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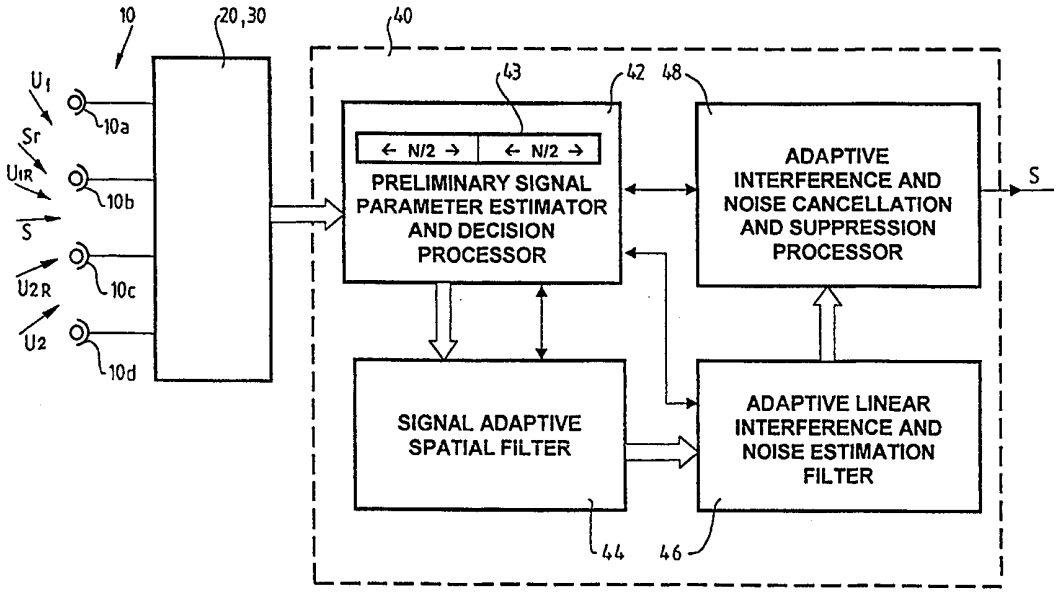
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(57) Abstract

A method of processing signals received from an array of sensors is disclosed comprising the steps of sampling and digitally converting the received signals and processing the digitally converted signals to provide an output signal, the processing including filtering the signals using a first adaptive filter arranged to enhance a target signal of the digitally converted signals and a second adaptive filter arranged to suppress an unwanted signal of the digitally converted signals and processing the filtered signals in the frequency domain to suppress the unwanted signal further.

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SIGNAL PROCESSING APPARATUS AND METHOD

Background and Field of the Invention

This invention relates to a method of signal processing and apparatus therefor.

In many situations, observations are made of the output of a multiple input and multiple output system such as phase array radar system, sonar array system or microphone array system, from which it is desired to recover the wanted signal alone with all the unwanted signals, including noise, cancelled or suppressed. For example, in a microphone array system for a speech recognition application, the objective is to enhance the target speech signal in the presence of background noise and competing speakers.

The most widely used approach to noise or interference cancellation in a multiple channel case was suggested by Widrow etc in "Adaptive Antenna Systems" Proc. IEEE, Vol.55 No.12, Dec. 1967 and "Signal Cancellation Phenomena in Antennas: causes and cures", IEEE Trans. Antennas Propag., Vol.AP30, May 1982. Also by L.J.Griffiths etc in " An Alternative Approach to Linearly Constrained Adaptive Beamforming". IEEE Trans. Antennas Propag. VolAP30, 1982. In these and other similar approaches, the signal processing apparatus separates the observed signal into a primary channel which comprises both the target signal and the interference signal and noise, and a secondary channel which comprises interference signal and noise alone. The interference signals and noise in the primary channel are estimated using an adaptive filter having the secondary channel signal as input, the estimated interference and noise signal being subtracted from the primary channel to obtain the desired target signal.

There are two major drawbacks of the above approaches. The first is that it is assumed that the secondary channel comprises interference signals and noise only. This assumption may not be correct in practice due to leakage of wanted signals into the secondary channel due to hardware imperfections and limited array dimension. The second is that it is assumed that the interference signals and noise can be estimated

accurately from the secondary channel. This assumption may also not be correct in practice because this will required a large number of degrees of freedom, this implying a very long filter and large array dimension. A very long filter leads to other problems such as rate of convergence and instability.

The first drawback will lead to signal cancellation. This degrades the performance of the apparatus. Depending on the input signal power, this degradation may be severe, leading to poor quality of the reconstructed speech because a portion of the desired signal is also cancelled by the filtering process. The second drawback will lead to poor interference and noise cancellation especially low frequency interference signals the wavelengths of which are many times the dimension of the array.

It is an object of the invention to provide an improved signal processing apparatus and method.

Summary of the Invention

According to the invention in a first aspect, there is provided a method of processing signals received from an array of sensors comprising the steps of sampling and digitally converting the received signals and processing the digitally converted signals to provide an output signal, the processing including filtering the signals using a first adaptive filter arranged to enhance a target signal of the digitally converted signals and a second adaptive filter arranged to suppress an unwanted signal of the digitally converted signals and processing the filtered signals in the frequency domain to suppress the unwanted signal further.

Further preferred features of the invention are recited in appendant claims 2-40.

According to the invention in a second aspect, there is provided a method of calculating a spectrum from a coupled signal comprising the steps of:

1) deriving a target signal component S and an interference signal component I from the coupled signal;

2) transforming the target and interference signal components into respective frequency domain equivalents F(S) and F(I);and

3) constructing the spectrum P(S) and P(I) of at least one equivalent in accordance with:

$$P(S)=|Real(F(S))| + |Imag(F(S))| + G[F(S)]*R(s)$$

$$P(I)=|Real(F(I))| + |Imag(F(I))| + G[F(I)]*R(i)$$

where Real and Imag refer to taking the absolute value of the real or imaginary part of the frequency domain equivalent R(s), R(i) are scalar adjustment factors and G[F(S)] and G[F(I)] are functions of F(S) and F(I) respectively.

According to the invention in a third aspect, there is provided a method of calculating a reverberation coefficient from a plurality of signals received from respective sensors in respective signal channels of a sensor array comprising the steps of:

1) calculating a correlation time delay between signals from a reference one of the channels and another one of the channels using an adaptive filter;

2) performing adaptive filtering, using a second adaptive filter, on the received signals;and

3) calculating a reverberation coefficient from the filter coefficients of the first and second filters.

According to the invention in a fourth aspect, there is provided a method of signal processing of a signal having wanted and unwanted components comprising the steps of:

1) processing the signal in the time domain with at least one adaptive filter to enhance the wanted signal and/or reduce the unwanted signal,

2) transforming the thus processed signal to the frequency domain; and

3) performing at least one unwanted signal reduction process in the frequency domain.

The invention extends to apparatus for performing the method of the aformentioned aspects.

Each aspect of the invention is usable independently of the others, for example in other signal processing apparatus which need not include other features of this invention as described.

The described embodiment of the invention discloses a method and apparatus to enhance an observed target signal from a predetermined or known direction of arrival. The apparatus cancels and suppresses the unwanted signals and noise from their coupled observation by the apparatus. An approach is disclosed to enhance the target signal in a more realistic scenario where both the target signal and interference signal and noise are coupled in the observed signals. Further, no assumption is made regarding the number or the direction of arrival of the interference signals.

The described embodiment includes an array of sensors e.g. microphones each defining a corresponding signal channel, an array of receivers with preamplifiers, an array of analog to digital converters for digitally converting observed signals and a digital signal processor that processes the signals. From the observed signals, the apparatus outputs an enhanced target signal and reduces the noise and interference signals. The apparatus allows a tradeoff between interference and noise suppression level and signal quality. No assumptions are make about the number of interference signals and the characteristic of the noise.

The digital signal processor includes a first set of adaptive filters which act as a signal spatial filter using a first channel as a reference channel. This filter removes the target signal "s" from the coupled signal and puts the remaining elements of the coupled signal, namely interference signals "u" and system noise "q" in an interference plus noise channel referred to as a Difference Channel. This filter also enhances the target signal "s" and puts this in another channel, referred to as the Sum Channel. The Sum Channel consists of the enhanced target signal "s" and the interference signals "u" and noise "q".

The target signal "s" may not be removed completely from the Difference Channel due to the sudden movement of the target speaker or of an object within the vicinity of the

speaker, so this channel may contain some residue target signal on occasions which can lead to some signal cancellation. However, the described embodiment greatly reduces this.

The signals from the Difference Channel are fed to a second adaptive filter set. This set of filters adaptively estimates the interference signals and noise in the Sum Channel.

The estimated signals are fed to an Interference Signal and Noise Cancellation and Suppression Processor which cancels and suppresses the noise and interference signals from the Sum Channel and outputs the enhanced target signal.

Updating of the parameters of the sets of adaptive filters is performed using a further processor termed a Preliminary Signal Parameters Estimator which receives the observed signal and estimates the reverberation level of the signal, the system noise level, the signal level, estimate signal detection thresholds and the angle of arrival of the signal. This information is used by the decision processor to decide if any parameter update is required.

One application of the described embodiment of the invention is speech enhancement in a car environment where the direction of the target signal with respect to the system is known. Yet another application is speech input for speech recognition applications. Again the direction of arrival of the signal is known.

Brief Description of the Drawings

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

Fig.1 illustrates a general scenario where the invention may be used.

Fig.2 is a schematic illustration of a general digital signal processing system embodying the present invention.

Fig.3 is a system level block diagram of the described embodiment of Fig. 2.

Fig. 4a-c is a flow chart illustrating the operation of the embodiment of Fig. 3.

Fig 5 illustrates a typical plot of nonlinear energy of a channel and the established thresholds.

Fig.6(a) illustrates a wavefront arriving from 40 degree off-boresight direction

Fig 6(b) represents a time delay estimator using an adaptive filter

Fig 6(c) shows the impulse response of the filter indicates a wave front from the boresight direction.

Fig 7 illustrates the reverberation level of the received signal over time.

Fig 8 shows the schematic block diagram the four channel Adaptive Spatial Filter.

Fig 9 shows the schematic block diagram of the Adaptive Interference and Noise Estimator of Fig. 3.

Fig 10 shows an input signal buffer.

Fig 11 shows the use of a Hanning Window on overlapping blocks of signals.

Fig. 12 illustrates a sudden rise of noise level of the nonlinear energy plot.

Fig. 13 illustrates the readjustment of the thresholds to reflect the sudden rise of noise energy level.

Detailed Description of the Embodiment of the Invention

FIG.1 illustrates schematically the operating environment of a signal processing apparatus 5 of the described embodiment of the invention, shown in a simplified example of a room. A target sound signal "s" emitted from a source s' in a known direction impinging on a sensor array, such as a microphone array 10 of the apparatus 5, is coupled with other unwanted signals namely interference signals u1, u2 from other sources A,B, reflections of these signals u1r, u2r and the target signal's own reflected signal sr. These unwanted signals cause interference and degrade the quality of the target signal "s" as received by the sensor array. The actual number of unwanted signals depends on the number of sources and room geometry but only three reflected (echo) paths and three direct paths are illustrated for simplicity of explanation. The sensor array 10 is connected to processing circuitry 20-60 and there will be a noise input q associated with the circuitry which further degrades the target signal.

An embodiment of signal processing apparatus 5 is shown in FIG.2. The apparatus observes the environment with an array of four sensors such as microphones 10a-10d. Target and noise/interference sound signals are coupled when impinging on each of the sensors. The signal received by each of the sensors is amplified by an amplifier 20a-d and converted to a digital bitstream using an analogue to digital converter 30a-d. The bit streams are feed in parallel to the digital signal processor 40 to be processed digitally. The processor provides an output signal to a digital to analogue converter 50 which is fed to a line amplifier 60 to provide the final analogue output.

FIG.3 shows the major functional blocks of the digital processor in more detail. The multiple input coupled signals are received by the four-channel microphone array 10a-10d, each of which forms a signal channel, with channel 10a being the reference channel. The received signals are passed to a receiver front end which provides the functions of amplifiers 20 and analogue to digital converters 30 in a single custom chip. The four channel digitized output signals are fed in parallel to the digital signal processor 40. The digital signal processor 40 comprises four sub-processors. They are (a) a Preliminary Signal Parameters Estimator and Decision Processor 42, (b) a Signal Adaptive Spatial Filter 44, (c) an Adaptive Linear Interference and Noise Estimator 46, and (d) an Adaptive Interference and Noise Cancellation and Suppression Processor 48. The basic signal flow is from processor 42, to processor 44, to processor 46, to processor 48. These connections being represented by thick arrows in Fig. 3. The filtered signal S is output from processor 48. Decisions necessary for the operation of the processor 40 are generally made by processor 42 which receives information from processors 44 - 48, makes decisions on the basis of that information and sends instructions to processors 44 - 48, through connections represented by thin arrows in Fig. 3.

It will be appreciated that the splitting of the processor 40 into the four component parts 42, 44, 46, 48 is essentially notional and is made to assist understanding of the operation of the processor. The processor 40 would in reality be embodied as a single multi-function digital processor performing the functions described under control of a program with suitable memory and other peripherals.

A flowchart illustrating the operation of the processors is shown in Figs 4a-c and this will firstly be described generally. A more detailed explanation of aspects of the processor operation will then follow.

The front end 20,30 processes samples of the signals received from array 10 at a predetermined sampling frequency, for example 16kHz. The processor 42 includes an input buffer 43 that can hold N such samples for each of the four channels. Upon initialization, the apparatus collects a block of N/2 new signal samples for all the channels at step 500, so that the buffer holds a block of N/2 new samples and a block of N/2 previous samples. The processor 42 then removes any DC from the new samples and preemphasizes or whitens the samples at step 502.

