input to switch 42. Switch 42 is open except when closed in response to a pulse on line 44 provided by the modulo m output of counter 33 to the random access memory 45. The counter 33 of modulo 8 provides clock pulses T_c on line 44 as an input to counter 47 which increments a

write address to memory 45 at a time such that the switch 42 provides the output $y(n)$ as an input to memory 45. Switch 46 is also responsive to clock pulses at the period T_c provided by counter 33 on line 44. Closing of switch 46 by a pulse on line 44 allows the 16-bit number from random number generator 36 to be provided to the register 37 at that time as stated earlier. Since pulses in lines 44 only occur during the eighth count of counter 33, during the remaining seven other outputs of counter 33, switch 46 has an input 461 connected to the output of memory 45. The write address provided by counter 47 at output 48 is provided as one input to subtract circuit 49. The other input to subtractor 49 is the output count from counter 33 on line 33'. The read address is provided at the time that the switch 46 input line 461 is providing a signal corresponding to that address from memory 45 to the register 37.

In operation, the circuit of FIG. 4 provides a newly calculated value of $y(n)$ at intervals corresponding to the original sampling period T_c . Initially, the RAM 31 generates an amplitude coefficient A which is stored in register 38 and multiplied in multiplier 39 by the output $x(n)$ of the random noise generator 38 provided to register 37 through closed switch 46. The product $A \cdot x(n)$ is stored in accumulation register 41. Switch 42 is open and no output appears to be read into memory 45 through its input register 60. At the next occurrence of clock pulse T_c , the switch 46 connects register 37 with the output of memory 45. Subtractor circuit 49 has an input address 48 and an input 33' which causes the next read address presented to memory 45 to be the address next preceding that at which the output $y(n)$ has been written in by write address counter 47. This address will cause the value $y(n - 1)$ to be read out to the register 37. At the same time, the address generator 32 is indexed to the second address of memory 31 and the value $-b_1$ will be read out and provided to register 38. The resulting product provided by multiplier 39, $-b_1 y(n - 1)$, is added in adder 40 to the previously stored value $Ax(n)$ in register 41 and the sum $(Ax(n) - b_1 y(n - 1))$ is then stored in accumulation register 41. The next timing pulse T_c/m causes the read address provided to memory 45 to be decremented by one and provide the output $y(n - 2)$ to the register 37 through switch 46. At the same time, the address provided to memory 31 is incremented by one to provide the coefficient $-b_2$ to register 38. The contents of the registers 37, 38 are multiplied in multiplier 39 to provide $-b_2 y(n - 2)$ which is added in adder 40 to the existing contents $(Ax(n) - b_1 y(n - 1))$ of accumulation register 41 and the result $(Ax(n) - b_1 y(n - 1) - b_2 y(n - 2))$ is then stored in register 41. This process continues until the last coefficient b_7 at the eighth address of memory 31 is provided to register 38 and the contents of memory 45 at the address containing $y(n - 7)$ are multiplied, added and accumulated in register 41 which is then cleared and read out through switch 42 by a pulse on line 44 to input register 60. Register 60 then contains the output $y(n + 1)$ (which becomes the new value of $y(n)$) which is written into the next sequential address of memory 45 inasmuch as the write address counter 47 responsive to a pulse at the rate $1/T_c$ on line 44 from counter 33 has caused the write address on line 48 to be incremented by one. When the new value of $y(n)$ appears at the output of register 41, switch 42 is caused to close by a pulse on line 44 to thereby provide a new $y(n)$ output and to provide this new value as the input to the memory 45 at the incremented address. The output $y(n)$ of switch 42 is in digital form and is converted to an analog signal $Y(n)$ in digital-to-analog converter 62. Signal $Y(n)$ is smoothed in filter 63 to remove the sampling frequency components, centered at frequencies $1/T_c$ and multiples thereof, and to thereby provide the synthesized analog signal $y(t)$ that is desired corresponding to the blocks of coefficients selected by the initial address provided by start address register 64 which is added in adder 65 to the output of buffer register 35 and stored in register 66 before being provided to coefficient memory 31. The computer 67 is programmed to provide one or a series of start addresses at predetermined time intervals to register 64 in response to a START command to thereby produce one or a series of timed, different synthesized audio output signals $y(t)$, each corresponding to a different start address.

