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☐ TERMINAL DISCLAIMER	DRAWINGS			CLAIMS ALLOWED	
	Sheets Drwg.	Figs. Drwg.	Print Fig.	Total Claims	Print Claim for O.G.
	7	7	1	37	1

☐ a) The term of this patent subsequent to ____________ (date) has been disclaimed.

JACQUES SAINT-SURIN 12/3/99
(Assistant Examiner) (Date)

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☐ b) The term of this patent shall not extend beyond the expiration date of U.S Patent. No. ____________.

FORESTER W. ISEN
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2700
(Primary Examiner) (Date)

ISSUE FEE

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☐ c) The terminal ____months of this patent have been disclaimed.

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PATENT
670025-7007

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

APPLICATION FOR LETTERS PATENT

Applicant: Joseph **MARASH** and Baruch **BERDUGO**

Title: **INTERFERENCE CANCELING METHOD AND APPARATUS**

No. of Pages (Spec): 20

No. of Claims: 37 (pp 21-28)

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Sheets of Drawings: 7 (Figs. 1-7)

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Sir:

With reference to the filing in the United States Patent and Trademark Office of an application for patent in the name of:
JOSEPH MARASH and BARUCH BERDUGO

entitled: **INTERFERENCE CANCELING ~~METHOD AND APPARATUS~~**

The following are enclosed:

- X Specification (20 pages) and One Page of Abstract (p. 29)
- X 37 Claims (including 3 independent claims; pp. 21-28)
- X 7 Sheets of Drawings (Figs. 1-7)
- X Unsigned Declaration and Power of Attorney (2 pages)
- X The filing fee will be paid later, in response to a Notice to File Missing Parts. Kindly accord the application a September 18, 1998 filing date and address all communications to the undersigned at the address above.

Respectfully submitted,
Attorney for Applicant

By: Thomas J. Kowalski
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157035

PATENT
670025-7007

TITLE OF THE INVENTION

INTERFERENCE CANCELING METHOD AND APPARATUS

RELATED APPLICATIONS

Reference is made to co-pending U.S. applications Serial Nos. 08/672,899 (allowed), 09/130,923, 08/840,159, 09/059,503 and 09/055,709, each of which is hereby incorporated herein by reference; and each and every document cited in those applications, as well as each and every document cited herein, is hereby incorporated herein by reference.

FIELD OF THE INVENTION

The present invention relates to an interference canceling method and apparatus and, for instance, to an echo canceling method and apparatus which provides echo-canceling in full duplex communication, especially teleconferencing communications.

BACKGROUND OF THE INVENTION

Tele-conferencing plays an extremely important role in communications today. The teleconference, particularly the telephone conference call, has become routine in business, in part because teleconferencing provides a convenient and inexpensive forum by which distant business interests communicate. Internet conferencing, which provides a personal forum by which the speakers can see one another, is enormously popular on the home front, in part because it brings together distant family and friends without the need for expensive travel.

In a teleconferencing system, the sounds present in a room, hereinafter referred to as the "near-end room" such as those of a near-end speaker are received by a microphone, transmitted to a "far end system" and broadcast by a far-end loudspeaker. Similarly, the far-end speaker is received by the far-end microphones and transmitted to the near-end system, and broadcast by the near-end loudspeaker. The near-end microphone receives the broadcasted sounds along with their reverberations and transmits them back to the far-end, together with the desired signals generated by, for example, speakers at the near-end, thereby resulting in a disturbing echo heard by the speaker at the far-end. The far-end speaker will hear himself after the sound has traveled to the near-end system and back, thereby resulting in a delayed echo which will annoy and confuse the far-end speaker. The problem is compounded in video and internet conferencing systems where the delay is more extremely pronounced.

The simplest way to overcome the problem of echo is by blocking the near-end microphone while the far-end signal is broadcast by the near-end loudspeaker. Sometimes referred to as "ducking", the technique of blocking the microphone is effectively a half-duplex communication. Problematically, if the microphone is blocked for a prolonged period to avoid transmission of the reverberations, the half-duplex communication becomes a significant drawback because the far-end speaker will lose too much of the near-end speaker. In the video or Internet conferencing system, where the delay created by the communication lines is extreme, ducking becomes quite annoying.

A more complex method to avoid echo is to employ an echo canceling system which measures the signals send from the far-end and broadcast at the near-end loudspeaker, estimates the resulting signal present at the near-end microphone (including the reverberations) and subtracts those signals representing the echo from the near-end microphone signals. The echo-free signals are then transmitted back to the far-end system.

In order to reduce the echo from the near-end microphone signal, it is required to obtain the transfer function that expresses the relationship between the near-end loudspeaker signal and the reverberations as they actually appear at the near-end microphone. This transfer function depends on the relative position of the near-end loudspeaker to the near-end microphone, the room structure, position of the system and even the presence of people in the room. Since it is impossible to predict these parameters *a priori*, it is preferred that the echo-canceling system updates the transfer function continuously in real time.

The adaptation process by which the echo-canceling system is updated in real time may be an LMS (least means square) adaptive filter (Widrow, et al., Proc. IEEE, vol. 63, pp. 1692-1716, Proc. IEEE, vol. 55, No. 12, Dec. 1967) with the far-end signal used as the reference signal. The LMS filter estimates the interference elements (echoes) present in the interfered channel by multiplying the reference channel by a filter and subtracting the estimated elements from the interfered signal. The resulting output is used for updating the filter coefficients. The adaptation process will converge when the resulting output energy is at a minimum, leaving an echo-free signal.

Important to the adaptation process is the selection of the size of the adaptation step of the filter coefficients. In the standard LMS algorithm the step size is controlled by a predetermined adaptation coefficient, the level of the reference channel and the output level. In other words, the adaptation process will have bigger steps for strong signals and smaller steps for weaker signals.

A better behaved system is one in which its adaptation steps are independent of the reference channel levels. This is accomplished by normalizing the adaptation coefficient by the reference channel energy, this method is called the Normalized Least Mean Square (NLMS) as, for example, described in see for example "A Family of Normalized LMS Algorithms", Scott C. Douglas, IEEE Signal Processing Letters, Vol. 1, No. 3, March 1994. It should be noted that the energy estimator, if not designed properly, may fail to track when large and fast changes in the level of the reference channel occur. Thus, the normalized coefficient may be too big during the transition period, and the filter coefficient may diverge.

Another problem is that the adaptive process feeds the output back to determine the new filter coefficients. When the interfering elements in the signal are less pronounced than the non-interfering signal, there is not much to reduce and the filter may diverge or converge to a wrong value which results in signal distortions.

When properly converged, the adaptive filter actually estimates the transfer function between the far-end loudspeaker signal and the echo elements in the main channel. However, changes in the room will effect a change in the transfer function and the adaptive process will adapt itself to the new conditions. Sudden or quick changes, in particular, will take the adaptive filter time to adjust for and an echo will be present until the filter adapts itself to the new conditions.

In order to improve the audio quality, sometimes a number of microphones are used instead of a single one. This system either selects a different microphone each time someone is speaking in the room or creates a directional beam using a linear combination of microphones. By multiplexing the microphones or steering the directional audio beam, the relationship between the loudspeaker signal and the audio signal obtained by the microphones can be changed. Problematically, each time such a transition takes place, an echo will "leak" into the system until the new condition has been studied by the adaptive filter. To allow the use of a steerable directional beam and prevent the transient echo, one can either perform continuous echo canceling on each of the microphones separately or on each of the microphone combinations (the combinations of microphones could be infinite). However, the increase in the computation load required to perform numerous echo-canceling systems concurrently on each of the microphones or allowable beams is not realistic.

An efficient echo-canceling system is needed which will reduce the echo drastically. However, because of the large dynamic ranges required by the microphone to be able to pick up very low voices, the microphone will most likely pick up some of the residual echo as well. The residual echo is most disturbing when no other signal is present but less noticed when a full duplex discussion is taking place.

Another problem typical to multi-user conferencing systems is that the background noise from several systems is transmitted to all the participating systems and it is preferred that this noise be reduced to a minimum. The beam forming process reduces the background noise but not enough to account for the plurality of systems.

OBJECTS AND SUMMARY OF THE INVENTION

It is therefore an object of the invention to provide an interference canceling system.

It is another object of the invention to provide an interference canceling system to cancel interference while providing full duplex communication.

It is yet another object of the invention to provide an interference canceling system to cancel an echo present in a teleconference.

It is still another object of the present invention to provide an interference canceling system to cancel an echo present in video teleconferencing.

It is further an object of the invention to allow a steerable directional audio beam to function with the interference canceling system of the present invention.

It is yet a further object of the invention to overcome background noise in the conferencing system and reduce the residual echo to a minimum.

In accordance with the foregoing objectives, the present invention provides an interference canceling system, method and apparatus for canceling, from a target signal generated from a target source, an interference signal generated by an interference source. A main input inputs the target signal generated by the target source. A reference input inputs the interference signal generated by the interference source. A beam splitter beam-splits the target signal into a plurality of band-limited target signals and beam-splits the interference signal into band-limited interference signals. Preferably, the amount and frequency of band-limited target signals equals the amount and frequency of band-limited interference signals, whereby for each band-limited target signal there is a corresponding band-limited interference signal. An adaptive filter adaptively filters, each band-limited interference signal from each corresponding band-limited target signal.

When the target signal represents speech generated at a near end of a teleconference, the adaptive filter of the present invention cancels an echo present in the reference signal broadcast from a far end of the teleconference. It is preferred that the adaptive filter is an adaptive filter array with each adaptive filter in the array filtering a different frequency band. In the exemplary embodiment the adaptive filter estimates a transfer function of the reference signal broadcast from the far end.

The adaptive filter of the present invention may further comprise an inhibitor. The inhibitor permits the adaptive filter to adapt (change coefficients) when a signal-to-noise ratio of the reference signal exceeds a predetermined threshold over a signal-to-noise ratio of the main signal. Preferably, the inhibitor determines the predetermined threshold periodically.