There then follows a short initialization period at step 504 in which the first 20 blocks of N/2 samples of signal after start-up are used to estimate the environment noise energy E_n and two detection thresholds, a noise threshold T_{n1} and a larger signal threshold T_{n2}, are calculated by processor 42 from E_n using scaling factors. During this short period, an assumption is made that no target signals are present. These signals do, however, continue to be processed, so that an initial Bark Scale system noise value may be derived at step 570, below.

After this initialisation period, the energies and thresholds update automatically as described below. The samples from the reference channel 10a are used for this purpose although any other channel could be used.

The total non-linear energy of the signal samples E_r is then calculated at step 506.

At step 508, it is determined if the signal energy E_r is greater than the signal threshold T_{n1}. If not, the environment noise E_n and the two thresholds are updated at step 510 using the new value of E_r calculated in step 506. The Bark Scale system noise B_n (see below) is also similarly updated via point F. The routine then moves to point B. If so, the signal is passed to a threshold adjusting sub-routine 512-518.

Steps 512-518 are used to compensate for abrupt changes in environment noise level which may capture the thresholds. A time counter is used to determine if the signal level shows a steady state increase which would indicate an increase in noise, since the speech target signal will show considerable variation over time and thus can be distinguished. This is illustrated in Fig. 12 in which a signal noise level rises from an initial level to a new level which exceeds both thresholds. At step 512 a time counter C_c is incremented. At step 514 C_c is checked against a threshold T_{cc}. If the threshold is not reached, the program moves to step 520 described below. If the threshold is reached, the estimated noise energy E_n is then increased at step 516 by a multiple σ and E_n, T_{n1} and T_{n2} are updated at step 518. The effect of this is illustrated in Fig. 13. The counter is reset and updating ceases when the the signal energy E_r is less than the second threshold T_{n2} as tested at step 520 below.

A test is made at step 520 to see if the estimated energy E_r in the reference channel 10a exceeds the second threshold T_{n2}. If so, a candidate target signal is deemed to be present. The apparatus only wishes to process candidate target signals that impinge on the array 10 from a known direction normal to the array, hereinafter referred to as the boresight direction, or from a limited angular departure therefrom, in this embodiment plus or minus 15 degrees. Therefore the next stage is to check for any signal arriving from this direction.

At step 524 two coefficients are established, namely a correlation coefficient C_x and a correlation time delay T_d, which together provide an indication of the direction from which the target signal arrived.

At step 526, two tests are conducted to determine if the candidate target signal is an actual target signal. First, the crosscorrelation coefficient C_x must exceed a predetermined threshold T_c and, second, the size of the time delay coefficient must be less than a value θ indicating that the signal has impinged on the array within the predetermined angular range. If these conditions are not met, the signal is not regarded as a target signal and the routine passes to point B. If the conditions are met, the routine passes to point A.

If at step 520, the estimated energy E_r in the reference channel 10a is found not to exceed the second threshold T_{n2}, the target signal is considered not to be present and the routine passes to point B via step 522 in which the counter C_c is reset. This is done since the second threshold at this point is above the level of the total signal energy E_r indicating that the threshold must be, consequently, above the environment noise energy level E_n and thus updating of E_n is no longer necessary.

Thus, the signal has, by points A and B, been preliminarily classified into a target signal (point A) or a noise signal (point B).

Following point A, the signal is subject to a further test at steps 528-532. At step 528, it is determined if the filter coefficients W_{su} of filter 44 have yet been updated. If not, the subsequent steps 530, 532 are skipped, since these rely on the coefficients of filter 44 for calculation purposes. If so, a reverberation coefficient C_{rv} which provides a measure of the degree of reverberation of the signal is calculated and at step 532 it is determined if C_{rv} exceeds a threshold T_{rv} If so, this indicates an acceptable level of reverberation in the signal and the routine passes to step 534 (target signal filtering). If not, the signal joins the path from point B to step 536 (non-target signal filtering).

The now confirmed target signal is fed to the Signal Adaptive Spatial Filter 44, the purpose of which is to enhance the target signal. The filter is instructed to perform adaptive filtering at steps 534 and 538, in which the filter coefficients W_{su} are adapted to provide a "target signal plus noise" signal in the reference channel and "noise only" signals in the remaining channels using the Least Mean Square (LMS) algorithm. The filter 44 output channel equivalent to the reference channel is for convenience referred to as the Sum Channel and the filter 44 output from the other channels, Difference Channels. The signal so processed will be, for convenience, referred to as A'.

If the signal is considered to be a noise signal, the routine passes to step 536 in which the signals are passed through filter 44 without the filter coefficients being adapted, to form the Sum and Difference channel signals. The signals so processed will be referred to for convenience as B'.

The effect of the filter 44 is to enhance the signal if this is identified as a target signal but not otherwise.

At step 540, an energy ratio R_{sd} between the Sum Channel and the Difference Channels is estimated by processor 42. At step 542 two tests are made. First, if the signals are A' signals from step 534, the routine passes to step 550. Second, for those signals for which $E_r > T_{n2}$ (i.e., high energy level), R_{sd} is compared to a threshold T_{sd}. If the ratio is lower than T_{sd}, this indicates probable noise but if higher, this may indicate that there has been some leakage of the target signal into the Difference channel, indicating the presence of a target signal after all. For such target signals the routine also passes to step 550. For all other non-target signals, the routine passes to step 544.

At steps 544-560, the signals are processed by the Adaptive Linear Interference and Noise Estimation Filter 46, the purpose of which is to reduce the unwanted signals. The filter 46, at step 544, is instructed to perform adaptive filtering on the non- target signals with the intention of adapting the filter coefficients to reducing the unwanted signal in the Sum channel to some small error value e_c.

To further prevent signal cancellation, the norm of the filter coefficients is calculated by processor 42 at step 546. If this norm exceeds a predetermined value $[T_{no}]$ at step 548, then the filter coefficients are scaled at step 549 to a reduced value.

In the alternative, at step 550, the target signals are fed to the filter 46 but this time, no adaptive filtering takes place, so the Sum and Difference signals pass through the filter.

An output of the Sum Channel signal without alteration is also passed through the filter 46.

The output signals from processor 46 are thus the Sum channel signal S_c (point C), filtered Difference signals D_c (point E) and the error signal e_c (point D). At step 562, a weighted average S(t) of the error signal e_c and the Sum Channel signal is calculated and the signals from the Difference channels D_c are Summed to form a single signal I(t).

These signals S(t) and I(t) are then collected for the new N/2 samples and the last N/2 samples from the previous block and a Hanning Window H_n is applied to the collected samples as shown in Fig. 10 to form vectors S_h and I_h. This is an overlapping technique with overlapping vectors S_n,I_n being formed from past and present blocks of N/2 samples continuously. This is illustrated in Fig. 11. A Fast Fourier Transform is then performed on the vectors S_h and I_h to transform the vectors into frequency domain equivalents S_f and I_f, at step 564.

At step 566 a modified spectrum is calculated for the transformed signals to provide "pseudo" spectrum values P_s and P_i and these values are warped into the same Bark Frequency Scale to provide Bark Frequency scaled values B_s and B_i at step 568.

The Bark value B_n of the system noise of the Sum Channel is updated at step 570 using B_s and the previous value of B_n, if the condition at step 508 is met (through path F). At start-up, B_n is initially calculated at this block whether or not the condition is met. At this time, there must be no target signal present, thus requiring a short initialization period after signal detection has begun, for this initial B_n value to be established.

A weighted combination By of B_n and B_i is then made at step 572 and this is combined with B_s to compute the Bark Scale nonlinear gain G_b at step 574.

G_b is then unwarped to the normal frequency domain to provide a gain value G at step 578 and this is then used at step 580 to compute an output spectrum S_{out} using the signal spectrum S_f from step 564. This gain-adjusted spectrum suppresses both the interference signals, the environmental noise and system noise.

An inverse FFT is then performed on the spectrum S_{out} at step 582 and the output signal is then reconstructed from the overlapping signals using the overlap add procedure at step 584.

Major steps in the above described flowchart will now be described in more detail.

NonLinear Energy and Threshold Estimation and Updating (STEPS 506,510)

The processor 42 estimates the energy output from a reference channel. In the four channel example described, channel 10a is used as the reference channel.

N/2 samples of the digitized signal are buffered into a shift register to form a signal vector of the following form:

$$X_r = \begin{bmatrix} X(0) \\ X(1) \\ \cdot \\ \cdot \\ \cdot \\ X(J-1) \end{bmatrix} \quad \ldots A.1$$

Where J = N/2. The size of the vector depends on the resolution requirement. In the preferred embodiment, J=256 samples.

The nonlinear energy of the vector is then estimated using the following equation:

$$E_r = \frac{1}{J-2} \sum_{i=1}^{J-2} X(i)^2 - X(i+1)\, X(i-1) \quad \ldots\ldots A.2$$

When the system is initialized, the average system and environment noise energy is estimated using the first 20 blocks of signal. A first order recursive filter is used to carry out this task as shown below:.

$$E_n^{K+1} = \alpha E_n^K + (1-\alpha) E_r^{K+1} \quad \ldots A.3$$

Where the superscript K is the block number and α is an empirically chosen weight between zero and one. In this embodiment, $\alpha = 0.9$.

Once the noise energy E_n is obtained, the two signal detection thresholds Tn1 and T_{n2} are established as follows:

$$T_{n1} = \delta_1 E_n \quad ....A.4$$

$$T_{n2} = \delta_2 E_n \quad ....A.5$$

δ_1 and δ_2 are scalar values that are used to select the thresholds so as to optimize signal detection and minimize false signal detection. As shown in Fig. 5, T_{n1} should be above the system noise level, with T_{n2} sufficient to be generally breached by the potential target signal. These thresholds may be found by trial and error. In this embodiment, δ_1=1.125 and δ_2=1.8 have been found to give good results.

Once the thresholds have been established, E_n may be updated after initialization in step 510 as follows:

$$If \quad E_r < T_{n1}$$
$$E_n = \alpha_n + (1-\alpha) E_r$$
$$Else$$
$$E_n = E_n$$
$$...A.6$$

The updated thresholds may then be calculated according to equations A.4 and A.5.

Time Delay Estimation (T_d) (STEP 524)

FIG 6A illustrates a single wave front impinging on the sensor array. The wave front impinges on sensor 10d first (A as shown) and at a later time impinges on sensor 10a (A' as shown), after a time delay t_d. This is because the signal originates at an angle of

40 degrees from the boresight direction. If the signal originated from the boresight direction, the time delay t_d will have been zero ideally.

Time delay estimation of performed using a tapped delay line time delay estimator included in the processor 42 which is shown in Fig. 6B. The filter has a delay element 600 ,having a delay $Z^{-L/2}$, connected to the reference channel 10a and a tapped delay line filter 610 having a filter coefficient W_{td} connected to channel 10d. Delay element 600 provides a delay equal to half of that of the tapped delay line filter 610. The outputs from the delay element is d(k) and from filter 610 is d'(k). The Difference of these outputs is taken at element 620 providing an error signal e(k) (where k is a time index used for ease of illustration). The error is fed back to the filter 610. The Least Mean Squares (LMS) algorithm is used to adapt the filter coefficient W_{td} as follows:

$$W_{td}(k+1) = W_{td}(k) + 2\mu_{td} S_{10d}(k)\, e(k) \quad ....B.1$$

$$W_{td}(k+1) = \begin{bmatrix} W_{td}^{0}(k+1) \\ W_{td}^{1}(k+1) \\ \cdot \\ \cdot \\ \cdot \\ W_{td}^{L_0}(k+1) \end{bmatrix} \quad .....B.2$$

$$S_{10d}(k) = \begin{bmatrix} S_{10d}^{0}(k) \\ S_{10d}^{1}(k) \\ \cdot \\ \cdot \\ \cdot \\ S_{10d}^{L_0}(k) \end{bmatrix} \quad .....B.3$$

$$e(k) = d(k) - d'(k) \quad \ldots\ldots B4$$

$$d'(k) = W_{td}(k)^T . S_{10d}(k) \quad \ldots\ldots B.5$$

$$\mu_{td} = \frac{\beta_{td}}{\|S_{10d}(k)\|} \quad \ldots\ldots B.6$$

where β_{td} is a user selected convergence factor $0<\beta_{td}\leq 2$, $\|\|$ denoted the norm of a vector, k is a time index, L_0 is the filter length.

The impulse response of the tapped delay line filter 620 at the end of the adaptation is shown in Fig. 6c. The impulse response is measured and the position of the peak or the maximum value of the impulse response relative to origin O gives the time delay T_d between the two sensors which is also the angle of arrival of the signal. In the case shown, the peak lies at the centre indicating that the signal comes from the boresight direction (T_d=0). The threshold θ at step 506 is selected depending upon the assumed possible degree of departure from the boresight direction from which the target signal might come. In this embodiment, θ is equivalent to
± 15°.

Normalized Cross Correlation Estimation C_x (STEP 524)

The normalized crosscorrelation between the reference channel 10a and the most distant channel 10d is calculated as follows:

Samples of the signals from the reference channel 10a and channel 10d are buffered into shift registers X and Y where X is of length J samples and Y is of length K samples, where J>K, to form two independent vectors X_r and Y_r:

$$X_r = \begin{bmatrix} x_r(1) \\ x_r(2) \\ \cdot \\ \cdot \\ \cdot \\ x_r(J) \end{bmatrix} \quad \ldots C.1$$

$$Y_r = \begin{bmatrix} y_r(1) \\ y_r(2) \\ \cdot \\ \cdot \\ \cdot \\ y_r(K) \end{bmatrix} \quad \ldots C.2$$

A time delay between the signals is assumed, and to capture this Difference, J is made greater than K. The Difference is selected based on angle of interest. The normalized cross-correlation is then calculated as follows:

$$C_x(l) = \frac{Y_r^T * X_{rl}}{\|Y_r\|\|X_{rl}\|} \quad \ldots . C.3$$

$$Where \ .. \ X_{rl} = \begin{bmatrix} X_r(l) \\ X_r(l+1) \\ \cdot \\ \cdot \\ \cdot \\ x_r(K+l-1) \end{bmatrix} \quad \ldots C.4$$

Where ^{T} represents the transpose of the vector and $\|\|$ represent the norm of the vector and l is the correlation lag. l is selected to span the delay of interest. For a sampling

frequency of 16kHz and a spacing between sensors 10a,10d of 18cm, the lag l is selected to be five samples for an angle of interest of 15°.