Having described a preferred embodiment of the invention, it will be apparent to one of skill in the art that other embodiments incorporating its concept may be used. It is believed, therefore, that this invention should not be restricted to the disclosed embodiment but rather should be limited only by the scope of the appended claims.

Claims

1. A method of synthesizing a given signal typical of that received from sources in a real, ocean environment, comprising generating a noise signal (36), and so filtering (39, 40, 41) the noise signal in accordance with linear prediction coefficients (b_i) as to replicate the given signal (10), characterised in that the noise signal is a pseudo-random noise signal (36) which changes at a sample rate (44), the linear prediction coefficients (b_i) are derived from the covariance matrix of a respective block of

consecutive signal samples (12) of the given signal (10) and are changed at a block rate (33'), the given signal (10) being divided into a set of contiguous equal duration frames (11) and each block of the samples (12) being obtained from a respective frame (11), the gain (A) of the filtering is derived from the covariance matrix coefficients (a_i) and the linear prediction coefficients (b_i) and is changed in synchronism with the coefficients; and the given signal (10) thereby synthesized is a transient signal.

2. Apparatus for synthesizing a given signal typical of that received from sources in a real ocean environment, comprising means (36) for generating a noise signal, and a filter (39, 40, 41) to so filter the noise signal in accordance with linear prediction coefficients (b_i) as to replicate the given signal (10), characterised in that the noise signal generating means comprises a pseudo-random noise generator (36) the output of which changes at a sample rate, the filter is a recursive filter including multiplying means (39) for multiplying the noise signal with the said linear prediction coefficients (b_i) and a gain coefficient (A), and means (61, 32, 65, 66, 31) for changing the gain and linear prediction coefficients (A, b_i) in synchronism at a block rate (33'), the linear prediction coefficients (b_i) being derived from the covariance matrix of a respective block (11) of consecutive signal samples (12) of the given signal (10), the gain coefficient (A) being derived from the covariance matrix coefficients (a_i) and the linear prediction coefficients (b_i), and the given signal (10) thereby synthesized being a transient signal.
3. Apparatus according to claim 2, characterised in that means (61, 33) are provided for producing clock pulses at the sample rate and are supplied to the pseudo-random noise generator (36), means (31) are provided for store sets of the said coefficients (b_i), and the recursive filter includes a plurality of delay means (45) having an output (46) coupled to the multiplying means (39), and an adder (40) to which the multiplying means (39) is connected, the said storing means (31) being adapted to supply each of said coefficients (b_i) in each respective set to the multiplying means (39), and means (67, 64, 65, 66) for changing each set of coefficients (b_i) in response to a predetermined number of clock pulses.
4. Apparatus according to claim 3, characterised in that the clock pulse means (61, 33) provide a fixed period between pulses which corresponds to the sampling period of the said transient signal.
5. Apparatus according to claim 2, characterised in that the pseudo-random noise generator (36) is responsive to a clock pulse generator (61, 33) and produces a random amplitude signal at each clock pulse; a first memory (31) contains a plurality of sets of a normalization factor and the linear prediction coefficients (b_i) in a like plurality of sets of addresses of said memory (31), each set of a normalization factor and coefficients corresponding to a corresponding block of samples of the given signal; means (67, 64, 65, 66, 38) are provided for repetitively providing sequentially each factor and coefficient of successive sets; a second memory (45) is provided for storing successive values $y(n)$ of the output of the filter (39, 40, 41) at successive addresses; the multiplying means (39) is adapted to multiply in sequence said random amplitude signal with said factor, and to multiply the first one of the sequences of coefficients (b_i) with the values $y(n)$ in reverse order of succession in which $y(n)$ values are stored in said second memory (45) to provide a succession of products; and means (40) are provided for adding said products of each sequence of coefficients to provide a value $y(n)$ for each sequence, said means (67, 64, 65, 66, 38) repetitively providing said sets of coefficients being coupled to the adding means (40) whereby a corresponding number of values $y(n)$ result, and said means (67, 64, 65, 66, 38) repetitively providing said sets of coefficients (b_i) providing the same set of coefficients (b_i) for the same number of times as said given signal (10) is sampled in each frame (11) whereafter the next successive set of coefficients (b_i) is repetitively provided.
6. Apparatus according to claim 2, characterised in that the pseudo-random noise generator (36) is responsive to a clock pulse generator (61, 33) to produce a random amplitude signal at each clock pulse; and the filter includes means (67, 64, 65, 66, 31, 38) repetitively providing a plurality of sets of said coefficients (A) for a fixed number of times corresponding to the number of samples in each frame (11); means (39) multiplying each coefficient of said sets of coefficients with one of said random input signals and successive earlier values $y(n)$ of the output of the filter to provide corresponding products; and means (40) for adding said corresponding products to provide a value of $y(n)$ for each set of coefficients, said means (67, 64, 65, 66, 31, 38) for repetitively providing said sets of coefficients operating in such a manner that a corresponding number of successive values $y(n)$ of the filter output results.