The beam splitter of the exemplary embodiment of the present invention is a DFT filter bank using single side band modulation. Additionally, the present invention may comprise a beam selector for selecting at least one of a plurality of beams for adaptive filtering by the adaptive filter representing a direction from which the main signal is received. In this case, the adaptive filter updates coefficients representing the transform function and comprehensively stores the coefficients for each beam selected by the beam selector. In the exemplary embodiment, the beam selector selects the plurality of the beams for simultaneous adaptive filtering by the adaptive filter. Further, the beam selector may select a beam having a fixed direction and a beam which rotates in direction.

The present invention may further comprise a noise gate for gating the main signal adaptively filtered by the adaptive filter by opening the noise gate when a signal-to-noise ratio at the near end is above a predetermined threshold and closing the noise gate when the signal-to-noise ratio at the near end is below the predetermined threshold. In this case, the noise gate determines the predetermined threshold by selecting a low threshold when a signal-to-noise ratio of the reference signal of the far end is low, updating the predetermined threshold upwards when the signal-to-noise ratio of the reference signal of the far end goes up and gradually reducing the predetermined threshold when the signal-to-noise ratio of the reference signal of the far end goes down.

BRIEF DESCRIPTION OF THE DRAWINGS

A more complete appreciation of the present invention and many of its attendant advantages will be readily obtained by reference to the following detailed description considered in connection with the accompanying drawings, in which:

Fig. 1 illustrates the interference canceling system of the present invention.

Fig. 2 illustrates the beamforming unit of the present invention.

Fig. 3 illustrates the decimation unit of the present invention.

Fig. 4 illustrates the beam splitting unit of the present invention.

Fig. 5 illustrates the adaptive filter of the present invention.

Fig. 6 illustrates the recombining unit of the present invention.

Fig. 7 illustrates the noise gate of the present invention.

DETAILED DESCRIPTION

Figure 1 illustrates the exemplary echo canceling system of the present invention. An array of microphone elements 102 receive and convert acoustic sound in a room into an analog signal which is amplified by the signal conditioning block 104 and converted into digital form by the A/D converter 106. While Figure 1 appears to depict the microphone elements 102 as an array, it will be appreciated by those skilled in the art that other configurations are readily applicable to the present invention. The microphone elements, for example, may be arranged in a circular array, a linear, or any other type of array. The A/D converter 106 may be an array of Delta Sigma converters set to, for example, a sampling frequency of 64KHz per channel but, of course, may be substituted with other types of converters and sampling frequencies which are suitable as those skilled in the art will readily understand.

The sampled signals of each microphone are stored in a tap delay line (not shown) and multiplied by a steering matrix in the beam forming unit 108 to form a number of directional beams. As an example, 6 beams are formed which are aimed in directions evenly spread over 360 degrees (60 degrees apart). Of course, the present invention is not limited to any specific number of beams as one skilled in the art will readily understand. The beam signals are then low pass filtered to, for example, 8KHz and decimated by decimating unit 110 to reduce the sampling rate and hence the computational load on the system. In this manner, the sampling rate is reduced to 16 KHz for each channel. It shall be appreciated that the decimation process may be performed prior to the beamforming process to further reduce the processing burden.

The system receives an indication as to the direction of the speaker either through a direction finding system or through a manual steering process. In the exemplary embodiment, the beam select logic unit 112 selects the beam with the closest direction to that actual and performs echo cancellation processing on the selected beam.

A particular aspect of the present invention is that the selected beam is split into a number of frequency bands, preferably 16 evenly spaced bands, by the beam splitter 114 such that echo cancellation processing is performed on each frequency band separately. Without this arrangement, an echo which typically lasts for more than 100 msec would require an adaptive filter, assuming that the filter samples the 100 msec of signal at a rate of 16KHz, to have 1600 coefficients. Such a long adaptive filter is not likely to converge in the time that the echo is present. Moreover, an adaptive filter of 1600 coefficients presents an enormous processing burden which is unrealistic to handle. By splitting the bands into, for example, 16 channels the present invention reduces the sampling rate for each adaptive filter to, in this case, 2 KHz per channel. It will be appreciated that, not only is this system much more manageable, the adaptive filters can be optimized for each frequency separately by, for example, selecting longer filters for lower frequencies where the echo is typically located and shorter filters for higher frequencies where the echo is less. In this case, the filter lengths range, for example, from 16 to 128 coefficients. With this arrangement, the adaptive filters can converge much more easily with these lengths, the treatment of each band is independent from the others thereby preventing the problem of a broadband filter concentrating on a band limited interference while ignoring less pronounced ones and the processing burden is reduced.

Meanwhile, the far end signal (referred to as the reference channel) is conditioned, sampled, decimated and split in the manner discussed above by respective signal conditioning block 122, A/D converters 124, decimating unit 126 and splitter 128. Each band of the selected beam is processed for echo reduction using echo canceling units $116_{1\text{-}m}$. While Normalized LMS filters are preferred, those skilled in the art will readily understand that other type of adaptive filters are applicable to the present invention. The resulting echo-free signals of the different frequency bands are recombined into one broadband output by a recombine output unit 118.

The output of the recombined process is fed into a noise gate processor 120. The purpose of the noise gate is to prevent steady background noise in the room (such as fan noise) from being transmitted to the far end system and eliminate residual echoes. The system of the present invention measures the level of the steady noise and blocks up the signals that are below a certain threshold above this noise level. When residual echoes are present they may penetrate the process and be transmitted to the far end system. In order to prevent that, the blocking threshold is actively adjusted to the level of the signal present at the reference channel (far end).

When a high level energy is detected at the far end signal, the threshold will be boosted up and gradually reduced when this signal disappears. This will prevent residual echoes from being transmitted while leaving only speech signals from the near end.

Figure 2 illustrates the beamforming unit 200 (Figure 1, 108) of the present invention. Signals originated at a certain relative direction to the microphone array arrive at different phases to each microphone. Summing them up will create a reduced signal depending on the phase shift between the microphones. The reduction goes down to zero when the phases of the microphones are the same, thus creating a preferred direction while reducing all other directions. In the beamforming process, the microphone signals are phase shifted to create a zero phase difference for signals originated at a predetermined direction. The phase shift is achieved by multiplying the microphone signal stored in the tap delay lines $202_{1\text{-}n}$ by a FIR filter coefficient or steering vector output from steering vector units $204_{1\text{-}n}$.

In one embodiment, a different weight is applied for each microphone to create a shading effect and reduce the side lobe level. The weighting factors are implemented as part of the FIR filter coefficients. The filters for each direction and each microphone are pre-designed and stored as a steering vector matrix $204_{1\text{-}n}$. The microphone signals are stored in a tapped delay line 2021-n with the length of the FIR filter. For each direction, each microphone delay line is multiplied by multipliers $206_{1\text{-}n}$ by its FIR and summed with the other microphones after they have been multiplied. The process repeats for each direction resulting in a beam output for each direction.

Figure 3 illustrates the decimation unit 300 (Figure 1, 110, 126) of the present invention. Decimation, which is intended to reduce the sampling frequency, can be done only once the high frequency elements are removed to maintain the Nyquist criteria. For example, if the sampling frequency is to be reduced to 16 KHz, it is necessary to make sure that the signal does not contain elements above 8KHz because sampling will result in aliasing. In order to remove the troublesome high frequencies, the signals are first filtered by a low pass filter that cuts off the higher frequencies. In more detail, the beam samples are stored in a tapped delay line 302 and multiplied via a multiplier 304 by a low pass filter coefficient produced by the low pass filter 306.

Figure 4 illustrates the beam splitting unit 400 (Figure 1, 114, 128) of the present invention. Although various beam splitting techniques may be employed, it is preferred that the generalized DFT filter bank using single side band modulation be employed as described, for example, in "<u>Multirate Digital Signal Processing</u>", Ronald E. Crochiere, Prentice Hall Signal Processing Series or "<u>Multirate Digitals Filters, Filter Banks, Polyphase Networks, and Applications A Tutorial</u>", P. P. Vaidyanathan, Proceedings of the IEEE, Vol. 78, No. 1, January 1990. The goal of the beam splitter is to split the input signal into a plurality of limited frequency bands, preferably 16 evenly spaced bands. In essence, the beam splitting processes, for example, 8 input points at a time resulting in 16 output points each representing 1 time domain sample per frequency band. Of course, other quantities of samples may be processed depending upon the processing power of the system as will be appreciated by those skilled in the art.

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In more detail, the 8 input points 402 are stored in a 128 tap delay line 404 representing a 128 points input vector which is multiplied via a multiplier 406 by the coefficients a 128 points complex coefficients pre-designed filter 408. The 128 complex points result vector is folded by storing the multiplication result in the 128 points buffer 410 and summing the first 16 points with the second 16 points and so on using a summer 412. The folded result, which is referred to as an aliasing sequence 414, is processed through a 16 points FFT 416. The output of the FFT is multiplied via a multiplier 418 by the modulation coefficients of a 16 points modulation coefficients cyclic buffer 420. The cyclic buffer which contains, for example, 8 groups of 16 coefficients, selects a new group each cycle. The real portion of the multiplication result is stored in the real buffer 422 as the requested 16-point output 424.

Figure 5 illustrates the adaptive filter 500 (Figure 1, $116_{1\text{-}n}$) of the present invention. The reference channel that contains the far end signal is stored in a tap delay line 502 and multiplied via a multiplier 504 by a filter 506 to obtain the estimated echo elements present in the beam signal. The estimated interference signal is then subtracted via subtractor 508 from the beam signal to obtain an echo free signal.

The filter 506 is adjusted by the NLMS (Normalized Least Mean Square) processor 510 to estimate the transfer function of the loudspeaker to the beamforming process. In other words, the filter 506 simulates the transform that the far end signal goes through when transmitted by the loudspeaker into the air, bouncing back from the walls, received by the microphones and applied to the beamforming process of the present invention. In order to determine the precise filter coefficients, the system tries to obtain minimum energy at the output by modifying the filter coefficients (W) according to the following formula:

(1) W(n,t+1)=W(n,t)+X(n)*E*A

Wherein, n is the nth coefficient of W, t is time, E is the error signal output and A is a normalized factor that determines the size of the adaptation process. The normalization is obtained by dividing a fixed value (adaptation factor) by P, the reference channel energy. The normalization is intended to prevent fast steps when the signal is strong (i.e., X and E are large) and small steps when weak (i.e., X and E are small) which provides smooth performance over all ranges of signal levels.