The threshold T_c is determined empirically. $T_c = 0.85$ is used in this embodiment.

Signal Reverberation Estimation C_{rv} (STEP 530)

The degree of reverberation of the received signal is calculated using the time delay estimator filter weight $[W_{td}]$ used in calculation of T_d above and the set of spatial filter weights $[W_{su}]$ from filter 44 (described below) as shown in the following equation:

$$C_{rv}^{m} = \frac{W_{td}^{T} W_{su}^{m}}{\|W_{td}\| \|W_{su}^{m}\|} \quad \ldots . D.1$$

Where ^{T} represents the transpose of the vector and M is the channel associated with the filter coefficient W_{su}. In this embodiment, three values for C_{rv}, one for each filter coefficient W_{su} are calculated. The largest is taken for subsequent processing.

The threshold T_{rv} used in step 506 is selected to ensure that the signal is selected as a target signal only when the level of reverberation is moderate, as illustrated in Fig. 7.

Adaptive Spatial Filter 44 (STEPS 534,536)

FIG.8 shows a block diagram of the Adaptive Linear Spatial Filter 44. The function of the filter is to separate the coupled target interference and noise signals into two types. The first, in a single output channel termed the Sum Channel, is an enhanced target signal having weakened interference and noise i.e. signals not from the target signal direction. The second, in the remaining channels termed Difference Channels, which in the four channel case comprise three separate outputs, aims to comprise interference and noise signals alone.

The objective is to adapt the filter coefficients of filter 44 in such a way so as to

enhanced the target signal and output it in the Sum Channel and at the same time eliminate the target signal from the coupled signals and output them into the Difference Channels.

The adaptive filter elements in filter 44 act as linear spatial prediction filters that predict the signal in the reference channel whenever the target signal is present. The filter stops adapting when the signal is deemed to be absent.

The filter coefficients are updated whenever the conditions of steps 504 and 506 are met, namely:

(i) The adaptive threshold detector detects the presence of signal;

(ii) The time delay estimator indicates that the signal arrived from the predetermined angle;

(iii) The normalized cross correlation of the signal exceeds the threshold; and

(iv) The reverberation level is low.

As illustrate in FIG.8, the digitized coupled signal X_0 from sensor 10a is fed through a digital delay element 710 of delay $Z^{-Lsu/2}$. Digitized coupled signals X_1,X_2,X_3 from sensors 10b,10c,10d are fed to respective filter elements 712,4,6. The outputs from elements 710,2,4,6 are Summed at Summing element 718, the output from the Summing element 718 being divided by four at divider element 719 to form the Sum channel output signal. The output from delay element 710 is also subtracted from the outputs of the filters 712,4,6 at respective Difference elements 720,2,4, the output from each Difference element forming a respective Difference channel output signal, which is also fed back to the respective filter 712,4,6. The function of the delay element 710 is to time align the signal from the reference channel 10a with the output from the filters 712,4,6.

The filter elements 712,4,6 adapt in parallel using the LMS algorithm given by Equations E.1..E.8 below, the output of the Sum Channel being given by equation E.1 and the output from each Difference Channel being given by equation E.6:

$$\hat{S}_c(k) = (\overline{S}(k) + \overline{X}_0(k))/4 \quad ...E.1$$

$$Where: \quad \overline{S}(k) = \sum_{m=1}^{M-1} \overline{S}_m(k) \quad ...E.2$$

$$\overline{S}_m(k) = (W_{su}^m(k))^T X_m(k) \quad ....E.3$$

Where m is 0,1,2...M-1, the number of channels, in this case 0..3 and ^{T} denotes the transpose of a vector.

$$X_m(k) = \begin{bmatrix} X_{1m}(k) \\ X_{2m}(k) \\ \cdot \\ \cdot \\ \cdot \\ X_{LSUm}(k) \end{bmatrix} \quad ...E.4$$

$$W_{su}^m(k) = \begin{bmatrix} W_{su1}^m(k) \\ W_{su2}^m(k) \\ \cdot \\ \cdot \\ \cdot \\ W_{su\ LSU}^m(k) \end{bmatrix} \quad ...E.5$$

Where $X_m(k)$ and $W_{su}^m(k)$ are column vectors of dimension $L_{su}*1$.

The weight $W_{su}^m(k)$ is updated using the LMS algorithm as follows:

$$\hat{d}_{cm}(k) = \overline{X}_0(k) - \overline{S}_m(k) \quad ...E.6$$

$$W_{su}^m(k+1) = W_{su}^m(k) + 2\mu_{su}^m X_m(k)\hat{d}_{cm}(k) \quad ....E.7$$

$$Where: \quad \mu_{su}^m = \frac{\beta_{su}}{\|X_m(k)\|} \quad ...E.8$$

and where β_{su} is a user selected convergence factor $0<\beta_{su}\leq 2$, $\|\|$ denoted the norm of a vector and k is a time index.

Calculation of Energy Ratio R_{sd}(step 540)

This is performed as follows:

$$\hat{S}_c = \begin{bmatrix} \hat{S}_c(0) \\ \hat{S}_c(1) \\ \cdot \\ \cdot \\ \cdot \\ \hat{S}_c(J-1) \end{bmatrix} \quad ...F.1$$

$$\hat{D}_c = \begin{bmatrix} \hat{d}_c(0) \\ \hat{d}_c(1) \\ \cdot \\ \cdot \\ \cdot \\ \hat{d}_c(J-1) \end{bmatrix} = \begin{bmatrix} \hat{d}_{c1}(0) \\ \hat{d}_{c1}(1) \\ \cdot \\ \cdot \\ \cdot \\ \hat{d}_{c1}(J-1) \end{bmatrix} + \begin{bmatrix} \hat{d}_{c2}(0) \\ \hat{d}_{c2}(1) \\ \cdot \\ \cdot \\ \cdot \\ \hat{d}_{c2}(J-1) \end{bmatrix} + \begin{bmatrix} \hat{d}_{c3}(0) \\ \hat{d}_{c3}(1) \\ \cdot \\ \cdot \\ \cdot \\ \hat{d}_{c3}(J-1) \end{bmatrix} \quad ...F.2$$

J=N/2, the number of samples, in this embodiment 256.

Where E_{SUM} is the sum channel energy and E_{DIF} is the difference channel energy.

$$E_{SUM} = \frac{1}{J-2}\sum_{j=1}^{J-2} \hat{S}_c(j)^2 - \hat{S}_c(j-1)\,\hat{S}_c(j-1) \quad .....F.3$$

$$E_{DIF} = \frac{1}{3(J-2)} \sum_{j=1}^{J-2} \hat{d}_c(j)^2 - \hat{d}_c(j-1)\ \hat{d}_c(j-1) \quad .....F.4$$

$$R_{sd} = \frac{E_{SUM}}{E_{DIF}} \quad ...F.5$$

The energy ratio between the Sum Channel and Difference Channel (R_{sd}) must not exceed a predetermined threshold. In the four channel case illustrated here the threshold isdetermined to be about 1.5.

Adaptive Interference and Noise Estimation Filter 46 (STEPS 544,550)

FIG.9 shows a schematic block diagram of the Adaptive Interference and Noise Estimation Filter 46. This filter estimates the noise and interference signals and subtracts them from the Sum Channel so as to derive an output with reduced noise and interference.

The filter 46 takes outputs from the Sum and Difference Channels of the filter 44 and feeds the Difference Channel Signals in parallel to another set of adaptive filter elements 750,2,4 and feeds the Sum Channel signal to a corresponding delay element 756. The outputs from the three filter elements 750,2,4 are subtracted from the output from delay element 756 at Difference element 758 to form an error output e_c, which is also fed back to the filter elements 750,2,4. The output from filter element 756 is also passed directly as an output, as are the outputs from the three filter elements 750,2,4.

Again, the Least Mean Square algorithm (LMS) is used to adapt the filter coefficients Wuq as follows:

$$e_c(k) = \hat{S}_c(k) - \sum_{m=1}^{M-1} \hat{d}_{cm}(k) \quad \ldots G.1$$

$$\text{Where: } \hat{d}_{cm}(k) = W_{uq}^{m}(k)^T \cdot Y^m(k) \quad \ldots G.2$$

$$Y^m(k) = \begin{bmatrix} \hat{d}_{c1m}(k) \\ \hat{d}_{c2m}(k) \\ . \\ . \\ \hat{d}_{cLuqm}(k) \end{bmatrix} \quad \ldots G.3$$

$$W_{uq}^{m}(k+1) = W_{uq}^{m}(k) + 2\mu_{uq}^{m} Y^m(k)\, e_c(k) \quad \ldots G.4$$

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$$\mu_{uq}^{m} = \frac{\beta_{uq}}{\|Y^m\|} \quad \ldots . G.5$$

and where β_{uq} is a user selected convergence factor $0<\beta_{uq}\leq 2$ and where m is 0,1,2...M-1, the number of channels, in this case 0..3.

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Calculation of Norm of filter coefficients (step 546)

The norms of the coefficients of filters 750,2,4 are also constrained to be smaller than a predetermined value. The rationale for imposing this constraint is because the norm 15 of the filter coefficients will be large if a target signal leaks into the Difference Channel. Scaling down the norm value of the filter coefficients will reduce the effect of signal cancellation.

This is calculated as follows:

$$If: \quad \|W_{uq}^{m}\| > T_{no} \quad ....G.6$$

$$Then: \quad W_{uq}^{m} = \frac{W_{uq}^{m}}{\|W_{uq}^{m}\|} * C_{no} \quad ....G.7$$

Where m is 1,2...M-1, the channels having W_{uq} filters. T_{no} is a predetermined threshold and C_{no} is a scaling factor, both of which can be estimated empirically.

The output e_c from equation F.1 is almost interference and noise free in an ideal situation. However, in a realistic situation, this can not be achieved. This will cause signal cancellation that degrades the target signal quality or noise or interference will feed through and this will lead to degradation of the output signal to noise and interference ratio. The signal cancellation problem is reduced in the described embodiment by use of the Adaptive Spatial Filter 44 which reduces the target signal leakage into the Difference Channel. However, in cases where the signal to noise and interference is very high, some target signal may still leak into these channels.

To further reduce the target signal cancellation problem and unwanted signal feed through to the output, The output signals from processor 46 are fed into the Adaptive NonLinear Interference and Noise Suppression Processor 48 as described below.

Adaptive NonLinear Interference and Noise Suppression Processor 48 (STEPS 562-584)

This processor processes input signals in the frequency domain coupled with the well-known overlap add block processing technique.

STEP 562: The output signal (e_c) and the Sum Channel output signal (S_c) combined as a weighted average as follows:

$$S(t) = W_1 S_c(t) + W_2 e_c(t) \quad ....H.1$$

The weights (W_1,W_2) can be empirically chosen to minimize signal cancellation or improve unwanted signal suppression. In this embodiment, $W_1=W_2=0.5$.

This combined signal is buffered into a memory as illustrated in FIG.10. The buffer consists of N/2 of new samples and N/2 of old samples from the previous block. Similarly, the unwanted signals from the Difference Channel are summed in accordance with the following and buffered the same way as the Sum Channel:

$$I(t) = \sum_{i=1}^{M-1} D_{ci} \quad ....H.2A$$

$$Where: \quad D_{ci} = \begin{bmatrix} d_{ci}(0) \\ d_{ci}(1) \\ . \\ . \\ d_{ci}(J-1) \end{bmatrix} \quad ...H.2B$$

Where i=1,2...M-1 and M is the number of channels, in this case M=4.

A Hanning Window is then applied to the N samples buffered signals as illustrated in FIG.11 expressed mathematically as follows:

$$S_h = \begin{bmatrix} S_{(t+1)} \\ S_{(t+2)} \\ . \\ . \\ . \\ S_{(t+N)} \end{bmatrix} . \; H_n \quad ...H.3$$

$$I_h = \begin{bmatrix} I_{(t+1)} \\ I_{(t+2)} \\ \cdot \\ \cdot \\ \cdot \\ I_{(t+N)} \end{bmatrix} \cdot H_n \quad \ldots H.4$$

Where (H_n) is a Hanning Window of dimension N, N being the dimension of the buffer. The "dot" denotes point by point multiplication of the vectors. t is a time index.

Step 5.64: The resultant vectors [S_h] and [I_h] are transformed into the frequency domain using Fast Fourier Transform algorithm as illustrated in equations H.5 and H.6 below:

$$S_f = FFT(S_h) \quad \ldots . H.5$$

$$I_f = FFT(I_h) \quad \ldots . H.6$$

Step 566: A modified spectrum is then calculated, which is illustrated in Equations H.7 and H.8:

$$P_s = |Real(S_f)| + |Imag(S_f)| + F(S_f) * r_s \quad \ldots H.7$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + F(I_f) * r_i \quad \ldots H.8$$

Where "Real" and "Imag" refer to taking the absolute values of the real and imaginary parts, r_s and r_i are scalars and $F(S_f)$ and $F(I_f)$ denotes a function of S_f and I_f respectively.