Revendications

1. Procédé pour Synthétiser un signal donné qui est typique de celui reçu de sources se trouvant dans un environnement océanique réel, comprenant la génération d'un signal de bruit (36) et le filtrage (39,40,41) du signal de bruit en fonction de coefficients de prédiction linéaire (b_i) de manière à obtenir une réplique du signal donné (10), caractérisé en ce que la signal de bruit est un signal de bruit pseudo-aléatoire (36) qui varie à une cadence d'échantillonnage (44) en ce que les coefficients de prédiction linéaire (b_i) sont dérivés de la matrice de covariance d'un bloc respectif d'échantillons de signal consécutifs (12) du signal donné (10) et Sont modifiés à une cadence de bloc (33'), le signal donné (10) étant divisé en un ensemble de trames (11) contigües de durée égale et chaque bloc d'échantillons (12) étant obtenu à partir d'une trame (11), an ce que le gain (A) du filtrage est dérivé des coefficients (a_i) de la matrice de covariance et des coefficients de prédiction linéaire (b_i) et est modifié en synchronisme avec ces coefficients; et en ce que le signal donné (10) ainsi synthétisé est un signal transitoire.
2. Appareil pour synthétiser un signal donné typique de celui reçu à partir de sources se trouvant dans un environnement océanique réel, comprenant un organe (36) destiné à générer un signal de bruit et un filtra (39,40,41) pour filtrer le signal de bruit en fonction de coefficients de prédiction linéaire (b_i) de manière à obtenir une réplique du signal donné (10), caractérisé en ce que l'organe destiné à générer le signal de bruit comprend un générateur (36) de bruit pseudo-aléatoire dont la sortie varie à une cadence d'échantillonnage, en ce que le filtre est un filtre récursif comprenant un organe (39) multiplicateur destiné à multiplier le signal de bruit par lesdits coefficients de prédiction linéaire (b_i) et un coefficient de gain (A) et un organe (61,32,65,66,31) destiné à modifier le gain et les coefficients de prédiction" linéaire (A, b_i) en synchronisme avec une cadence de bloc (33'), les coefficients de prédiction linéaire (b_i) étant dérivés de la matrice de covariance d'un bloc respectif (11) d'échantillons (12) de signal consécutifs du signal donné (10), le coefficient de gain (A) étant dérivé des coefficients (a_i) de la matrice de covariance et des coefficients de prédiction linéaire (b_i) et le signal donné (10) ainsi synthétisé étant un signal transitoire.
3. Appareil selon la revendication 2, caractérisé en ce qu'un organe (61,33) destiné à produire des impulsions d'horloge à la cadence d'échantillonnage est prévu, les impulsions étant fournies au générateur de bruit pseudo-aléatoire (36), en ce qu'un organe (31) est prévu pour mémoriser des ensembles desdits coefficients (b_i) et en ce que le filtre récursif comprend une pluralité d'éléments à retard (45) comportant une sortie (46) reliée à l'organe multiplicateur (39) et un additionneur (40) auquel est relié l'organe multiplicateur (39), ledit organe de mémorisation (31) étant adapté pour fournir chacun desdits coefficients (b_i) se trouvant dans chaque ensemble respectif à l'organe multiplicateur (39) et un organe (67,64,65,66) destiné à faire varier chaque ensemble de coefficients (b_i) en réponse à un nombre prédéterminé d'impulsions d'horloge.
4. Appareil selon la revendication 3, caractérisé en ce que l'organe d'impulsions d'horloge (61,33) fournit une période fixe entre les impulsions qui correspond à la période d'échantillonnage dudit signal transitoire.
5. Appareil selon la revendication 2, caractérisé en ce que le générateur (36) de bruit pseudoaléatoire est sensible à un générateur d'impulsions d'horloge (61,33) et produit un signal d'amplitude aléatoire à chaque impulsion d'horloge ; en ce qu'une première mémoire (31) contient une pluralité d'ensembles d'un facteur de normalisation et des coefficients de prédiction linéaire (b_i) dans une pluralité semblable d'ensembles d'adresses de ladite mémoire (31), chaque ensemble d'un facteur de normalisation et de coefficients correspondant à un bloc correspondant d'échantillons du signal donné ; en ce qu'un organe (67,64,65,66,38) est prévu pour fournir de manière répétitive en séquence chaque facteur et coefficient d'ensembles successifs; en ce qu'une seconde mémoire (45) est prévue pour mémoriser des valeurs successives $y(n)$ de la sortie du filtre (39,40,41) à des adresses successives ; en ce que l'organe multiplicateur (39) est adapté pour multiplier en séquence ledit signal d'amplitude aléatoire par ledit facteur et pour multiplier la première des séquences de coefficients (b_i) par les valeurs $y(n)$ en ordre inverse de la succession dans laquelle les valeurs $y(n)$ sont mémorisées dans ladite seconde mémoire (45) pour fournir une succession de produits ; et en ce qu'un organe (40) est prévu pour additionner lesdits produits de chaque séquence de coefficients pour fournir une valeur $y(n)$ pour chaque séquence, ledit organe (67,64,65,66,38) fournissant de manière répétitive lesdits ensembles de coeffi-