When a fast attack in the reference signal appears, such as when an abrupt sound, e.g., speech, noise, is generated at the far end, the energy estimation process may be too slow in reaction resulting in large steps of adaptation and divergence of the filter. To prevent this, the new X*X is compared to the energy estimation calculated by power estimator 512 and if the ratio exceeds a certain threshold (meaning a fast increase in the signal level) the value of X*X replaces the energy estimation.

If the content of the near end signal is much stronger than the content of the far end signal the filter may diverge or converge to wrong values and start distorting the desired signal. It is preferred that the adaptation process will occur when relevant echo signals are present in the beam signal. To determine this, the system calculates the SNR of the far end signal and the SNR of the near end signal using the SNR estimation units 514, 516. If speech is present in the near end signal, the SNR of the beam will be stronger than that of the reference channel. Thus, when the SNR of the reference channel raises up above a predetermined threshold over the near end SNR, the inhibit update logic block 518 immediately allows the LMS coefficient to be updated. Conversely, the inhibit update logic block will allow, for example, 100 msec of adaptation and then inhibit the adaptation when the ratio drops below the threshold. At this point, the coefficients of the adaptive filter of the present invention "freeze" and the filtering will use the latest value of the coefficients. Later, when adaptation is no longer inhibited, the filters are updated from the values at which they were "frozen".

The exemplary embodiment determines the predetermined threshold for the inhibit update logic block 518 in discrete periods. The timing of these discrete periods is determined in part by the hysteresis that differentiates between the reaction time of the attack to that of the decay of the SNR ratios which are obtained through the reaction time of the energy calculation. More specifically, the SNR is computed by dividing two values, the noise level and the signal level. The energy of each block of both the reference and the beam are calculated using a exponential running average of the absolute value of the data. In the exemplary embodiment, the block size is defined as 20 msec of data which is considered to contain the signal level. The present invention searches the lowest energy of a block in the current period, for example, previous 2 sec. Every 2 Sec the system resets and starts recording the value of the block energy and replacing the value when a lower value is calculated. When the current 2 sec time period has elapsed, the calculated noise level is copied and recorded as the current noise level while the system resets the calculation process for the next noise level which will be used for the next 2 sec period.

It will be appreciated from the foregoing description that the present invention stores the values of the coefficients for each frequency band and for each beam direction separately. Once the beam selector 112 selects a new beam, the appropriate values of the beam will be selected. In this way, the system will keep a record of the transfer function between each beam and the beamformer, and the adaptation to the echoes in the new direction will be updated. This process allows the use of directional beamforming while providing a fast adaptation time which obviates the need to perform while the process for either all of the microphones or all the beams.

In another embodiment, which updates the adaptation coefficients even more frequently, the present invention as described is applied on a plurality of beams at a time. For purposes of example, the present invention selects two beams, one which is selectively directed and the other which is actively rotated periodically, for example, every 40 msec. In the alternative, predetermined beams may be selected more often than others. With this arrangement, a different beam will be selected for each block in addition to the main beam and will be processed according to the afore-mentioned adaptation process of the present invention. While this method increases computation load, it ensures that the coefficients in all directions, particularly those predetermined, are updated more frequently.

Figure 6 illustrates the recombining unit 600 (Figure 1, 118) of the present invention which is symmetrical, i.e., opposite, to the band splitting technique described above. The goal here is to recombine the 16 limited frequency bands of the echo free signal into one broad band output. The process goes through an IFFT process but both the input and output are time domain signals. The recombining unit of the exemplary embodiment processes 16 input points 602 each representing 1 time domain sample per frequency band resulting in 8 output points 604 of the broadband signal. Of course, those skilled in the art will readily understand that other quantities of sampling input points are applicable to the present invention.

In more detail, the new 16 input points 602 are multiplied by a multiplier 606 with a 16 points demodulation filter coefficient which is stored in a demodulation coefficients cyclic buffer 608 containing, for example, 8 groups of 16 coefficients wherein a new group is selected each cycle. The result is processed through a 16 points IFFT 610, or any equivalent transform, and the result of this Inverse Fast Fourier Transform is extracted to 128 complex points by duplicating the 16 points data 8 times. The 128 points result vector which is stored in a buffer 612 is multiplied via the multiplier 614 by a 128 point complex coefficient generated by a predesigned complex filter 616 and stored in real buffer 618. The real portion of the result is summed by summer 620 into a 128 points cyclic history buffer 622 in which the oldest 8 points are taken as the result 604 and replaced with zeros in the buffer 622 for the next iteration of the recombination process.

Figure 7 illustrates the noise gate system 700 (Figure 1, 120) of the present invention. The far end signal-to-noise ratio SNR is calculated by SNR estimation unit 702 which estimates the signal energy of the current block (40 msec in the exemplary embodiment) and divides the signal energy by the lowest estimated block energy in the current period (2 sec in the exemplary embodiment). The threshold is selected by the threshold select depending on the far end signal-to-noise ratio SNR. When the far end SNR is low, a low threshold is selected. Once the SNR of the far end goes up, the threshold is updated immediately upwards by the threshold selection unit 704. When the far end SNR goes down, the threshold is gradually reduced to a minimum with a decay time in the exemplary embodiment around 100 msec.

The near end signal-to-noise ratio SNR is measured by the SNR estimation unit 706 in the same manner. Then, the near end SNR signal is compared by the comparator 708 to the selected threshold. According to the logic provided by the logic circuit 710, if the difference is positive, meaning that the near end signal is present, the gate 712 is open, preferably immediately or quickly (e.g., so as to not miss a syllable, for instance in less than about 10 msec or less such as instantly or nearly instantly). On the other hand, if the result of the comparison is negative, meaning that the near end signal is not above the allowed threshold, the gate is closed and the level of sound is significantly reduced such that the reduced signal is transmitted to the far end system. The reduction of the sound or the closure of the gate is preferably gradual such as over about 100 msec or longer, e.g., over about 0.5 sec or 1.0 sec, so as to prevent a pumping sound or noise transmission when a user is speaking fast and to have the gate truly close when there is a real pause or silence.

It will be appreciated from the foregoing description that the present invention provides an echo-canceling system which overcomes the problem of background noise in the conferencing system, reduces the residual echo to a minimum, allows full duplex communication and provides a steerable directional audio beam.

Although preferred embodiments of the present invention and modifications thereof have been described in detail herein, it is to be understood that this invention is not limited to those precise embodiments and modifications, and that other modifications and variations may be effected by one skilled in the art without departing from the spirit and scope of the invention as defined by the appended claims.

WE CLAIM:

1. An interference canceling apparatus for canceling, from a target signal generated from a target source, an interference signal generated by an interference source, said apparatus comprising:

a main input for inputting said target signal;

a reference input for inputting said interference signal;

a beam splitter for beam-splitting said target signal into a plurality of band-limited target signals and beam-splitting said interference signal into band-limited interference signals, wherein the amount and frequency of band-limited target signals ~~equals~~ equal the amount and frequency of band-limited interference signals, whereby for each band-limited target signal there is a corresponding band-limited interference signal;

an adaptive filter for adaptively filtering, each band-limited interference signal from each corresponding band-limited target signal.

2. The apparatus according to claim 1, wherein said target signal represents speech generated at a near end of a teleconference, said reference signal represents said target signal broadcast from a far end of said teleconference and said interference signal represents an echo generated by said broadcast of said reference signal of said far end.

3. The apparatus according to claim 2, wherein said adaptive filter is an adaptive filter array with each adaptive filter in said array filtering a different frequency band.

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4. The apparatus according to claim 2, wherein said adaptive filter estimates a transfer function of said reference signal broadcast of said far end.

5. The apparatus according to claim 4, further comprising an inhibitor for permitting said adaptive filter to change coefficients when a signal-to-noise ratio of said reference signal exceeds a predetermined threshold over a signal-to-noise ratio of said main signal.

6. The apparatus according to claim 5, wherein said inhibitor determines said predetermined threshold periodically.

7. The apparatus according to claim 2, wherein said beam splitter is a DFT filter bank using single side band modulation.

8. The apparatus according to claim 2, further comprising a beam selector for selecting at least one of a plurality of beams for adaptive filtering by said adaptive filter representing a direction from which said main signal is received.

9. The apparatus according to claim 8, wherein said adaptive filter updates coefficients representing said transform function and comprehensively stores said coefficients for each beam selected by said beam selector.

10. The apparatus according to claim 8, wherein said beam selector selects said plurality of said beams for simultaneous adaptive filtering by said adaptive filter.

11. The apparatus according to claim 10, wherein said beam selector selects a beam having a fixed direction and a beam which rotates in direction.

12. The apparatus according to claim 2, further comprising a noise gate for gating said main signal adaptively filtered by said adaptive filter by opening said noise gate when a signal-to-noise ratio at the near end is above a predetermined threshold and gradually closing said noise gate when said signal-to-noise ratio at the near end is below the predetermined threshold; wherein said noise gate determines said predetermined threshold by selecting a low threshold when a signal-to-noise ratio of said reference signal of the far end is low, updating said predetermined threshold upwards when said signal-to-noise ratio of said reference signal of the far end goes up and gradually reducing said predetermined threshold when said signal-to-noise ratio of the reference signal at the far end goes down.

13. An interference canceling apparatus for canceling, from a target signal generated from a target source an interference signal generated by an interference source, said apparatus comprising:

main input means for inputting said target signal;

reference input means for inputting said interference signal;

beam splitter means for beam-splitting said target signal into a plurality of band-limited target signals and beam-splitting said interference signal into band-limited interference signals, wherein the amount and frequency of band-limited target signals ~~equals~~ equal the amount and frequency of band-limited interference signals, whereby for each band-limited target signal there is a corresponding band-limited interference signal; and

a

adaptive filter means for adaptively filtering, according to said plurality of frequency bands, each band-limited interference signal from each corresponding band-limited target signal.

14. The apparatus according to claim 13, wherein said target signal represents speech generated at a near end of a teleconference, said reference signal represents said target signal broadcast from a far end of said teleconference and said interference signal represents an echo generated by said broadcast of said reference signal of said far end.

15. The apparatus according to claim 14, wherein said adaptive filter means is an adaptive filter array with each adaptive filter in said array filtering a different frequency band.