One preferred function F using a power function is shown below in equations H.9 and

H.10 where "Conj" denotes the complex conjugate:

$$P_s = |Real(S_f)| + |Imag(S_f)| + (S_f * conj(S_f)) * r_s \quad ...H.9$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + (I_f * conj(I_f)) * r_i \quad ...H.10$$

A second preferred function F using a multiplication function is shown below in equations H.11 and H.12:

$$P_s = |Real(S_f)| + |Imag(S_f)| + |Real(S_f)| * |Imag(S_f)| * r_s \quad H.11$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + |Real(I_f)| * |Imag(I_f)| * r_i \quad H.12$$

The values of the scalars (r_s and r_i) control the tradeoff between unwanted signal suppression and signal distortion and may be determined empirically. (r_s and r_i) are calculated as $1/(2^{vs})$ and $1/(2^{vi})$ where vs and vi are scalars. In this embodiment, vs=vi is chosen as 8 giving $r_s=r_i = 1/256$. As vs,vi reduce, the amount of suppression will increase.

Step 568: The Spectra (P_s) and (P_i) are warped into (Nb) critical bands using the Bark Frequency Scale [see Lawrence Rabiner and Bing Hwang Juang, Fundamentals of Speech Recognition, Prentice Hall 1993]. The number of Bark critical bands depend on the sampling frequency used. For a sampling of 16Khz, there will be Nb = 25 critical bands. The warped Bark Spectrum of (P_s) and (P_i) are denoted as (B_s) and (B_i).

Step 570: A Bark Spectrum of the system noise and environment noise is similarly computed and is denoted as (B_n). B_n is first established during system initialization as $B_n=B_s$ and continues to be updated when no target signal is detected (step 508) by the system i.e. any silence period. B_n is updated as follows:

If $E_r < T_{n1}$

$B_n = \alpha B_n + (1-\alpha)B_s$

Else

$B_n = B_n$

.....H.13

Where $0<\alpha<1$; in this embodiment, $\alpha=0.9$

Steps 572,574: Using (B_s, B_i and B_n) a nonlinear technique is used to estimate a gain (G_b) as follows :

First the unwanted signal Bark Spectrum is combined with the system noise Bark Spectrum using an appropriate weighting function as illustrate in Equation J.1.

$$B_y = \Omega_1 B_i + \Omega_2 B_n \quad ...J.1$$

Ω_1 and Ω_2 are weights which can be chosen empirically so as to maximize unwanted signals and noise suppression with minimize signal distortion.

Follow that a post signal to noise ratio is calculated using Equations J.2 and J.3 below:

$$R_{po} = \frac{B_s}{B_y} \quad ...J.2$$

$$R_{pp} = R_{po} - I_c \quad ...J.3$$

The division in equation J.2 means element by element division and not vector division. R_{po} and R_{pp} are column vectors of dimension Nb*1, Nb being the dimension of the Bark Scale Critical Frequency Band and I_c is a column unity vector of dimension Nb*1 as shown below:

$$R_{po} = \begin{bmatrix} r_{po(1)} \\ r_{po(2)} \\ \cdot \\ \cdot \\ \cdot \\ r_{po(N_b)} \end{bmatrix} \quad \ldots J.4$$

$$R_{pp} = \begin{bmatrix} r_{pp(1)} \\ r_{pp(2)} \\ \cdot \\ \cdot \\ \cdot \\ r_{pp(N_b)} \end{bmatrix} \quad \ldots J.5$$

$$I_c = \begin{bmatrix} 1 \\ 1 \\ \cdot \\ \cdot \\ \cdot \\ 1 \end{bmatrix} \quad \ldots J.6$$

If any of the r_{pp}(nb)elements of R_{pp} are less than zero, they are set equal to zero.

Using the Decision Direct Approach [see Y. Ephraim and D. Malah: Speech Enhancement Using Optimal NonLinear Spectrum Amplitude Estimation; Proc. IEEE International Conference Acoustics Speech and Signal Processing (Boston) 1983, pp1118-1121.], the a-priori signal to noise ratio R_{pr} is calculated as follows:

$$R_{pr} = (1-\beta_i) * R_{pp} + \beta_i \frac{B_o}{B_y} \quad \ldots J.7$$

The division in Equation J.7 means element by element division. B_o is a column vector of dimensions Nb*1 and denotes the output signal Bark Scale Bark Spectrum from the previous block $B_o=G_b.B_s$ (see Eqn J.15) (B_o initially is zero). R_{pr} is also a column vector of dimension Nb*1. The value of β_i is given in Table 1 below:

i	β
1	0.01625
2	0.01225
3	0.245
4	0.49
5	0.98

TABLE 1

The value i is set equal to 1 on the onset of a signal and the β value is therefore equal to 0.01625. Then the i value will count from 1 to 5 on each new block of N/2 samples processed and stay at 5 until the signal is off. The i will start from 1 again at the next signal onset and the β is taken accordingly.

Instead of β being constant, in this embodiment β is made variable and starts at a small value at the onset of the signal to prevent suppresion of the target signal and increases, preferably exponentially, to smooth R_{pr}.

From this, R_{rr} is calculated as follows:

$$R_{rr} = \frac{R_{pr}}{I_c + R_{pr}} \quad \ldots J.8$$

The division in Equation J.8 is again element by element. R_{rr} is a column vector of dimension Nb*1.

From this, L_x is calculated:

$$L_X = R_{rr} . R_{po} \quad \ldots J.9$$

The value of L_x is limited to Pi (≈3.14). The multiplication in Equation J.9 means element by element multiplication. L_x is a column vector of dimension Nb*1 as shown below:

$$L_X = \begin{bmatrix} l_x(1) \\ l_x(2) \\ . \\ . \\ l_x(nb) \\ . \\ . \\ l_x(Nb) \end{bmatrix} \quad \ldots J.10$$

A vector L_y of dimension Nb*1 is then defined as:

$$L_Y = \begin{bmatrix} l_y(1) \\ l_y(2) \\ . \\ . \\ l_y(nb) \\ . \\ . \\ l_y(Nb) \end{bmatrix} \quad \ldots J.11$$

Where nb = 1,2...Nb. Then L_y is given as:

$$l_y(nb) = \exp\left(\frac{E(nb)}{2}\right) \quad \ldots J.12$$

and

$$E(nb) = -0.57722 - \log(l_x(nb)) + l_x(nb) - (l_x(nb))^2/4 + l_x(nb)^3/8 = l_x(nb)^4/96 \ldots$$

$$\ldots J.13$$

E(nb) is truncated to the desired accuracy. L_y can be obtained using a table look-up approach to reduce computational load.

Finally, the Gain G_b is calculated as follows:

$$G_b = R_{rr} . L_y \quad \ldots J.14$$

The "dot" again implies element by element multiplication. G_b is a column vector of dimension Nb*1 as shown:

$$G_b = \begin{bmatrix} g(1) \\ g(2) \\ . \\ . \\ g(nb) \\ . \\ . \\ g(Nb) \end{bmatrix} \quad \ldots J.15$$

Step 578: As G_b is still in the Bark Frequency Scale, it is then unwarped back to the normal linear frequency scale of N dimensions. The unwarped G_b is denoted as G.

The output spectrum with unwanted signal suppression is given as:

$$\overline{S}_f = G.S_f \quad \ldots J.16$$

The "dot" again implies element by element multiplication.

Step 580: The recovered time domain signal is given by:

$$\overline{S}_t = Real(IFFT(\overline{S}_f)) \quad \ldots J.17$$

IFFT denotes an Inverse Fast Fourier Transform, with only the Real part of the inverse transform being taken.

Step 584: Finally, the output time domain signal is obtained by overlap add with the previous block of output signal :

$$\hat{S}_t = \begin{bmatrix} \overline{S}_t(1) \\ \overline{S}_t(2) \\ \cdot \\ \cdot \\ \cdot \\ \overline{S}_t(N/2) \end{bmatrix} + \begin{bmatrix} Z_t(1) \\ Z_t(2) \\ \cdot \\ \cdot \\ \cdot \\ Z_t(N/2) \end{bmatrix} \quad \ldots J.18$$

$$Where: \quad Z_t = \begin{bmatrix} \overline{S}_{t-1}(1+N/2) \\ \overline{S}_{t-1}(2+N/2) \\ \cdot \\ \cdot \\ \cdot \\ \overline{S}_{t-1}(N) \end{bmatrix} \quad \ldots J.19$$

The embodiment described is not to be construed as limitative. For example, there can be any number of channels from two upwards. Furthermore, as will be apparent to one skilled in the art, many steps of the method employed are essentially discrete and may be employed independently of the other steps or in combination with some but not all of the other steps. For example, the adaptive filtering and the frequency domain processing may be performed independently of each other and the frequency domain processing steps such as the use of the modified spectrum, warping into the Bark scale and use of the scaling factor β can be viewed as a series of independent tools which need not all be used together.

Use of first, second etc. in the claims should only be construed as a means of identification of the integers of the claims, not of process step order. Any novel feature or combination of features disclosed is to be taken as forming an independent invention whether or not specifically claimed in the appendant claims of this application as initially filed.

CLAIMS

1. A method of processing signals received from an array of sensors comprising the steps of sampling and digitally converting the received signals and processing the digitally converted signals to provide an output signal, the processing including filtering the signals using a first adaptive filter arranged to enhance a target signal of the digitally converted signals and a second adaptive filter arranged to suppress an unwanted signal of the digitally converted signals and processing the filtered signals in the frequency domain to suppress the unwanted signal further.

2. A method as claimed in claim 1 further comprising the step of determining a signal energy from the signals and determining a noise energy from the signal energy.

3. A method as claimed in claim 2 wherein the signal energy is determined by buffering N/2 samples of the digitized signal into a shift register to form a signal vector of the following form:

$$X_r = \begin{bmatrix} X(0) \\ X(1) \\ \cdot \\ \cdot \\ \cdot \\ X(J-1) \end{bmatrix}$$

Where J = N/2; and estimating the signal energy using the following equation:

$$E_r = \frac{1}{J-2} \sum_{i=1}^{J-2} X(i)^2 - X(i+1)\ X(i-1)$$

where E_r is the signal energy.

4. A method as claimed in claim 2 or claim 3 wherein the noise energy is determined by measuring the signal energy E_r of blocks of the digitally converted signals and calculating the noise nergy E_n in accordance with:

$$E_n^{K+1} = \alpha E_n^K + (1-\alpha) E_r^{K+1}$$

Where the superscript K is the block number and α is an empirically chosen weight.

5. A method as claimed in any one of claims 2 to 4 further comprising the step of determining a noise threshold from the noise energy and updating the noise energy and noise threshold when the signal energy is below the noise threshold.

6. A method as claimed in claim 5 further comprising the step of determining if a target signal is present by comparing the signal energy to a signal threshold.

7. A method as claimed in claim 6 further comprising the step of determining the signal threshold from the noise energy and updating the signal threshold when the signal energy is below the noise threshold.

8. A method as claimed in any one of claims 5 to 7 wherein the noise threshold T_{n1} is determined in accordance with:

$$T_{n1} = \delta_1 E_n$$

Where δ_1 is an empirically chosen value.

9. A method as claimed in claim 6 or claim 7 wherein the signal threshold T_{n2} is determined in accordance with:

$$T_{n2} = \delta_2 E_n$$

Where δ_2 is an empirically chosen value.

10. A method as claimed in any one of the preceding claims further comprising the step of determining the direction of arrival of the target signal.

11. A method as claimed in claim 10 further comprising the step of processing the

signals from two spaced sensors of the array with a third adaptive filter to determine said direction of arrival.

12. A method as claimed in claim 10 or claim 11 further comprising the step of treating the signal as an unwanted signal if the signal has not impinged on the array from within a selected angular range.

13. A method as claimed in any one of the preceding claims further comprising the step of calculating a measure of the cross-correlation of signals from two spaced sensors of the array and treating the signal as an unwanted signal if the degree of cross correlation is less than a selected value.

14. A method as claimed in claim 11 further comprising the step of calculating a measure of reverberation of the signal from filter weights of the first and third adaptive filters.

15. A method as claimed in claim 14 wherein the reverberation measure C_{rv} is calculated in accordance with

$$C_{rv} = \frac{W_{td}^T W_{su}}{\|W_{td}\| \|W_{su}\|}$$

where T denotes the transpose of a vector, W_{su} is the filter coefficient of the first filter and W_{td} is the filter coefficient of the third filter.

16. A method as claimed in claim 14 or claim 15 further comprising the step of treating the signal as an unwanted signal if the reverberation measure indictates a degree of reverberation in excess of a selected value.

17. A method as claimed in any one of the preceding claims further comprising the step of controlling the operation of the first filter to perform adaptive filtering only when

a said target signal is deemed to be present.

18. A method as claimed in any one of the preceding claims wherein the first adaptive filter has a plurality of channels receiving as input the digitized signals and providing as output a sum and at least one difference signal, the difference signal channels including filter elements having corresponding filter weights.

19. A method as claimed in claim 18 further comprising the step of calculating a ratio of the energy in the sum and difference channels.

20. A method as claimed in claim 19 further comprising the step of treating the signal as including a said target signal if the ratio indicates that the energy in the sum channel is greater than the energy in the difference channels by more than a selected factor.

21. A method as claimed in claim 20 further comprising the step of treating the signal as including a said target signal only if the signal energy exceeds a threshold.

22. A method as claimed in any one of the preceding claims further comprising the step of controlling the operation of the second filter to perform adaptive filtering only when a said target signal is deemed not to be present.

23. A method as claimed in any one of the preceding claims wherein the first adaptive filter has a plurality of channels receiving input signals from the first adaptive filter and providing as output a sum signal received from the first adaptive filter, an error signal and at least one difference signal, the difference signal channels including further filter elements having corresponding further filter weights.

24. A method as claimed in claim 23 further comprising the step of scaling the further filter weights if the norms of the further filter weights exceed a threshold.

25. A method as claimed in claim 23 or claim 24 further comprising the step of

combining the sum signal and the error signal to form a single signal S(t) of the form:

$$S(t) = W_1 S_c(t) + W_2 e_c(t)$$

where $S_c(t)$ is the sum signal at time t, $e_c(t)$ is the error signal at time t and W_1 and W_2 are weight values.