cients étant relié à l'organe additionneur (40), dont il résulte un nombre correspondant de valeurs $y(n)$, et ledit organe (67,64,65,66,38) fournissant de manière répétitive lesdits ensembles de coefficients (b_i) fournissant le même ensemble de coefficients (b_i) pour le même nombre de fois que ledit signal (10) est échantillonné dans chaque trame (11), l'ensemble successif suivant de coefficients (b_i) étant ensuite
 5 fourni de manière répétitive.

6. Appareil selon la revendication 2, caractérisé en ce que le générateur (36) de bruit pseudoaléatoire est sensible à un générateur d'impulsions d'horloge (61,33) pour fournir un signal d'amplitude aléatoire à chaque impulsion d'horloge ; et en ce que le filtre comprend un organe (67,64,65,66,31,38) fournissant
 10 de manière répétitive une pluralité d'ensembles desdits coefficients (A, b_i) pour un nombre fixé de fois correspondant au nombre d'échantillons dans chaque trame (11), un organe (39) multipliant chaque coefficient desdits ensembles de coefficients par l'un desdits signaux d'amplitude aléatoires et des valeurs successives précédentes $y(n)$ de la sortie du filtre pour fournir des produits correspondants, et
 15 un organe (40) pour additionner lesdits produits correspondants pour fournir une valeur de $y(n)$ pour chaque ensemble de coefficients, ledit organe (67,64,65,66,31,38) destiné à fournir de manière répétitive lesdits ensembles de coefficients fonctionnant de manière telle qu'un nombre correspondant de valeurs successives $y(n)$ de la sortie du filtre en résulte.