16. The apparatus according to claim 14, wherein said adaptive filter means estimates a transfer function of said reference signal broadcast of said far end.

17. The apparatus according to claim 16, further comprising inhibitor means for permitting said adaptive filter to change coefficients means when a signal-to-noise ratio of said reference signal exceeds a predetermined threshold over a signal-to-noise ratio of said main signal.

18. The apparatus according to claim 17, wherein said inhibitor means determines said predetermined threshold periodically.

19. The apparatus according to claim 14, wherein said beam splitter means is a DFT filter bank using single side band modulation.

20. The apparatus according to claim 14, further comprising beam selector means for selecting at least one of a plurality of beams for adaptive filtering by said adaptive filter means representing a direction from which said main signal is received.

21. The apparatus according to claim 20, wherein said adaptive filter means updates coefficients representing said transform function and comprehensively stores said coefficients for each beam selected by said beam selector means.

22. The apparatus according to claim 20, wherein said beam selector means selects said plurality of said beams for simultaneous adaptive filtering by said adaptive filter means.

23. The apparatus according to claim 22, wherein said beam selector means selects a beam having a fixed direction and a beam which rotates in direction.

24. The apparatus according to claim 14, further comprising noise gate means for gating said main signal adaptively filtered by said adaptive filter means by opening said noise gate means when a signal-to-noise ratio at the near end is above a predetermined threshold and closing said noise gate means when said signal-to-noise ratio at the near end is below the predetermined threshold; wherein said noise gate means determines said predetermined threshold by selecting a low threshold when a signal-to-noise ratio of said reference signal from the far end is low, updating said predetermined threshold upwards when said signal-to-noise ratio of said reference signal of the far end goes up and gradually reducing said predetermined threshold when said signal-to-noise ratio of the reference signal at the far end goes down.

25. An interference canceling method for canceling, from a target signal generated from a target source, an interference signal generated by an interference source, said method comprising the steps of:

inputting said target signal;

inputting said interference signal;

beam-splitting said target signal into a plurality of band-limited target signals and beam-splitting said interference signal into band-limited interference signals, wherein the amount and frequency of band-limited target signals ~~equals~~ equal the amount and frequency of band-limited interference signals, whereby for each band-limited target signal there is a corresponding band-limited interference signal; and

adaptively filtering, each band-limited interference signal from each corresponding band-limited target signal.

26. The method according to claim 25, wherein said target signal represents speech generated at a near end of a teleconference, said reference signal represents said target signal broadcast from a far end of said teleconference and said interference signal represents an echo generated by said broadcast of said reference signal of said far end.

27. The method according to claim 26, wherein said step of adaptive filtering filters said band-limited target signals separately according to the frequency band.

28. The method according to claim 26, wherein said step of adaptive filtering estimates a transfer function of said reference signal broadcast of said far end.

29. The method according to claim 28, further comprising the step of permitting said step of adaptive filtering to include changing coefficients when a signal-to-noise ratio of said reference signal exceeds a predetermined threshold over a signal-to-noise ratio of said main signal.

30. The method according to claim 29, wherein said step of inhibiting determines said predetermined threshold periodically.

31. The method according to claim 26, wherein said step of beam splitting performs beam splitting using a DFT filter bank with single side band modulation.

32. The method according to claim 26, further comprising the step of beam selecting at least one of a plurality of beams for adaptive filtering in said step of adaptive filtering representing a direction from which said main signal is received.

33. The method according to claim 32, wherein said step of adaptive filtering updates coefficients representing said transform function and comprehensively stores said coefficients for each beam selected in said step of beam selecting.

34. The method according to claim 32, wherein said step of beam selecting selects said plurality of said beams for simultaneous adaptive filtering in said step of adaptive filtering.

35. The method according to claim 34, wherein said step of beam selecting selects a beam having a fixed direction and a beam which rotates in direction.

36. The method according to claim 26, further comprising the step of gating said main signal adaptively filtered in said step of adaptive filtering by opening a noise gate when a signal-to-noise ratio at the near end is above a predetermined threshold and closing said noise gate when said signal-to-noise ratio at the near end is below the predetermined threshold.

37. The method according to claim 36, further comprising the step of determining said predetermined threshold by selecting a low threshold when a signal-to-noise ratio of said reference signal at the far end is low, updating said predetermined threshold upwards when said signal-to-noise ratio of said reference signal at the far end goes up and gradually reducing said predetermined threshold when said signal-to-noise ratio of the reference signal from the far end goes down.

157035

PATENT
670025-7007

ABSTRACT OF THE DISCLOSURE

Interference canceling is provided for canceling, from a target signal generated from a target source, an interference signal generated by an interference source. The beam splitter beam-splits the target signal into a plurality of band-limited target signals band-limited frequency bands and beam-splits the interference signal into corresponding band-limited frequency bands. The adaptive filter adaptively filters each band-limited interference signal from each corresponding band-limited target signal. The inhibitor can permit the adaptive filter to adapt or change coefficients when a signal-to-noise ratio of the reference signal exceeds a predetermined threshold, to be determined periodically, over a signal-to-noise ratio of the main signal. The beam selector selects at least one of a plurality of beams for adaptive filtering by the adaptive filter representing a direction from which the main signal is received. The beam selector selects beams simultaneously to improve accuracy and, in particular, selects a beam having a fixed direction and a beam which rotates in direction. The noise gate gates the main signal adaptively filtered by the adaptive filter by opening the noise gate when a signal-to-noise ratio at the near end is above a predetermined threshold and closing the noise gate when the signal-to-noise ratio at the near end is below the predetermined threshold. When the target signal represents speech generated at a near end of a teleconference, the adaptive filter cancels an echo present in the reference signal broadcast to a far end of the teleconference.

DECLARATION FOR PATENT APPLICATION AND POWER OF ATTORNEY

(Includes reference to PCT International Applications)

FROMMER LAWRENCE & HAUG, LLP
File No.: 670025-7007

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe I am an original, first and joint inventor (if plural, names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention ENTITLED: *INTERFERENCE CANCELING METHOD AND APPARATUS*

the specification of which:

☐ is attached hereto
x was filed on SEPTEMBER 18, 1998 as:
x United States Application Serial No. ____________________
☐ PCT Application No. ___________
x with amendments through DATE EVEN HEREWITH (if applicable, give details).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose to the United States Patent and Trademark Office all information known to me to be material to patentability as defined in Title 37, Code of Federal Regulations, § 1.56.

I hereby claim foreign priority benefits under Title 35, United States Code § 119 (a) - (d) or § 365 (b) of any foreign application(s) for patent or inventor's certificate, or § 365 (a) of any PCT International application(s) designating at least one country other than the United State of America listed below and have also identified below any foreign application for patent or inventor's certificate or any PCT International applications designating at least one country other than the United States of America filed by me on the same subject matter having a filing date before that of the application(s) on which priority is claimed:

Prior Foreign/PCT Application(s) [list additional applications on separate page]:

Country (or PCT)	Application Number:	Filed (Day/Month/Year)	Priority Claimed: Yes	Priority Claimed: No

I hereby claim the benefit under 35 U.S.C. § 119(e) of any United States provisional application(s) listed below.

____________________ ____________________
(Application Number) (Filing Date)

I hereby claim the benefit under Title 35, United States Code § 120 of any United States application(s) or § 365 (c) of any PCT international application(s) designating the United States of America that is/are listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in that/those prior United States or PCT International application(s) in the manner provided by the first paragraph of Title 35, United States Code § 112, I acknowledge the duty to disclose to the United States Patent and Trademark Office all information known to me to be

material to patentability as defined in Title 37, Code of Federal Regulations, § 1.56 which became available between the filing date of the prior application and the national or PCT international filing date of this application:
Prior U.S. (or U.S.-designating PCT) Application(s) [list additional applications on separate page]:

U.S. Serial No.:	Filed (Day/Month/Year)	PCT Application No.	Status (patented, pending, abandoned)

I hereby appoint Thomas J. Kowalski, Registration No. 32,147, and I. Marc Asperas, Registration No. 37,274, and FROMMER LAWRENCE & HAUG, LLP or their duly appointed associates, my attorneys or agents, with full power of substitution and revocation, to prosecute this application, to make alterations and amendments therein, to file continuation and divisional applications thereof, to receive the Patent, and to transact all business in the Patent and Trademark Office and in the Courts in connection therewith, and to insert the Serial Number of the application in the space provided above, and specify that all communications about the application are to be directed to the following correspondence address:

Thomas J. Kowalski, Esq.
c/o FROMMER LAWRENCE & HAUG, LLP
745 Fifth Avenue
New York, NY 10151
FAX (212) 588-0500

Direct all telephone calls to:
(212) 588-0800
to the attention of:
Thomas J. Kowalski

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

INVENTOR(S):

Signature: ______________________________ Date: ______________

Full name of sole or first inventor: Joseph Marash
Residence: Haifa, Isreal; P.O. Box 7752, Haifa, Isreal 31077
Citizenship: Isreali

Signature: ______________________________ Date: ______________

Full name of 2ND joint inventor (if any): Baruch Berdugo
Residence: Kiriat-Ata 28000, Isreal
Citizenship: Isreali

Post Office Address(es) of inventors [if different from residence]:

NOTE: In order to qualify for reduced fees available to Small Entities, each inventor and any other individual or entity having rights to the invention must also sign an appropriate separate "Verified Statement (Declaration) Claiming [or Supporting a Claim by Another for] Small Entity Status" form [e.g. for Independent Inventor, Small Business Concern, Nonprofit Organization, Individual Non-Inventor].