26. A method as claimed in claim 25 further comprising the step of combining the difference signals to form a single signal.

27. A method as claimed in claim 25 or claim 26 further comprising the step of applying a Hanning window to the single signals.

28. A method as claimed in any one of the preceding claims further comprising the step of transforming the filtered signals into two frequency domain signals a desired signal S_f and an interference signal I_f, , processing the transformed signals to provide a gain for the desired signal and transforming the gain modified desired signal back to the time domain to provide an output.

29. A method as claimed in claim 28 wherein the processing step comprises the step of forming spectra for the frequency domain signals.

30. A method as claimed in claim 29 whererin the spectra are modified spectra P_s, P_i of the desired signal and the interference signal of the form:

$$P_s = |Real(S_f)| + |Imag(S_f)| + F(S_f) * r_s$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + F(I_f) * r_i$$

Where "Real" and "Imag" refer to taking the absolute values of the real and imaginary parts, r_s and r_i are scalars and $F(S_f)$ and $F(I_f)$ denotes a function of S_f and I_f respectively.

31. A method as claimed in claim 30 wherein the function is a power function.

32. A method as claimed in claim 31 wherein the spectra are of the form, where "Conj" denotes the complex conjugate:

$$P_i = |Real(I_f)| + |Imag(I_f)| + (I_f * conj(I_f)) * r_i$$

$$P_s = |Real(S_f)| + |Imag(S_f)| + (S_f * conj(S_f)) * r_s$$

33. A method as claimed in claim 30 wherein the function is a multiplication function

34. A method as claimed in claim 33 wherein the spectra are of the form:

$$P_s = |Real(S_f)| + |Imag(S_f)| + |Real(S_f)| * |Imag(S_f)| * r_s$$

$$P_i = |Real(I_f)| + |Imag(I_f)| + |Real(I_f)| * |Imag(I_f)| * r_i$$

35. A method as claimed in any one of claims 29 to 34 wherein the processing step includes the step of warping the signal and interference spectra into a Bark scale to form corresponding signal and interference Bark spectra.

36. A method as claimed in claim 35 wherein the processing step further includes the step of calculating a system noise Bark spectrum.

37. A method as claimed in claim 36 further comprising the step of combining the interference Bark spectrum and the system noise Bark spectrum to form a combined noise Bark spectrum.

38. A method as claimed in claim 37 wherein the combined noise Bark spectrum B_y is of the following form:

$$B_y = \Omega_1 B_i + \Omega_2 B_n$$

where Ω_1 and Ω_2 are weighting values B_i is the interference Bark spectrum and B_n is the system noise Bark spectrum.

39. A method as claimed in any one of claims 29 to 38 further comprising the step of calculating a signal to noise ratio from the spectra and deriving the gain from the signal to noise ratio.

40. A method as claimed in claim 39 further comprising the step of modifying the signal to noise ratio with a scaling factor which gradually changes from a first value at onset of the signal to a second value at which the scaling factor remains as the signal continues, until the signal ceases at which time the scaling factor is rest to the first value.

41. A method as claimed in claim 40 wherein the scaling factor changes in a plurality of steps.

42. A method as claimed in claim 40 or claim 41 wherein the scaling factor changes exponentially.

43. A method of calculating a spectrum from a coupled signal comprising the steps of:

1) deriving a target signal component S and an interference signal component I from the coupled signal;

2) transforming the target and interference signal components into respective frequency domain equivalents F(S) and F(I);and

3) constructing the spectrum P(S) and P(I) of at least one equivalent in accordance with:

$P(S)=|\text{Real}(F(S))| + |\text{Imag}(F(S))| + G[F(S)]*R(S)$

$P(I)=|\text{Real}(F(I))| + |\text{Imag}(F(I))| + G[F(I)]*R(I)$

Where Real and Imag refer to taking the absolute value of the real or imaginary part of the frequency domain equivalent R(S), R(I) are scalar adjustment factors and G[F(S)] and G[F(I)] are functions of F(S) and F(I) respectively.

44. A method of calculating a reverberation coefficient from a plurality of signals received from respective sensors in respective signal channels of a sensor array comprising the steps of:

1) calculating a correlation time delay between signals from a reference one of the channels and another one of the channels using an adaptive filter;

2) performing adaptive filtering, using a second adaptive filter, on the received signals;and

3) calculating a reverberation coefficient from the filter coefficients of the first and second filters.

45. A method of signal processing of a signal having wanted and unwanted components comprising the steps of:

1) processing the signal in the time domain with at least one adaptive filter to enhance the wanted signal and/or reduce the unwanted signal,

2) transforming the thus processed signal to the frequency domain; and

3) performing at least one unwanted signal reduction process in the frequency domain.

46. Apparatus for performing the method of any one of the preceding claims.

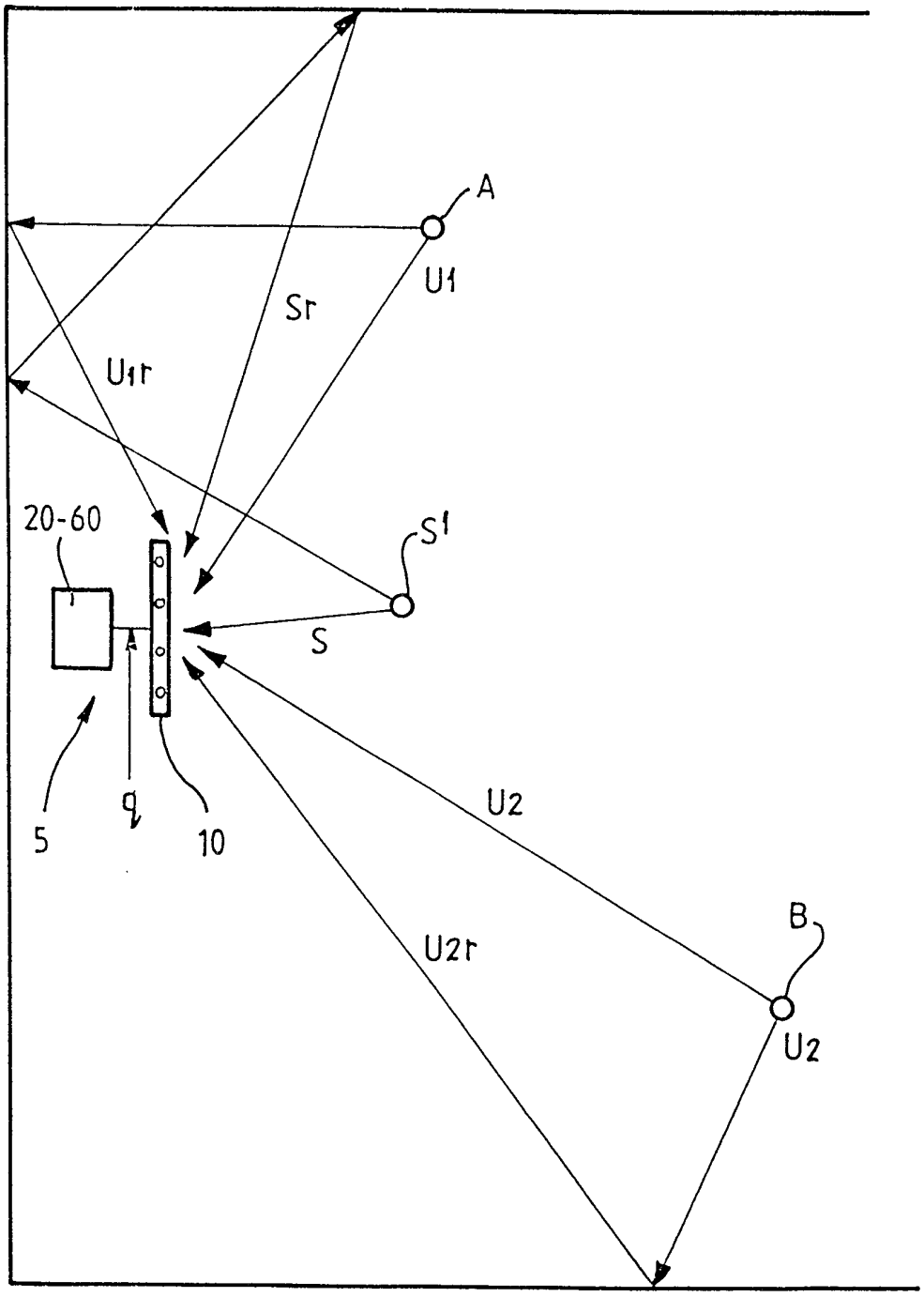
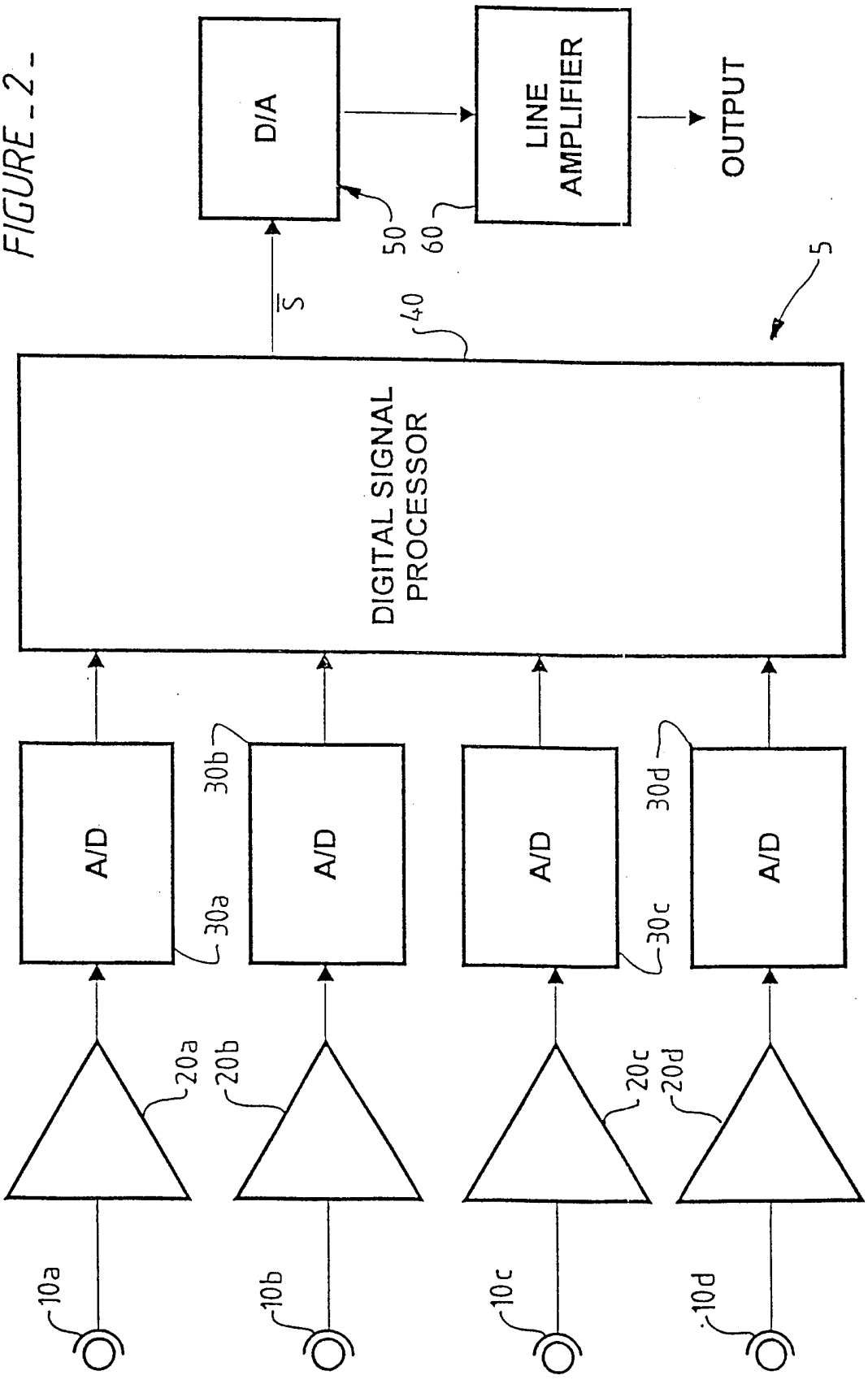


FIGURE _ 1 _

FIGURE - 2 -


10a
10b
10c
10d
20a
20b
20c
20d
A/D
A/D
A/D
A/D
30a
30b
30c
30d
DIGITAL SIGNAL PROCESSOR
40
5
S
D/A
50
LINE AMPLIFIER
60
OUTPUT