20 Ansprüche

1. Verfahren zum Synthesieren eines gegebenen Signales, wie es charakteristischerweise von Quellen im Meer empfangen wird, bei welchem ein Rauschsignal (36) erzeugt und das Rauschsignal entsprechend linearen Vorhersage-Koeffizienten (b_i) so gefiltert (39, 40, 41) wird, daß das gegebene Signal (10) nachgebildet wird, dadurch gekennzeichnet, daß das Rauschsignal ein Pseudo-Zufalls-Rauschsignal (36) ist, welches sich mit einer bestimmten Tastungsgeschwindigkeit (44) ändert, daß die linearen Vorhersage-Koeffizienten (b_i) von der Kovarianz-Matrix eines jeweiligen Blocks von aufeinanderfolgenden Signaltastungen (12) des gegebenen Signals (10) abgeleitet werden und mit einer Blockgeschwindigkeit (33') geändert werden, wobei das gegebene Signal (10) in einen Satz von aneinandergrenzenden Feldern gleicher Dauer (11) aufgeteilt wird und jeder Block von Tastungen (12) von einem jeweiligen Feld (11) gewonnen wird, daß der Verstärkungsgewinn (A) bei der Filterung von den Kovarianz-Matrix-Koeffizienten (a_i) und den linearen Vorhersage-Koeffizienten (b_i) abgeleitet wird und sich im Synchronismus mit den Koeffizienten ändert und daß das hierdurch synthetisierte gegebene Signal (10) ein transientes Signal ist.
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2. Einrichtung zum Synthesieren eines gegebenen Signales, wie es charakteristischerweise von Quellen im Meer empfangen wird, mit Mitteln (36) zur Erzeugung eines Rauschsignales und einem Filter (39, 40, 41) zur derartigen Filterung des Rauschsignales entsprechend linearen Vorhersage-Koeffizienten (b_i), daß das gegebene Signal (10) nachgebildet wird, dadurch gekennzeichnet, daß die Mittel zur Erzeugung des Rauschsignales einen Pseudo-Zufalls-Rauschgenerator (36) enthalten, dessen Ausgang sich mit einer bestimmten Tastungsgeschwindigkeit ändert, daß das Filter ein Rekursivfilter ist, welches Multipliziereinrichtungen (39) zum Multiplizieren des Rauschsignales mit den genannten linearen Vorhersage-Koeffizienten (b_i) und einem Verstärkungsgewinn-Koeffizienten (A) und Mittel (61, 32, 65, 66, 31) zur Änderung des Verstärkungsgewinns und der linearen Vorhersage-Koeffizienten (A, b_i) im Synchronismus mit einer Blockgeschwindigkeit (33') enthält, wobei die linearen Vorhersage-Koeffizienten (b_i) von der Kovarianz-Matrix eines jeweiligen Blockes (11) von aufeinanderfolgenden Signaltastungen (12) des gegebenen Signales (10) abgeleitet werden, der Verstärkungsgewinkoeffizient (A) von den Kovarianz-Matrix-Koeffizienten (a_i) und den linearen Vorhersage-Koeffizienten (b_i) abgeleitet wird und das gegebene Signal (10), welches hierdurch synthetisiert wird, ein transientes Signal ist.
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3. Einrichtung nach Anspruch 2, dadurch gekennzeichnet, daß Mittel (61, 33) zur Erzeugung von Taktimpulsen mit der Tastungsgeschwindigkeit vorgesehen sind, welche zu dem Pseudo-Zufalls-Rauschgenerator (36) geführt werden, daß weitere Mittel (31) zur Speicherung von Sätzen der genannten Koeffizienten (b_i) vorgesehen sind und daß das Rekursivfilter eine Mehrzahl von Verzögerungsmitteln (45) enthält, die mit einem Ausgang (46) an die Multipliziereinrichtungen (39) angeschlossen sind und weiter einen Addierer (40) enthält, mit welchem die Multipliziereinrichtungen (39) verbunden sind, wobei die genannten Speichermittel (31) so ausgebildet sind, daß sie jeden der genannten Koeffizienten (b_i) in jedem Satz an die Multipliziereinrichtungen (39) liefern und daß
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schließlich Mittel (67, 64, 65, 66) zur Änderung jedes Satzes von Koeffizienten (b_i) in Abhängigkeit von einer vorbestimmten Zahl von Taktimpulsen vorgesehen sind.