PRINT OF DRAWINGS
AS ORIGINALLY FILE

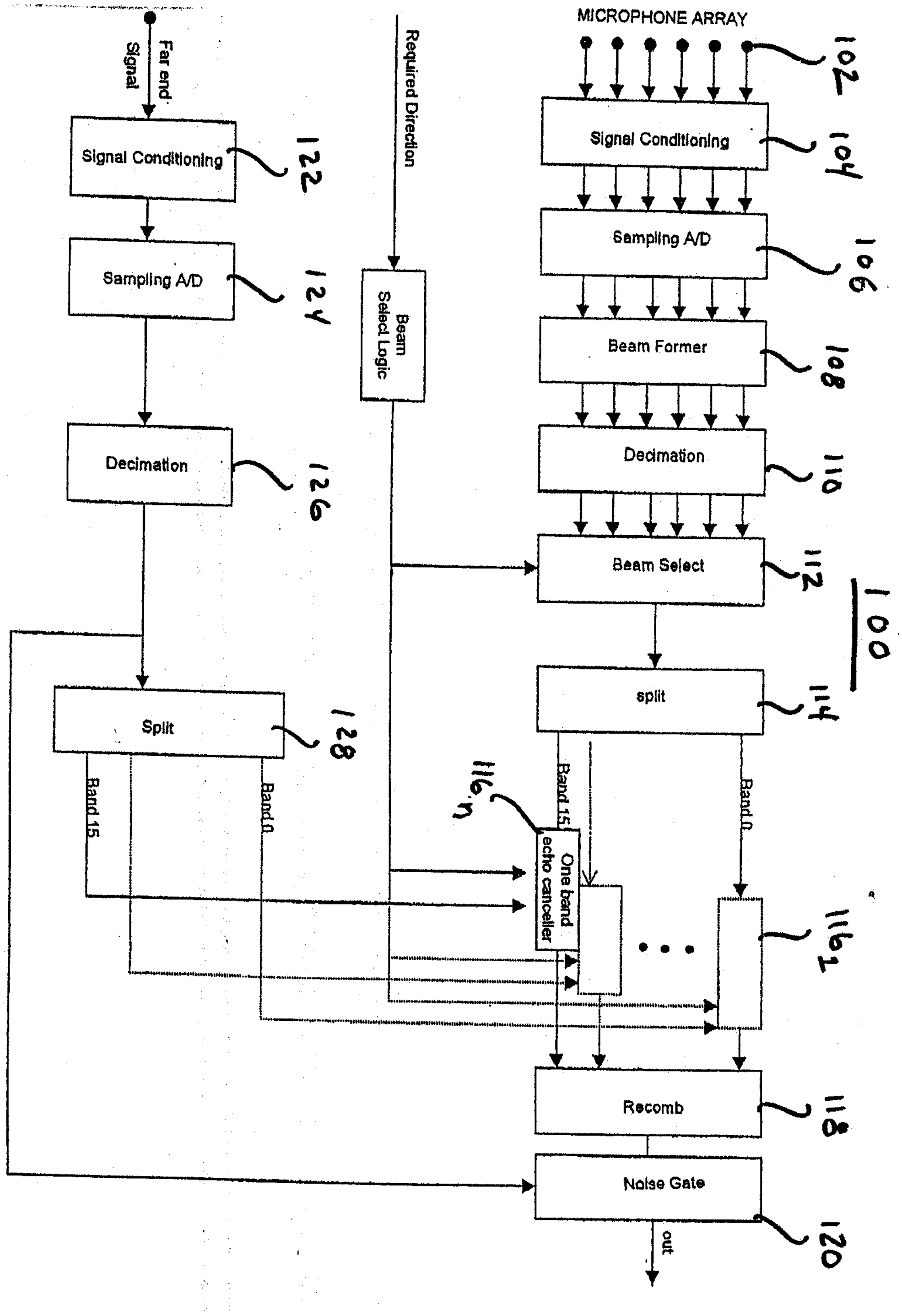

Fig. 1

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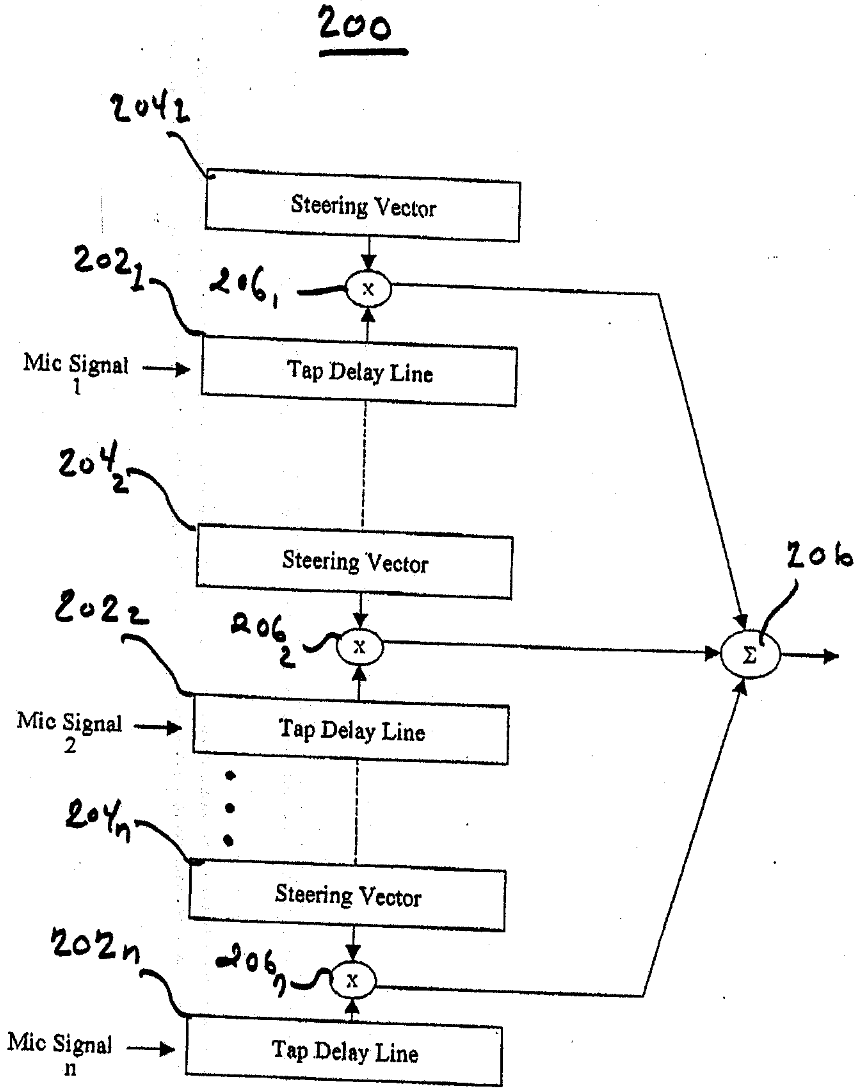

Fig. 2

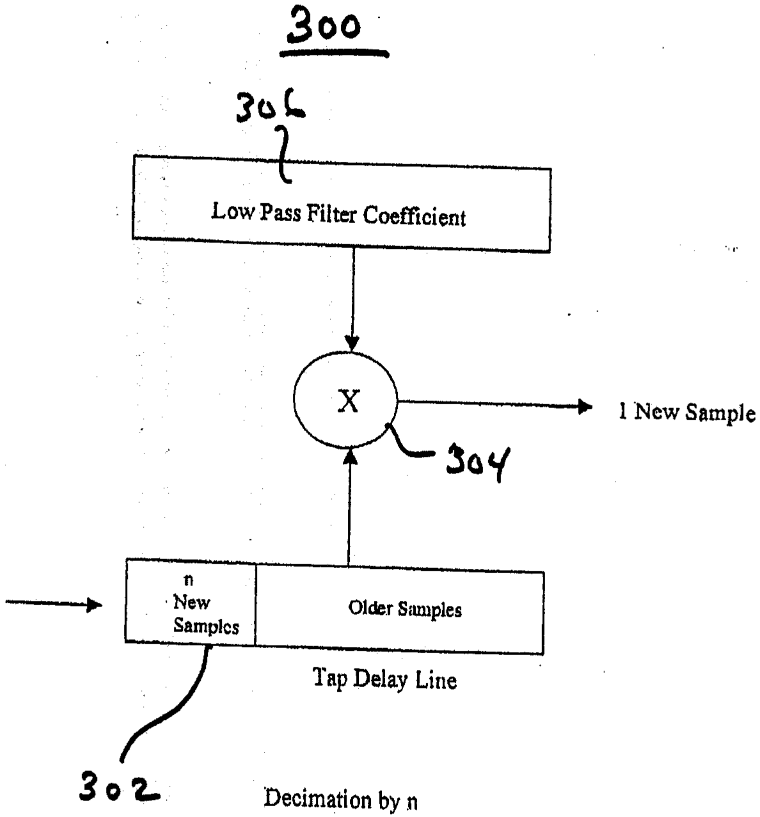
300
306
Low Pass Filter Coefficient
X
1 New Sample
304
n
New
Samples
Older Samples
Tap Delay Line
302
Decimation by n