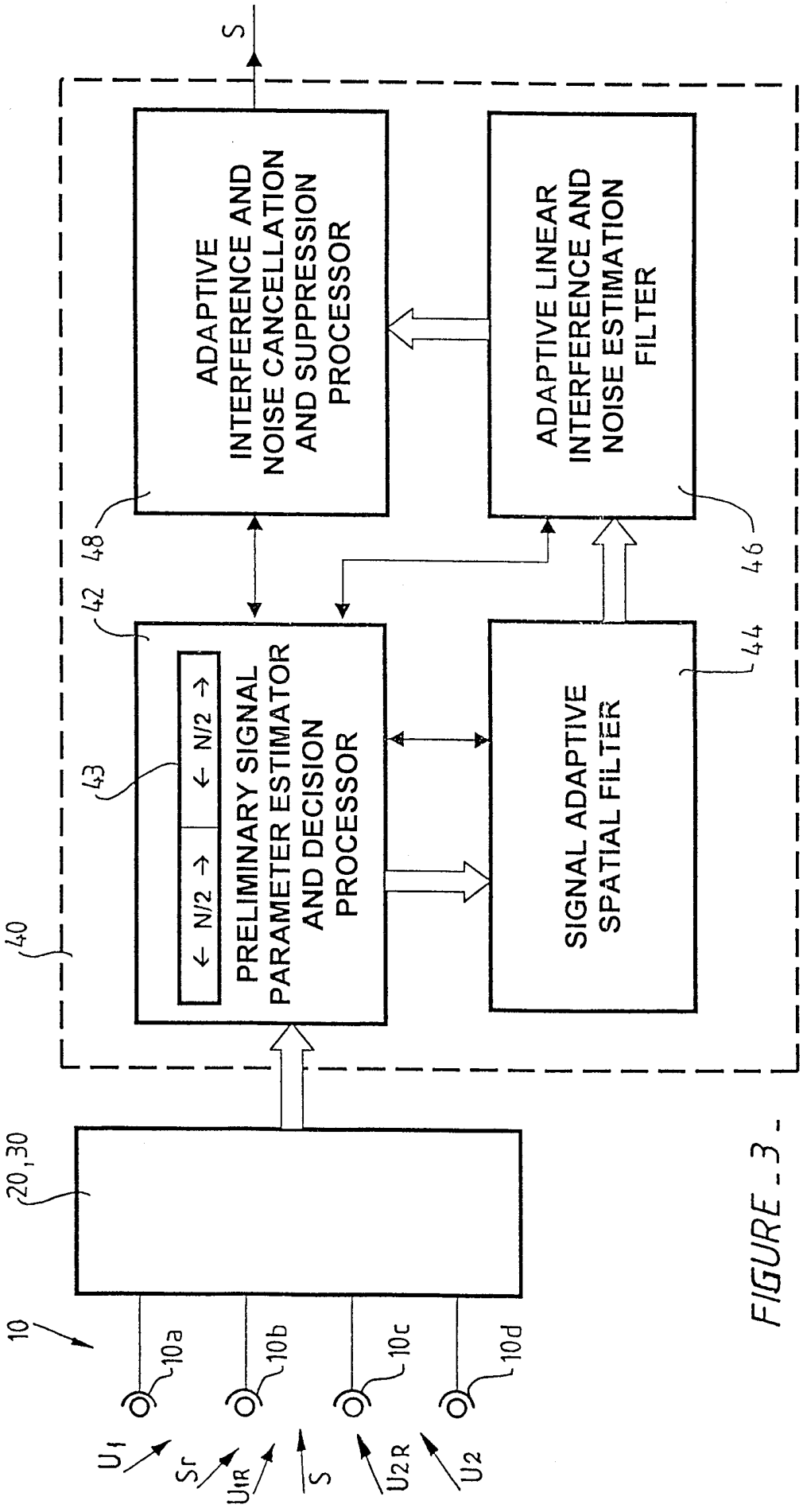

S
ADAPTIVE INTERFERENCE AND NOISE CANCELLATION AND SUPPRESSION PROCESSOR
ADAPTIVE LINEAR INTERFERENCE AND NOISE ESTIMATION FILTER
48
46
42
43
← N/2 →
← N/2 →
PRELIMINARY SIGNAL PARAMETER ESTIMATOR AND DECISION PROCESSOR
SIGNAL ADAPTIVE SPATIAL FILTER
44
40
20,30
10
10a
10b
10c
10d
U1
Sr
U1R
S
U2R
U2

FIGURE - 3 -

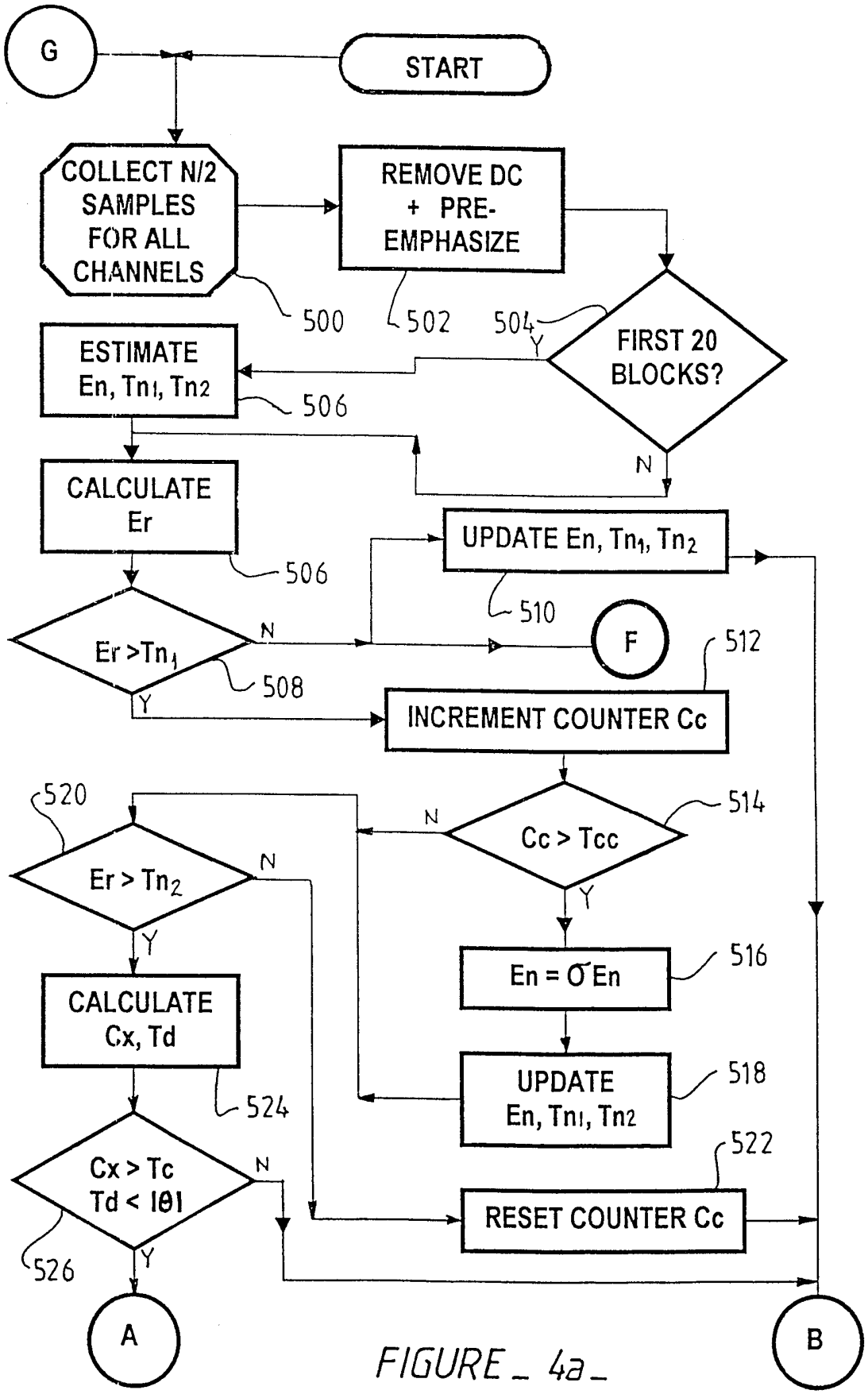

G
START
COLLECT N/2 SAMPLES FOR ALL CHANNELS
500
REMOVE DC + PRE-EMPHASIZE
502
504
FIRST 20 BLOCKS?
Y
N
ESTIMATE En, Tn1, Tn2
506
CALCULATE Er
506
UPDATE En, Tn1, Tn2
510
Er >Tn1
508
N
Y
F
512
INCREMENT COUNTER Cc
520
Er > Tn2
N
Y
514
Cc > Tcc
N
Y
516
En = σ En
CALCULATE Cx, Td
524
UPDATE En, Tn1, Tn2
518
522
Cx > Tc
Td < |θ|
N
Y
526
RESET COUNTER Cc
A
B

FIGURE_ 4a_

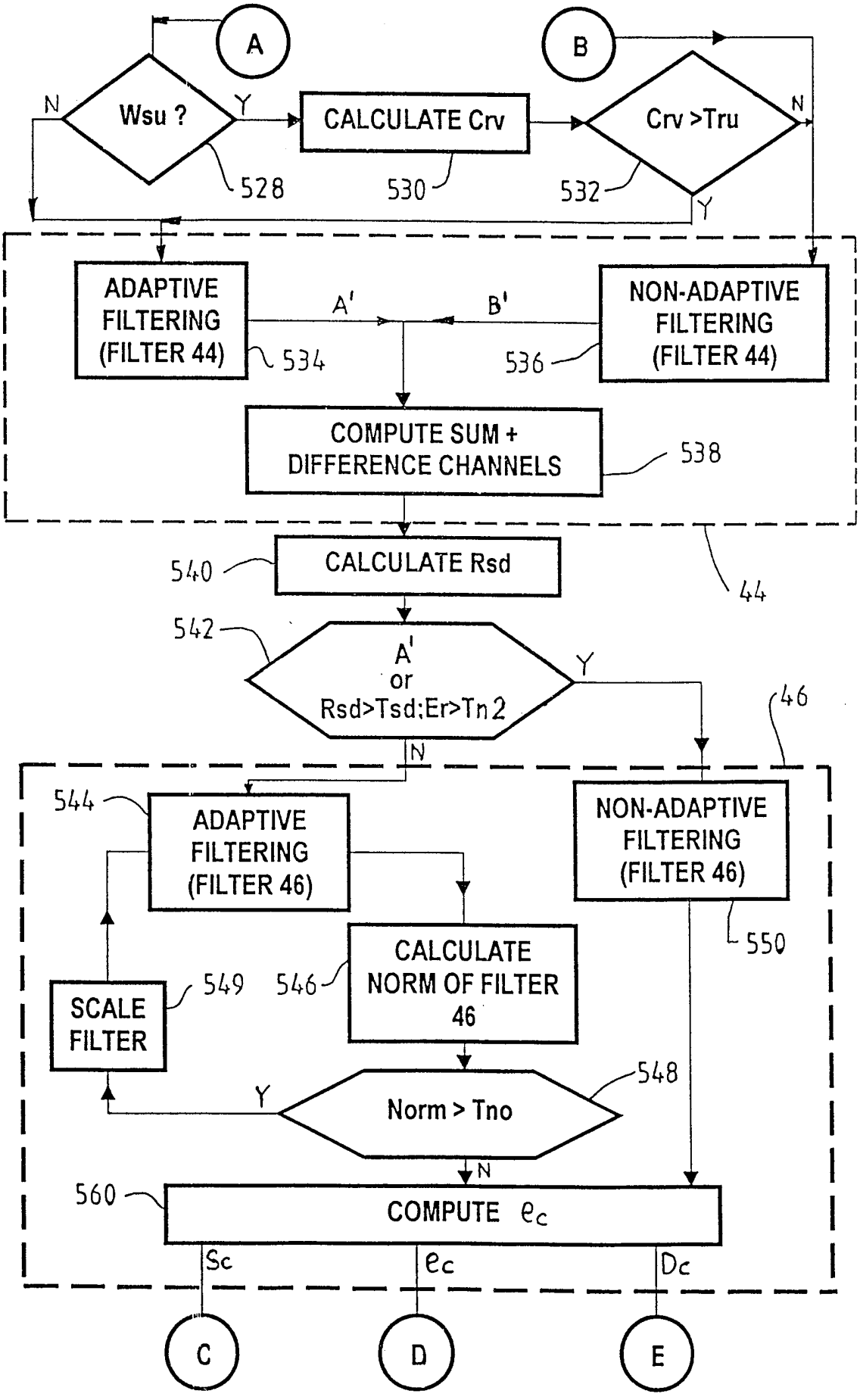

A
B
Wsu ?
N
Y
CALCULATE Crv
Crv >Tru
N
Y
528
530
532
ADAPTIVE FILTERING (FILTER 44)
534
A'
B'
NON-ADAPTIVE FILTERING (FILTER 44)
536
COMPUTE SUM + DIFFERENCE CHANNELS
538
44
540
CALCULATE Rsd
542
A' or Rsd>Tsd;Er>Tn2
Y
N
46
544
ADAPTIVE FILTERING (FILTER 46)
NON-ADAPTIVE FILTERING (FILTER 46)
550
SCALE FILTER
549
546
CALCULATE NORM OF FILTER 46
548
Y
Norm > Tno
N
560
COMPUTE ec
Sc
ec
Dc
C
D
E