4. Einrichtung nach Anspruch 3, dadurch gekennzeichnet, daß die Mittel (61, 33) zur Erzeugung der Taktimpulse eine feste Zeitdauer zwischen den Impulsen vorgeben, welche der Tastungsperiode des transienten Signals entspricht.
5. Einrichtung nach Anspruch 2, dadurch gekennzeichnet, daß der Pseudo-Zufalls-Rauschgenerator (36) auf einen Taktimpulsgenerator (61, 33) anspricht und bei jedem Taktimpuls ein Signal zufälliger Amplitude erzeugt, daß ein erster Speicher (31) eine Anzahl von Sätzen eines Normalisierungsfaktors und der linearen Vorhersage-Koeffizienten (b_i) an einer entsprechenden Anzahl von Sätzen von Adressen des genannten Speichers (31) enthält, wobei jeder Satz eines Normalisierungsfaktors und von Koeffizienten einem entsprechenden Block von Tastungen des gegebenen Signales entspricht, daß weiter Mittel (67, 64, 65, 66, 38) vorgesehen sind, um sich wiederholend der Reihe nach jeweils den Faktor und die Koeffizienten aufeinanderfolgender Sätze zu erzeugen, daß ein zweiter Speicher (45) vorgesehen ist, um aufeinanderfolgende Werte $y(n)$ des Ausgangs des Filters (39, 40, 41) an aufeinanderfolgenden Adressen zu speichern, daß die Multipliziereinrichtungen (39) so ausgebildet sind, daß sie der Reihe nach das genannte Signal zufälliger Amplitude mit dem genannten Faktor und außerdem die jeweils erste der Folgen von Koeffizienten (b_i) mit den Werten $y(n)$ in umgekehrter Reihenfolge multiplizieren, wobei die Werte $y(n)$ in dem genannten zweiten Speicher (45) gespeichert werden, so daß man eine Folge von Produkten erhält, und daß schließlich Mittel (40) vorgesehen sind, um die genannten Produkte jeder Folge von Koeffizienten zu addieren und für jede Folge einen Wert $y(n)$ zu bilden, wobei die Mittel (67, 64, 65, 66, 38) zur sich wiederholenden Bildung der genannten Gruppen von Koeffizienten mit dem Mittel (40) zum Addieren verbunden sind, so daß eine entsprechende Anzahl von Werten $y(n)$ erhalten wird, und wobei die Mittel (67, 64, 65, 66, 38) zur sich wiederholenden Bildung der genannten Sätze von Koeffizienten (b_i) denselben Satz von Koeffizienten (b_i) für dieselbe Zahl von Malen liefern, die das gegebene Signal (10) in jedem Feld (11) getastet wird, wonach der nächstfolgende Satz von Koeffizienten (b_i) sich wiederholend gebildet wird.
6. Einrichtung nach Anspruch 2, dadurch gekennzeichnete daß der Pseudo-Zufalls-Rauschgenerator (36) auf einen Taktimpulsgenerator (61, 33) anspricht, um bei jedem Taktimpuls ein Signal mit zufälliger Amplitude zu erzeugen und daß das Filter Mittel (67, 64, 65, 66, 31, 38) zur wiederholten Bildung einer Mehrzahl von Sätzen der genannten Koeffizienten (A, b_i) für eine feste Anzahl von Malen entsprechend der Anzahl von Tastungen in jedem Feld (11), weiter Mittel (39) zum Multiplizieren jedes Koeffizienten der genannten Sätze von Koeffizienten mit einem der genannten Signale zufälliger Amplitude und aufeinanderfolgenden früheren Werten $y(n)$ des Ausgangs des Filters zur Bildung entsprechender Produkte, und schließlich Mittel (40) enthält, um die entsprechenden Produkte zu addieren und einen Wert von $y(n)$ für jeden Satz von Koeffizienten zu bilden, wobei die Mittel (67, 64, 65, 66, 31, 38) zur wiederholten Bildung der genannten Sätze von Koeffizienten in der Weise arbeiten, daß eine entsprechende Anzahl von aufeinanderfolgenden Werten $y(n)$ von Filterausgängen erhalten wird.