Fig. 3

PRINT OF DRAWINGS AS ORIGINALLY FILED

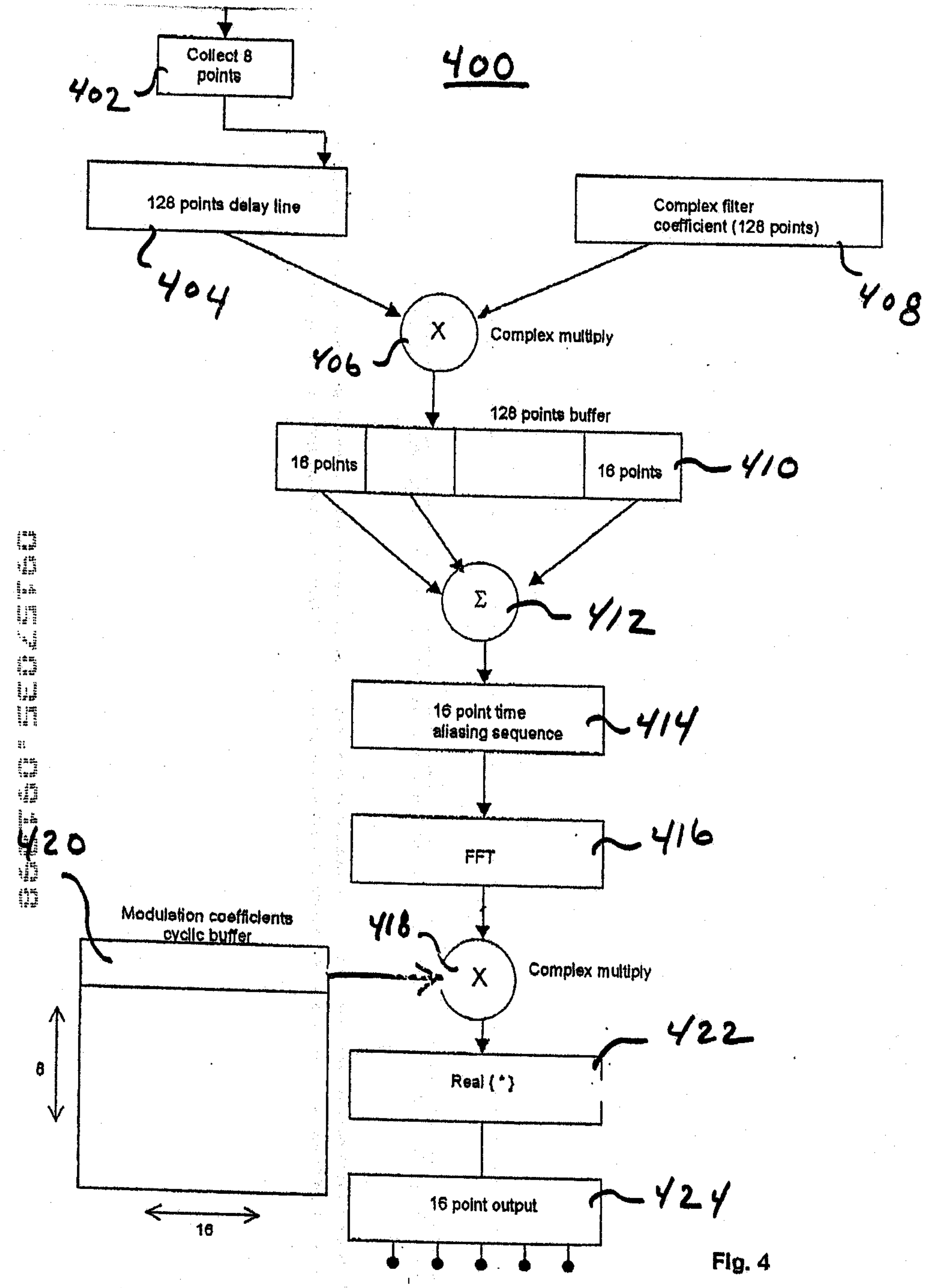

Fig. 4

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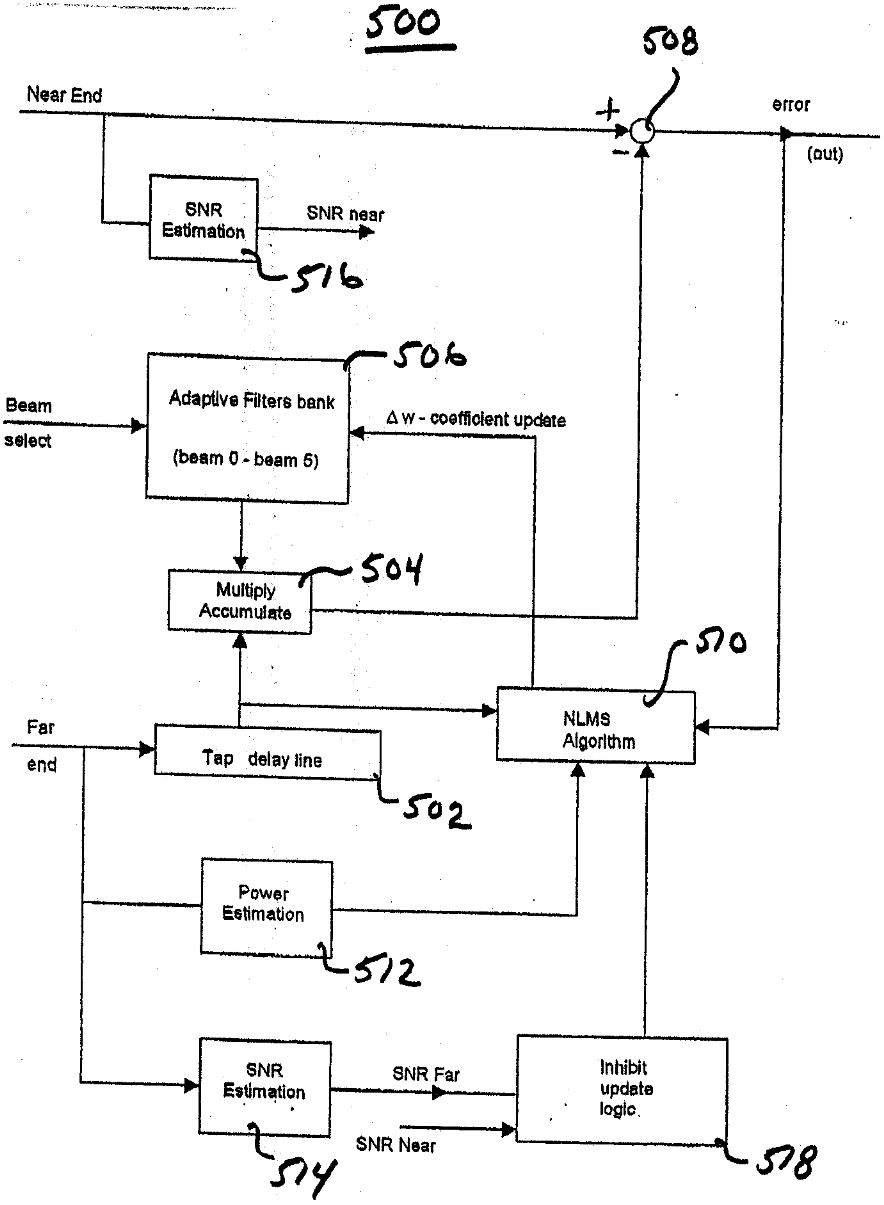

Fig. 5

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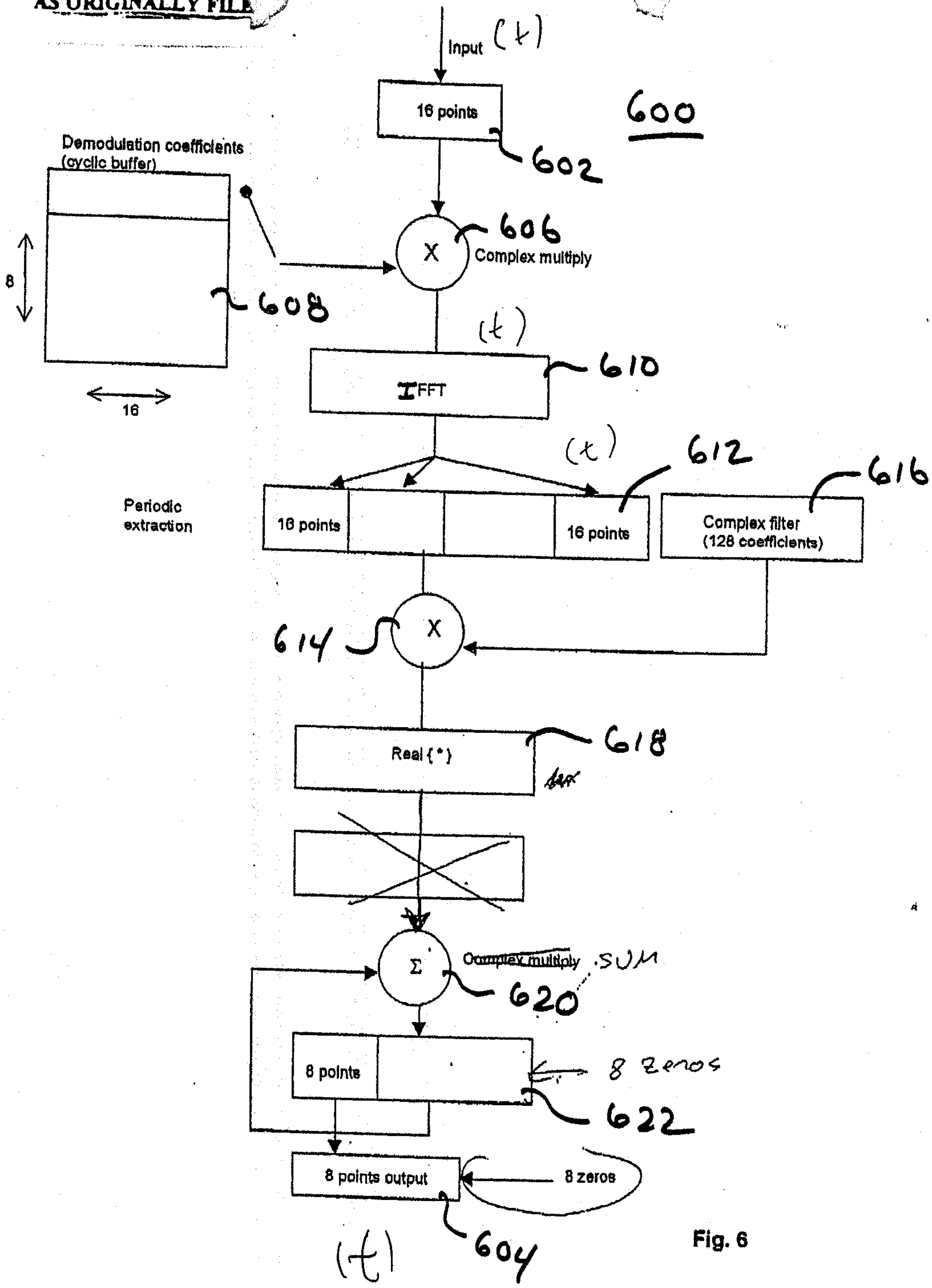