FIGURE _ 4b _

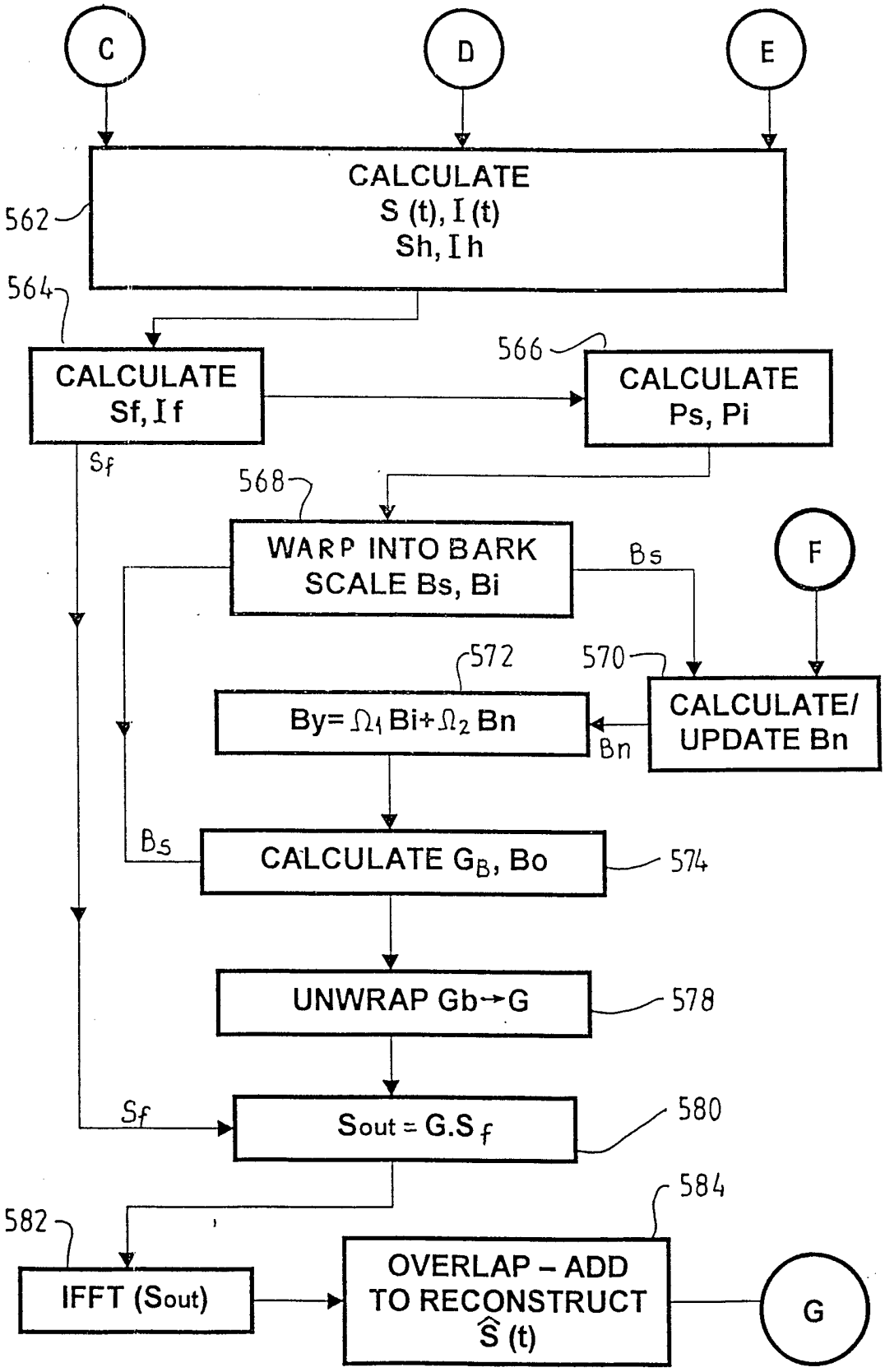

C
D
E
CALCULATE
S (t), I (t)
Sh, I h
562
564
CALCULATE
Sf, I f
566
CALCULATE
Ps, Pi
Sf
568
WARP INTO BARK
SCALE Bs, Bi
Bs
F
572
570
By= Ω1 Bi+Ω2 Bn
CALCULATE/
UPDATE Bn
Bn
Bs
CALCULATE GB, Bo
574
UNWRAP Gb→G
578
Sf
Sout = G.S f
580
584
582
IFFT (Sout)
OVERLAP – ADD
TO RECONSTRUCT
Ŝ (t)
G

FIGURE _ 4c _

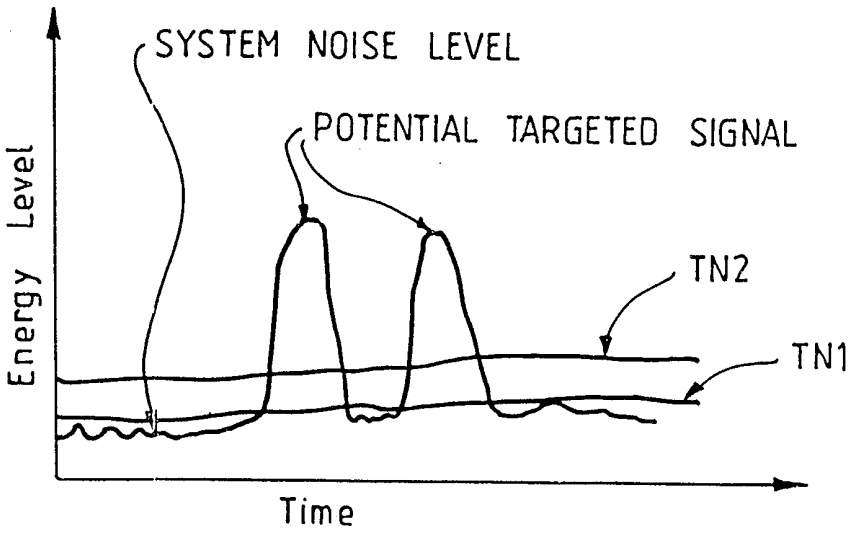


FIGURE_5_

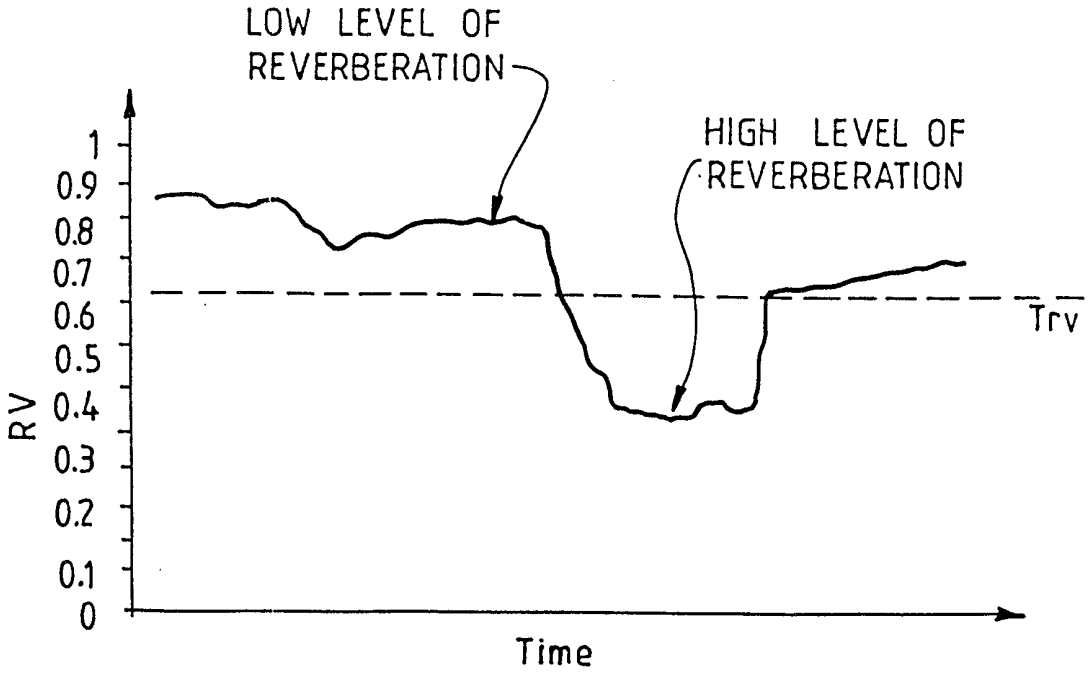


FIGURE_7_

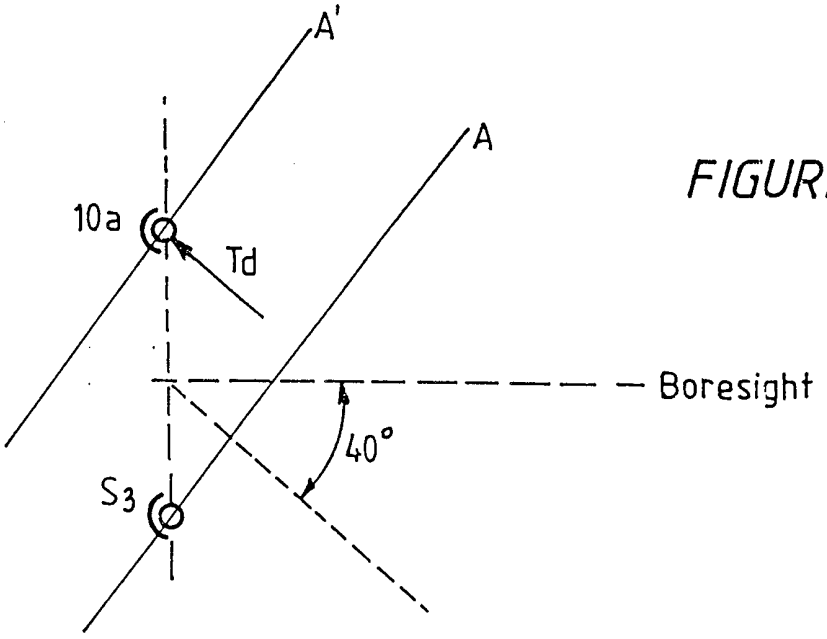


FIGURE _ 6A _

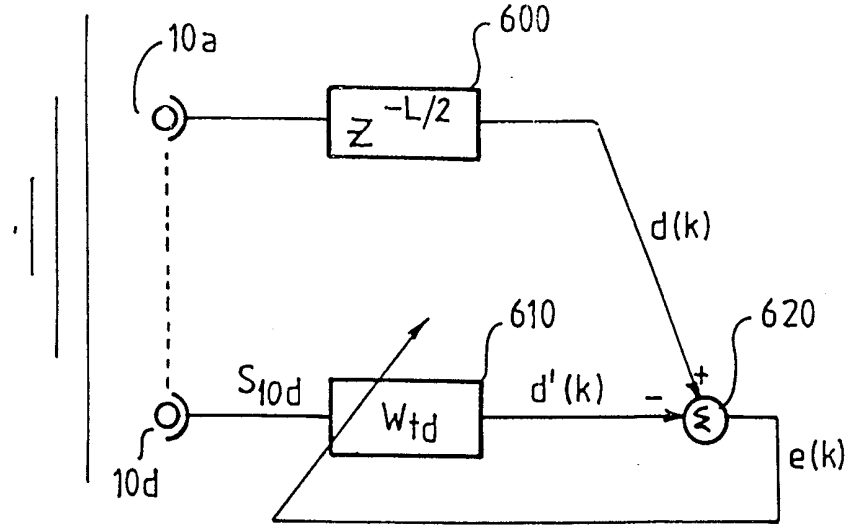


FIGURE _ 6B _

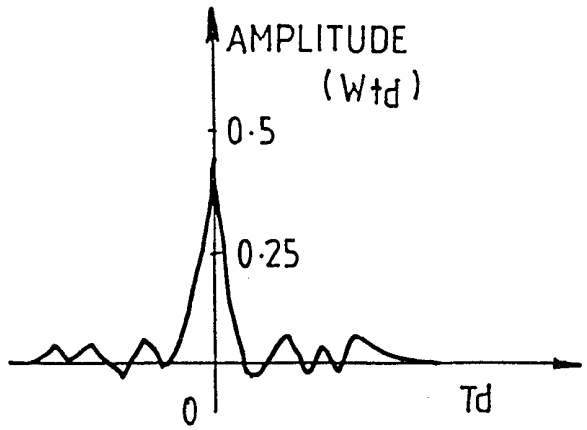


FIGURE _ 6C _

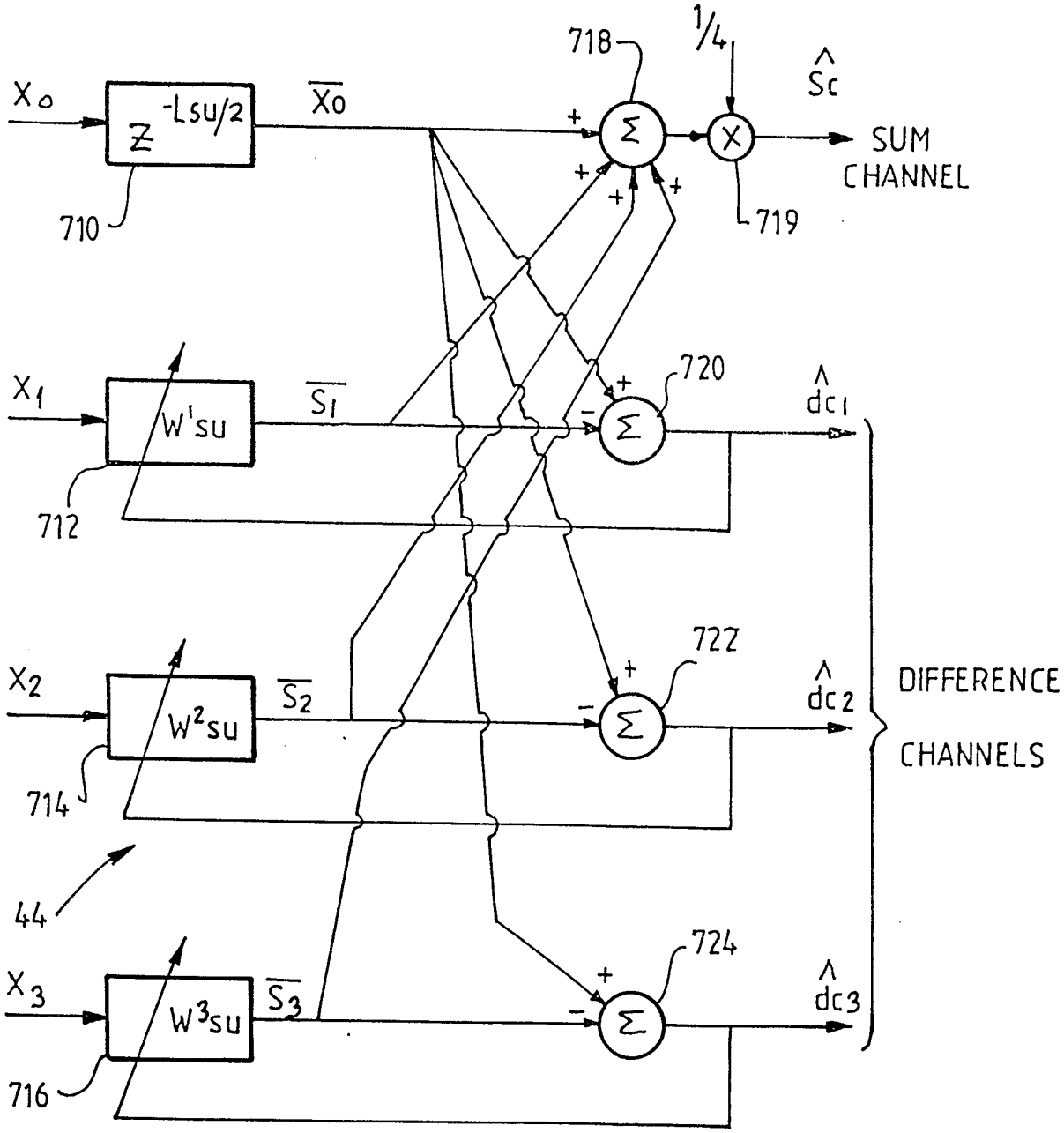

X0
-Lsu/2
z
710
X0
718
1/4
Sc
SUM
CHANNEL
719
X1
W'su
712
S1
720
dc1
X2
W2su
714
S2
722
dc2
DIFFERENCE
CHANNELS
44
X3
W3su
716
S3
724
dc3