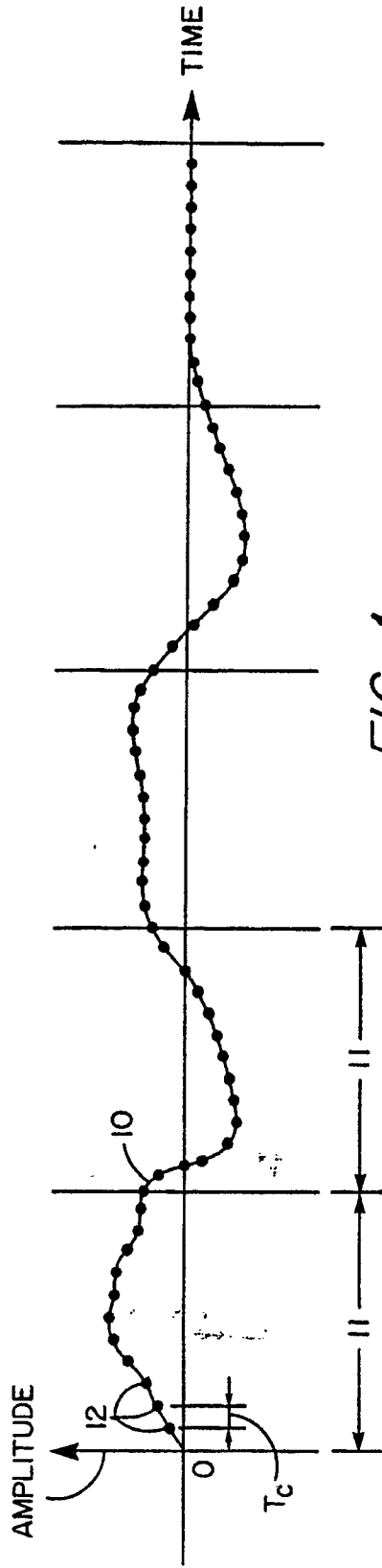


FIG. 1

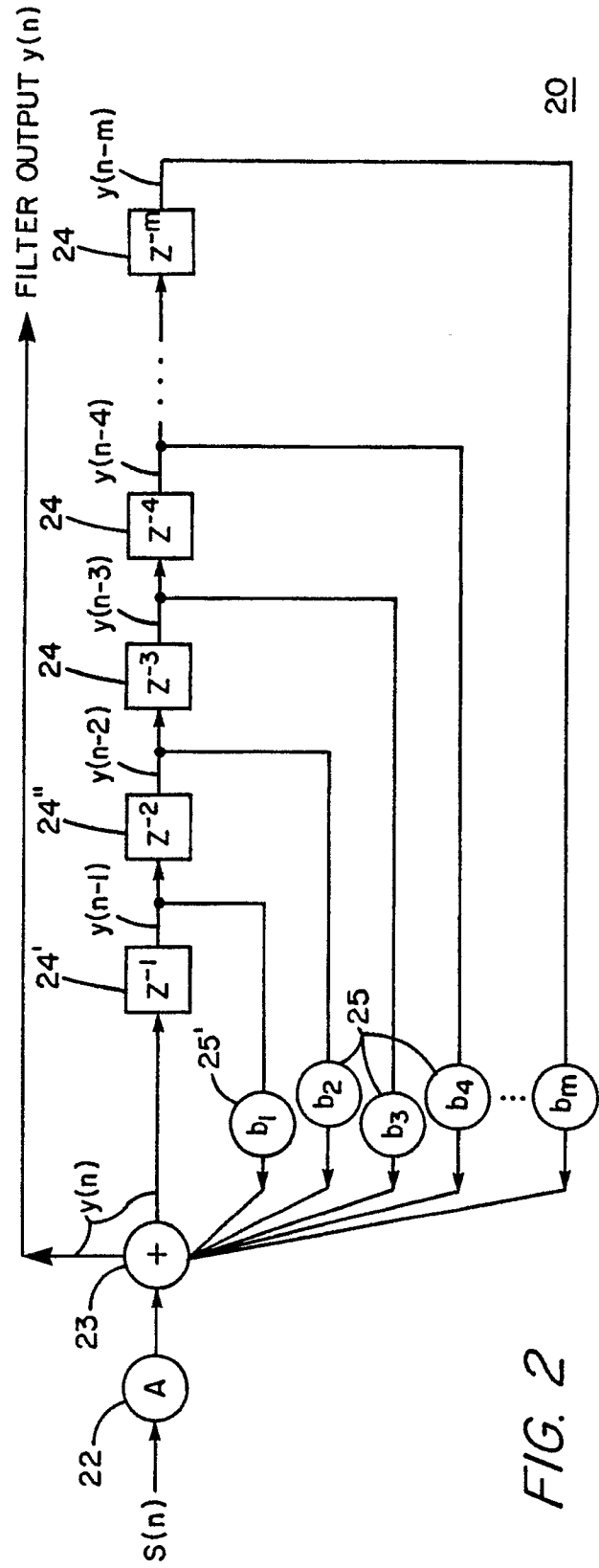


FIG. 2