Fig. 6

PRINT OF DRAWINGS
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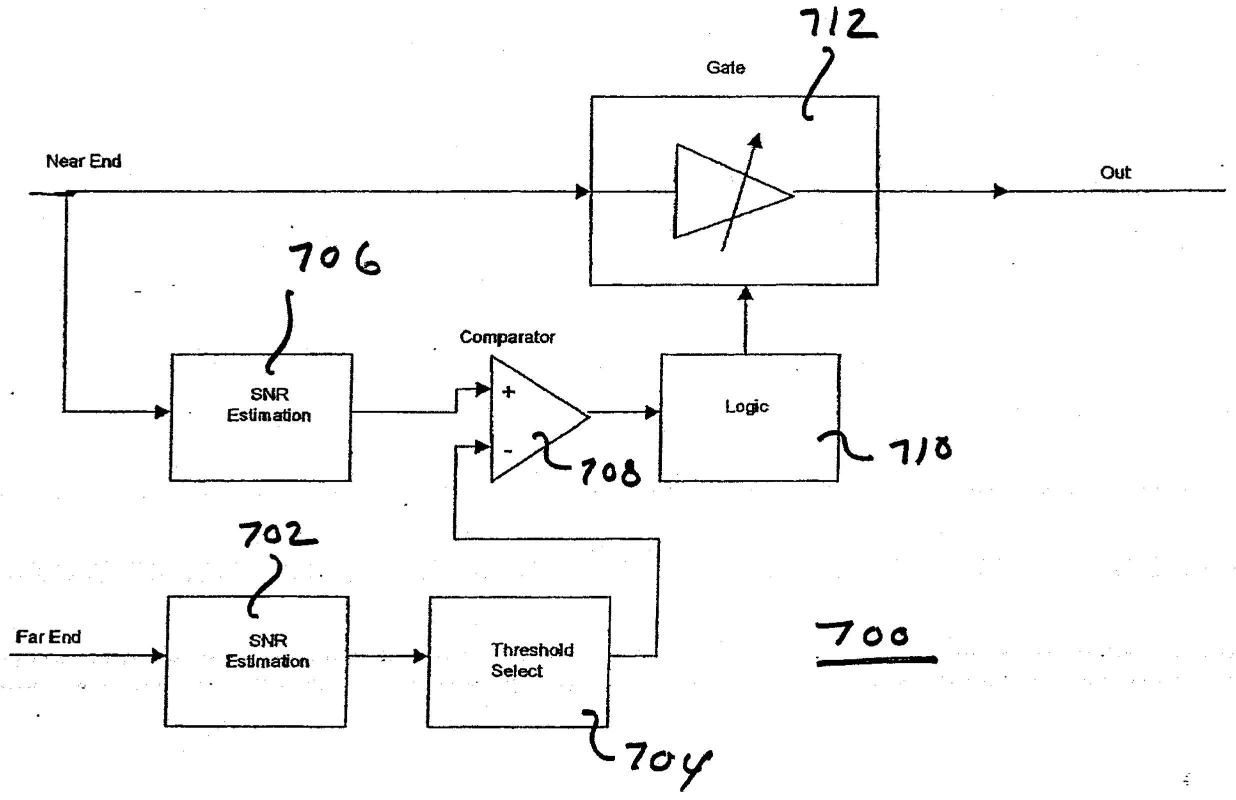

Fig. 7

09157035 .091898

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NUMBER	FILING/RECEIPT DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NO./TITLE
09/157,035	09/18/98	MARASH	J 670025-7007

#2

023271013

THOMAS J KOWALSKI
FROMMER LAWRENCE & HAUG
745 FIFTH AVENUE
NEW YORK NY 10151

NOT ASSIGNED

157035

2743

DATE MAILED: 10/13/98

9/18/98

NOTICE TO FILE MISSING PARTS OF APPLICATION
Filing Date Granted

An Application Number and Filing Date have been assigned to this application. The items indicated below, however, are missing. Applicant is given TWO MONTHS FROM THE DATE OF THIS NOTICE within which to file all required items and pay fees required below to avoid abandonment. Extensions of time may be obtained by filing a petition accompanied by the extension fee under the provisions of 37 CFR 1.136(a). **If any of items 1 or 3 through 5 are indicated as missing, the SURCHARGE set forth in 37 CFR 1.16(e) of ☐ $65.00 for a small entity in compliance with 37 CFR 1.27, or ☑ $130.00 for a non-small entity, must also be timely submitted in reply to this NOTICE to avoid abandonment.**

If all required items on this form are filed within the period set above, the total amount owed by applicant as a ☐ small entity (statement filed) ☑ non-small entity is $ 1294.00

☑ 1. The statutory basic filing fee is:
  - ☑ missing.
  - ☐ insufficient.

Applicant must submit $ 790.00 *to complete the basic filing fee and/or file a small entity statement claiming such status (37 CFR 1.27).*

☑ 2. Additional claim fees of $________, including any multiple dependent claim fees, are required.

$________ for ________ independent claims over 3.
$ 374.00 for 17 dependent claims over 20.
$________ for multiple dependent claim surcharge.

Applicant must either submit the additional claim fees or cancel additional claims for which fees are due.

☑ 3. The oath or declaration:
  - ☑ is missing or unexecuted.
  - ☐ does not cover the newly submitted items.
  - ☐ does not identify the application to which it applies.
  - ☐ does not include the city and state or foreign country of applicant's residence.

An oath or declaration in compliance with 37 CFR 1.63, including residence information and identifying the application by the above Application Number and Filing Date is required.

☐ 4. The signature(s) to the oath or declaration is/are by a person other than inventor or person qualified under 37 CFR 1.42, 1.43 or 1.47.
A properly signed oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.

☐ 5. The signature of the following joint inventor(s) is missing from the oath or declaration:
____
An oath or declaration in compliance with 37 CFR 1.63 listing the names of all inventors and signed by the omitted inventor(s), identifying this application by the above Application Number and Filing Date, is required.

☐ 6. A $50.00 processing fee is required since your check was returned without payment (37 CFR 1.21(m)).

☐ 7. Your filing receipt was mailed in error because your check was returned without payment.

☐ 8. The application does not comply with the Sequence Rules.
See attached "Notice to Comply with Sequence Rules 37 CFR 1.821-1.825."

☐ 9. OTHER: ________________________

Direct the reply and any questions about this notice to "Attention: Box Missing Parts."

A copy of this notice MUST be returned with the reply.

Customer Service Center
Initial Patent Examination Division (703) 308-1202

PART 3 - OFFICE COPY

FORM PTO-1533 (REV.9-97)

OIPE JC25
APR 13 1999
PATENT & TRADEMARK OFFICE

SECTION 7

PATENT
670025-7007

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

#3

Applicant(s) : Joseph Marash et al.

Serial No. : 09/157,035

For : INTERFERENCE CANCELING METHOD AND APPARATUS

Filed : September 18, 1998

Art Unit : 2743

745 Fifth Avenue
New York, NY 10151

I hereby certify that this correspondence is being deposited with the United States Postal Service as first class mail in an envelope addressed to: Assistant Commissioner for Patents, Washington, DC 20231, on April 9, 1999.

THOMAS J. KOWALSKI, Reg. No. 32,147

Name of Applicant, Assignee or Registered Representative

Thomas J. Kowalski

Signature

April 9, 1999

Date of Signature

COMMUNICATION

Assistant Commissioner for Patents
Washington, D.C. 20231

Dear Sir:

Attached is the original executed inventors' declaration and power of attorney along with the Notice to File Missing Parts mailed on October 13, 1998. Also enclosed is the Small Entity Declaration. A four month extension of the period for reply to the Notice under C.F.R. §1.136(a) and 1.17(a) is respectfully requested. A check is enclosed in the sum of

04/15/1999 CVORACHA 00000019 09157035
01 FC:218 680.00 OP

$1,327.00 to cover the cost of the Surcharge, the Filing Fee, the Additional Claims for a small entity ($647.00) and the four-month Extension of Time for a small entity ($686.00).

Please charge any fee deficiencies or credit any overpayment to Deposit Account No. 50-0320.

Respectfully submitted,
FROMMER LAWRENCE & HAUG LLP

By: Thomas J. Kowalski
THOMAS J. KOWALSKI
Reg. No. 32,147
(212) 588-0800

#3

OIPE C25 APR 13 1999 PATENT & TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NUMBER	FILING/RECEIPT DATE	FIRST NAMED APPLICANT	ATTORNEY DOCKET NO./TITLE
09/157,035	09/18/98	MARASH	J 670025-7007

OCT 15 AM 32

0232/1013

THOMAS J KOWALSKI
FROMMER LAWRENCE & HAUG
745 FIFTH AVENUE
NEW YORK NY 10151

NOT ASSIGNED

& HAUG, LLP

2743

DATE MAILED: 10/13/98

NOTICE TO FILE MISSING PARTS OF APPLICATION
Filing Date Granted

An Application Number and Filing Date have been assigned to this application. The items indicated below, however, are missing. Applicant is given TWO MONTHS FROM THE DATE OF THIS NOTICE within which to file all required items and pay fees required below to avoid abandonment. Extensions of time may be obtained by filing a petition accompanied by the extension fee under the provisions of 37 CFR 1.136(a). **If any of items 1 or 3 through 5 are indicated as missing, the SURCHARGE set forth in 37 CFR 1.16(e) of ☐ $65.00 for a small entity in compliance with 37 CFR 1.27, or ☑ $130.00 for a non-small entity, must also be timely submitted in reply to this NOTICE to avoid abandonment.**

If all required items on this form are filed within the period set above, the total amount owed by applicant as a ☐ small entity (statement filed) ☑ non-small entity is $ 1294.00

☑ 1. The statutory basic filing fee is:
  - ☑ missing.
  - ☐ insufficient.

  Applicant must submit $ 790.00 *to complete the basic filing fee and/or file a small entity statement claiming such status (37 CFR 1.27).*

☑ 2. Additional claim fees of $ ______, including any multiple dependent claim fees, are required.
  - $ ______ for ______ independent claims over 3.
  - $ 374.00 for 17 dependent claims over 20.
  - $ ______ for multiple dependent claim surcharge.

  Applicant must either submit the additional claim fees or cancel additional claims for which fees are due.

☑ 3. The oath or declaration:
  - ☑ is missing or unexecuted.
  - ☐ does not cover the newly submitted items.
  - ☐ does not identify the application to which it applies.
  - ☐ does not include the city and state or foreign country of applicant's residence.

  An oath or declaration in compliance with 37 CFR 1. 63, including residence information and identifying the application by the above Application Number and Filing Date is required.