FIGURE _8 _

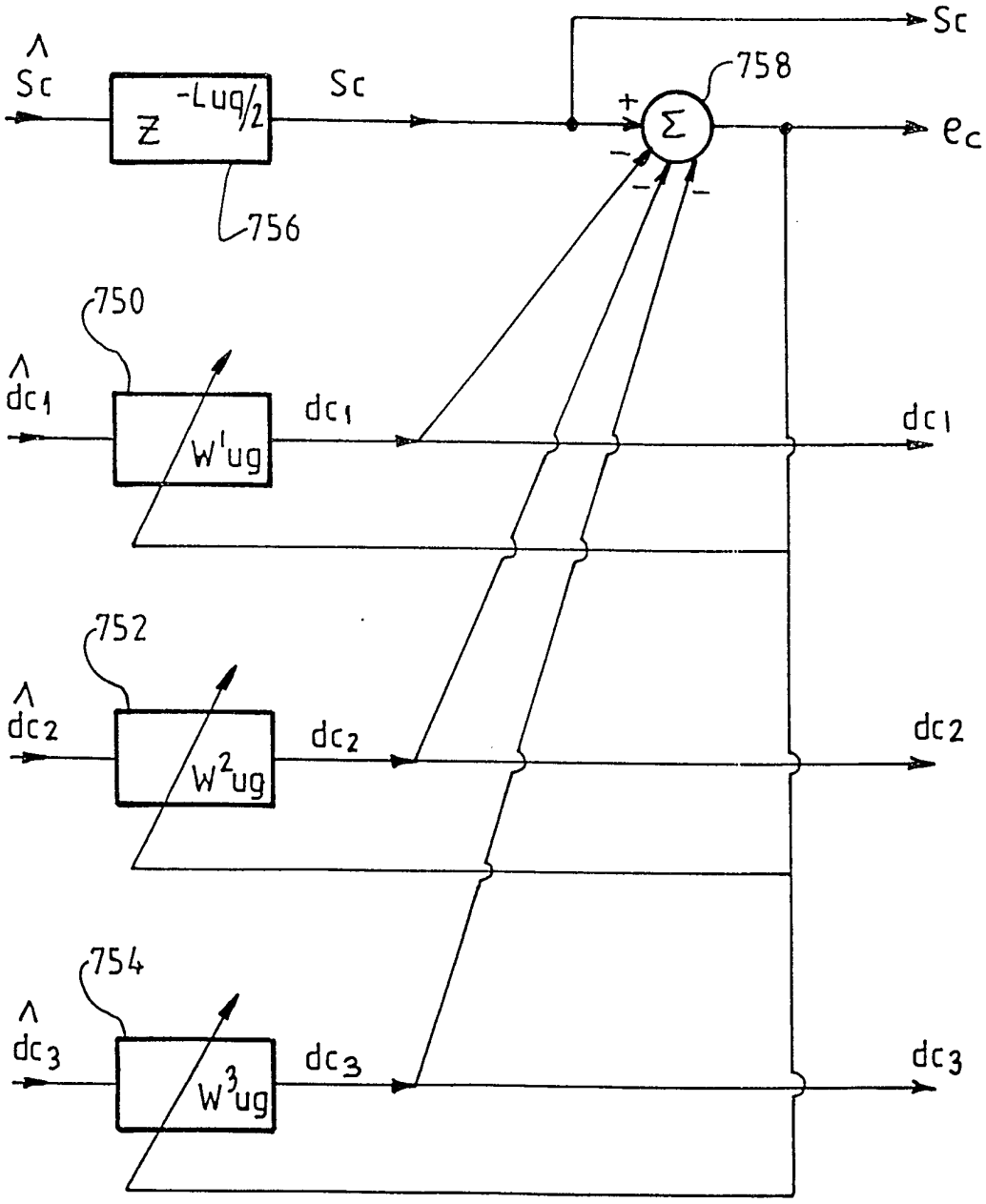

Sc
Sc
$z^{-Luq/2}$
Sc
758
Σ
+
−
−
−
Sc
e_c
756
750
$\hat{dc}_1$
W^1ug
dc_1
dc_1
752
$\hat{dc}_2$
W^2ug
dc_2
dc_2
754
$\hat{dc}_3$
W^3ug
dc_3
dc_3

FIGURE_9_

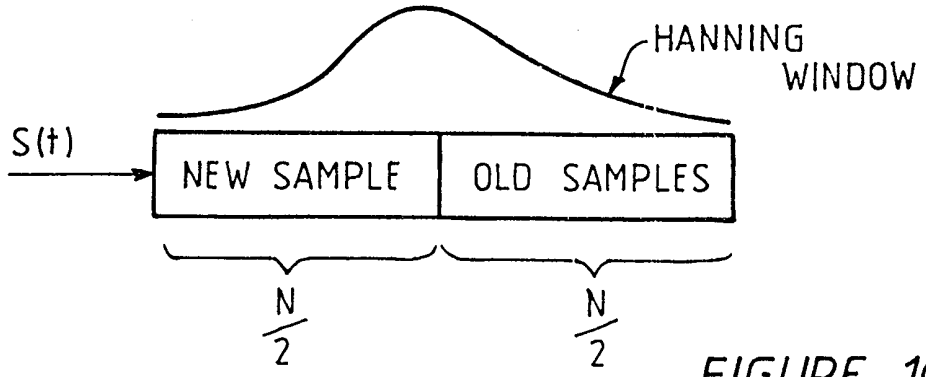


FIGURE _10_

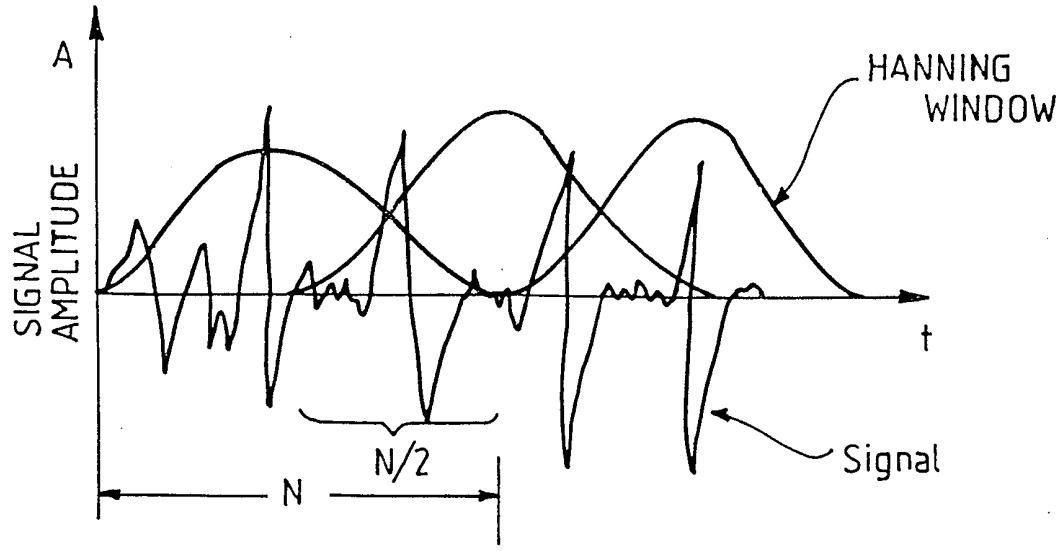


FIGURE _11_

Over-lap block processing of a signal using Hanning Window. The over-lap in this case is 50%.

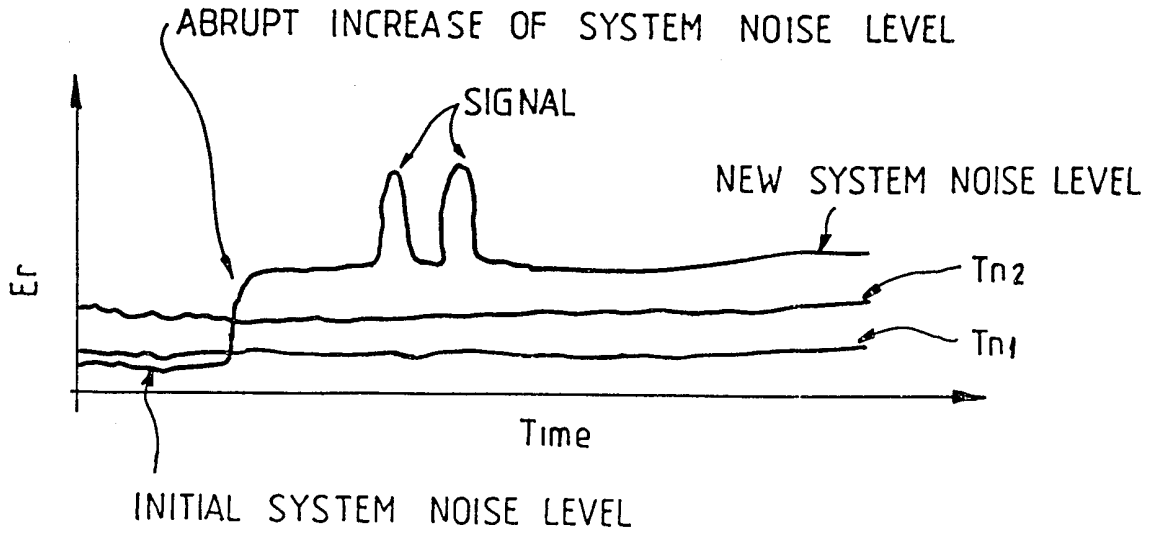


FIGURE _12_

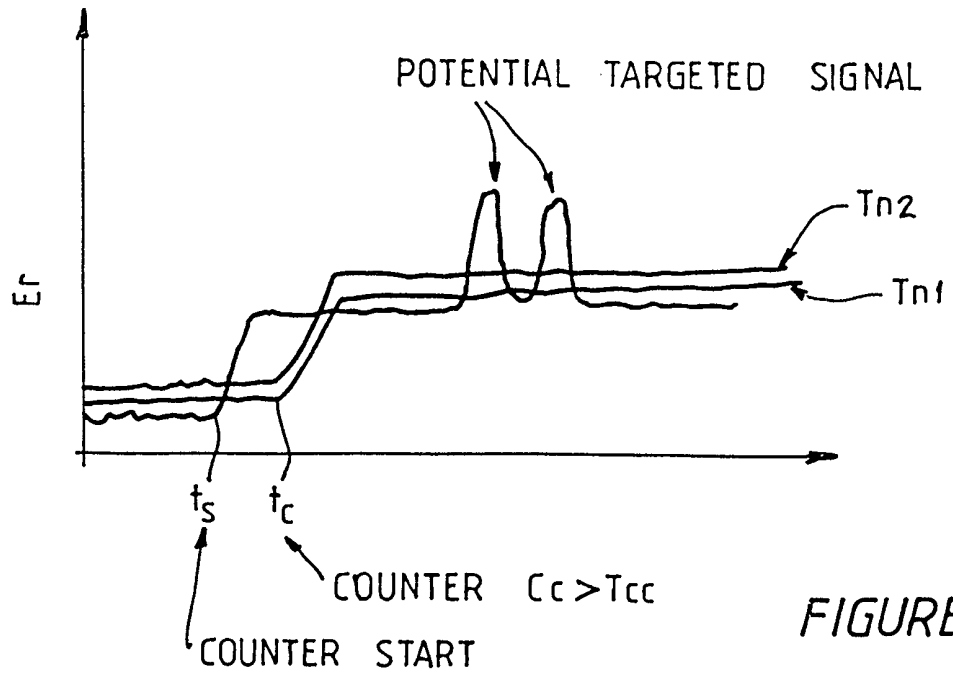


FIGURE _13_

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SG 99/00119

A. CLASSIFICATION OF SUBJECT MATTER

IPC[7]: H 03 M 13/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC[7]: H 03 M 13/00; G 06 G 7/00, 7/12

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

WPT

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 4931977 A (M. KLEMES) 05 June 1990 (05.06.90), totality.	1-6

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

* Special categories of cited documents:
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„L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
„O" document referring to an oral disclosure, use, exhibition or other means
„P" document published prior to the international filing date but later than the priority date claimed
„T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
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„Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
„&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 February 2000 (10.02.00)	31 March 2000 (31.03.00)
Name and mailing adress of the ISA/AT Austrian Patent Office Kohlmarkt 8-10; A-1014 Vienna Facsimile No. 1/53424/200	Authorized officer Zugarek Telephone No. 1/53424/383

Form PCT/ISA/210 (second sheet) (July 1998)

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No.
PCT/SG 99/00119

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US A 4931977	05-06-1990	CA A1 1272249	31-07-1990