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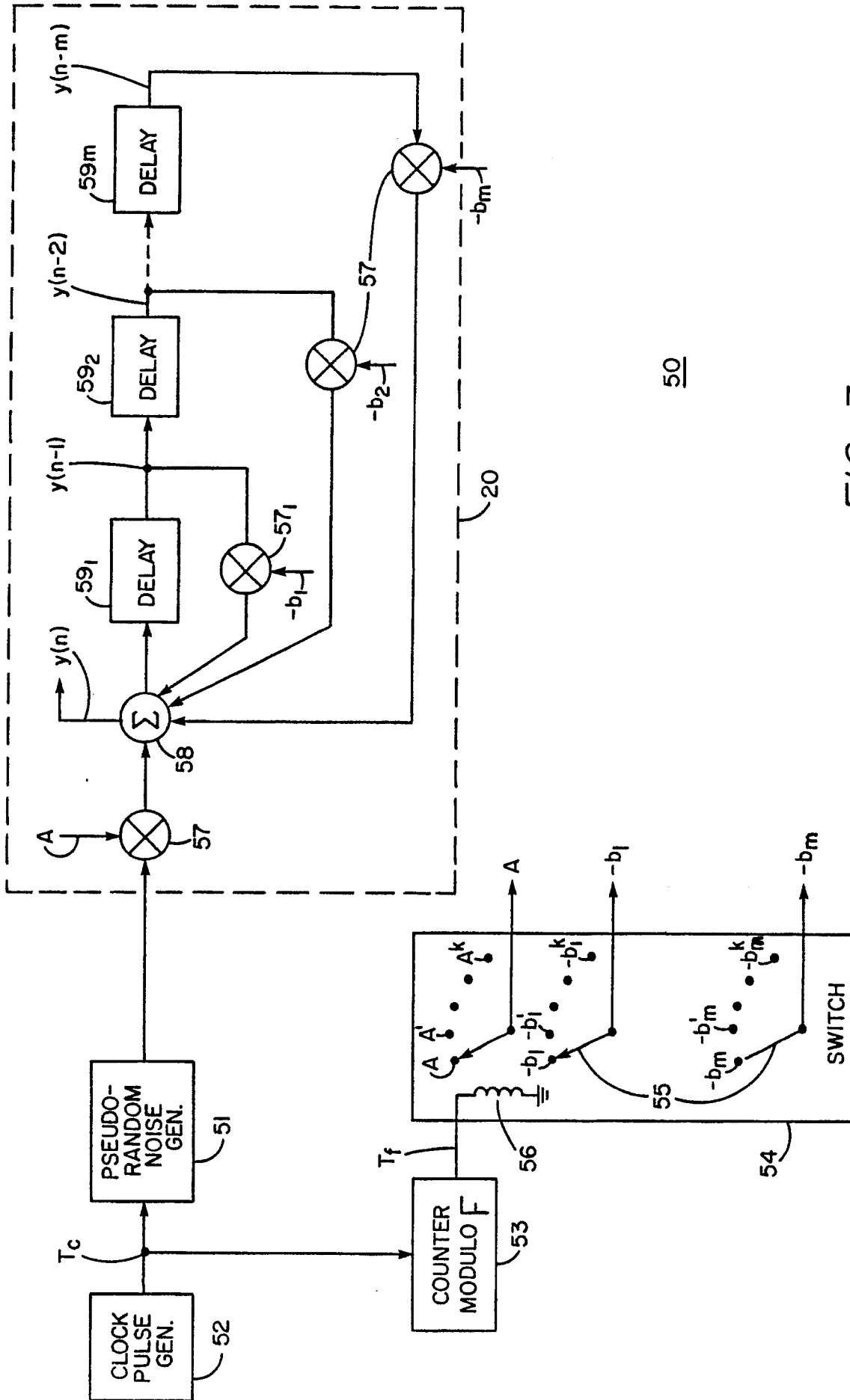


FIG. 3