☐ 4. The signature(s) to the oath or declaration is/are by a person other than inventor or person qualified under 37 CFR 1.42, 1.43 or 1.47.
A properly signed oath or declaration in compliance with 37 CFR 1.63, identifying the application by the above Application Number and Filing Date, is required.

☐ 5. The signature of the following joint inventor(s) is missing from the oath or declaration:

An oath or declaration in compliance with 37 CFR 1.63 listing the names of all inventors and signed by the omitted inventor(s), identifying this application by the above Application Number and Filing Date, is required.

☐ 6. A $50.00 processing fee is required since your check was returned without payment (37 CFR 1.21(m)).

☐ 7. Your filing receipt was mailed in error because your check was returned without payment.

☐ 8. The application does not comply with the Sequence Rules.
See attached "Notice to Comply with Sequence Rules 37 CFR 1.821-1.825."

☐ 9. OTHER:

Direct the reply and any questions about this notice to "Attention: Box Missing Parts."

A copy of this notice MUST be returned with the reply.

Customer Service Center
Initial Patent Examination Division (703) 308-1202

PART 2 - COPY TO BE RETURNED WITH RESPONSE

FORM PTO-1533 (REV.9-97)

04/15/1999 CVORACHA 00000020 09157035
01 FC:201 395.00 OP
02 FC:205 65.00 OP
03 FC:203 187.00 OP

OIPE JC25
APR 13 1999
PATENT & TRADEMARK OFFICE

Applicant or Patentee: MARASH, Joseph et al.
Serial or Patent No.: 09/157,035
Filed or Issued: September 18, 1998
For: **INTERFERENCE CANCELING METHOD AND APPARATUS**

FROMMER LAWRENCE & HAUG LLP
File No.: 670025-7007
Page 1 of 3

VERIFIED STATEMENT (DECLARATION) CLAIMING SMALL ENTITY STATUS (37 CFR 1.9(f) and 1.27(c)) - SMALL BUSINESS CONCERN

I hereby declare that I am

__ the owner of the small business concern identified below:

x an officer of the small business concern empowered to act on behalf of the concern identified below:

NAME OF CONCERN **LAMAR SIGNAL PROCESSSING LTD., a wholly owned subsidiary of ANDREA ELECTRONICS CORPORATION**

ADDRESS OF CONCERN
KOHAV YOKNEAM BUILDING, 5TH FLOOR
P.O. BOX 273
YOKNEAM 20692
ISRAEL

I hereby declare that the above-identified small business concern qualifies as a small business concern as defined in 13 CFR 121.12, and reproduced in 37 CFR 1.9(d), for purposes of paying reduced fees to the United States Patent and Trademark Office, in that the number of employees of the concern, including those of its affiliates, does not exceed 500 persons. For purposes of this statement, (1) the number of employees of the business concern is the average over the previous fiscal year of the concern of the persons employed on a full-time, part-time or temporary basis during each of the pay periods of the fiscal year, and (2) concerns are affiliates of each other when either, directly or indirectly, one concern controls or has the power to control the other, or a third party or parties controls or has the power to control both.

I hereby declare that rights under contract or law have been conveyed to and remain with the small business concern identified above with regard to the invention, entitled **INTERFERENCE CANCELING METHOD AND APPARATUS** by inventor(s) **MARASH, Joseph and BERDUGO, Baruch** described in

__ the specification filed herewith.

X application serial no. 09/157,035, filed **September 18, 1998**.

__ patent no. __, issued __.

If the rights held by the above-identified small business concern are not exclusive, each individual, concern or organization having rights to the invention is listed below* and no rights to the invention are held by any person, other than the inventor, who would not qualify as an independent inventor under 37 CFR 1.9(c) if that person made the invention, or by any concern which would not qualify as a small business concern under 37 CFR 1.9(d) or a nonprofit organization under 37 CFR 1.9(e).

ANDREA.7\7007SM.ENT

Applicant or Patentee: **MARASH, Joseph et al.** **FROMMER LAWRENCE & HAUG LLP**

Serial or Patent No. 09/157,035 **File No.: 670025-7007**

Filed or Issued: September 18, 1998 **Page 2 of 3**

For: **INTERFERENCE CANCELING METHOD AND APPARATUS**

*NOTE: Separate verified statements are required from each named person, concern or organization having rights to the invention averring to their status as small entities (37 CFR 1.27).

FULL NAME **LAMAR SIGNAL LTD.**

ADDRESS
KOHAV YOKNEAM BUILDING, 5TH FLOOR
P.O. BOX 273
YOKNEAM 20692
ISRAEL

__ INDIVIDUAL X SMALL BUSINESS CONCERN __ NONPROFIT ORGANIZATION

FULL NAME ______

ADDRESS ______

__ INDIVIDUAL __ SMALL BUSINESS CONCERN __ NONPROFIT ORGANIZATION

FULL NAME ______

ADDRESS ______

__ INDIVIDUAL __ SMALL BUSINESS CONCERN __ NONPROFIT ORGANIZATION

I acknowledge the duty to file, in this provisional application or patent, notification of any change in status resulting in loss of entitlement to small entity status prior to paying, or at the time of paying, the earliest of the issue fee or any maintenance fee due after the date on which status as a small entity is no longer appropriate. (37 CFR 1.28(b))

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code, and that such willful false statements may jeopardize the validity of

ANDREA.7\7007SM.ENT

the application, any patent issuing thereon, or any patent to which this verified statement is directed.

NAME OF PERSON SIGNING **MARASH, Joseph**

TITLE OF PERSON
(if other than owner) President

ADDRESS OF PERSON SIGNING
KOHAV YOKNEAM BUILDING, 5TH FLOOR
P.O. BOX 273
YOKNEAM 20692
ISRAEL

SIGNATURE Joseph Marash DATE 1/31/99

ANDREA.7\7007SM.ENT

#3

DECLARATION FOR PATENT APPLICATION AND POWER OF ATTORNEY

(Includes reference to PCT International Applications)

FROMMER LAWRENCE & HAUG, LLP
File No.: 670025-7007

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe I am an original, first and joint inventor (if plural, names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention ENTITLED: *INTERFERENCE CANCELING METHOD AND APPARATUS*

the specification of which:

☐ is attached hereto
X was filed on SEPTEMBER 18, 1998 as:
X United States Application Serial No. 09/157,035
☐ PCT Application No. ___________
X with amendments through DATE EVEN HEREWITH (if applicable, give details).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims, as amended by any amendment referred to above.

I acknowledge the duty to disclose to the United States Patent and Trademark Office all information known to me to be material to patentability as defined in Title 37, Code of Federal Regulations, § 1.56.

I hereby claim foreign priority benefits under Title 35, United States Code § 119 (a) - (d) or § 365 (b) of any foreign application(s) for patent or inventor's certificate, or § 365 (a) of any PCT International application(s) designating at least one country other than the United State of America listed below and have also identified below any foreign application for patent or inventor's certificate or any PCT International applications designating at least one country other than the United States of America filed by me on the same subject matter having a filing date before that of the application(s) on which priority is claimed:

Prior Foreign/PCT Application(s) [list additional applications on separate page]:

Country (or PCT)	Application Number:	Filed (Day/Month/Year)	Priority Claimed: Yes	No

I hereby claim the benefit under 35 U.S.C. § 119(e) of any United States provisional application(s) listed below.

(Application Number) (Filing Date)

I hereby claim the benefit under Title 35, United States Code § 120 of any United States application(s) or § 365 (c) of any PCT international application(s) designating the United States of America that is/are listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in that/those prior United States or PCT International application(s) in the manner provided by the first paragraph of

DECLARATION FOR PATENT APPLICATION
AND POWER OF ATTORNEY

FLH Docket No. 670025-7007

Title 35, United States Code § 112, I acknowledge the duty to disclose to the United States Patent and Trademark Office all information known to me to be material to patentability as defined in Title 37, Code of Federal Regulations, § 1.56 which became available between the filing date of the prior application and the national or PCT international filing date of this application:
Prior U.S. (or U.S.-designating PCT) Application(s) [list additional applications on separate page]:

U.S. Serial No.:	Filed (Day/Month/Year)	PCT Application No.	Status (patented, pending, abandoned)

I hereby appoint Thomas J. Kowalski, Registration No. 32,147, and FROMMER LAWRENCE & HAUG, LLP or their duly appointed associates, my attorneys or agents, with full power of substitution and revocation, to prosecute this application, to make alterations and amendments therein, to file continuation and divisional applications thereof, to receive the Patent, and to transact all business in the Patent and Trademark Office and in the Courts in connection therewith, and to insert the Serial Number of the application in the space provided above, and specify that all communications about the application are to be directed to the following correspondence address:

Thomas J. Kowalski, Esq.
c/o FROMMER LAWRENCE & HAUG, LLP
745 Fifth Avenue
New York, NY 10151
FAX (212) 588-0500

Direct all telephone calls to:
(212) 588-0800
to the attention of:
Thomas J. Kowalski

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

INVENTOR(S):

Signature: Joseph Mar Date: 1/31/99

Full name of sole or first inventor: JOSEPH MARASH
Residence: 1A Shimkin Street, Haifa 34750 Israel
Citizenship: Israel

Signature: B. Berdugo Date: 1/31/99

Full name of 2ND joint inventor: BARUCH BERDUGO
Residence: 6 Hanarkisim Street, Kiriat-Ata 28000, Israel
Citizenship: Israel

Post Office Address(es) of inventors [if different from residence]:
P.O. Box 273, Yokream 20692, Israel

NOTE: In order to qualify for reduced fees available to Small Entities, each inventor and any other individual or entity having rights to the invention must also sign an appropriate separate "Verified Statement (Declaration) Claiming [or Supporting a Claim by Another for] Small Entity Status" form [e.g. for Independent Inventor, Small Business Concern, Nonprofit Organization, Individual Non-Inventor].

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office

NOTICE OF ALLOWANCE AND ISSUE FEE DUE

LM41/1221

THOMAS J KOWALSKI
FROMMER LAWRENCE & HAUG
745 FIFTH AVENUE
NEW YORK NY 10151

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED
09/157,035	09/18/98	037	SAINT SURIN, J 2747	12/21/99

First Named Applicant MARASH, 35 USC 154(b) term ext. = 0 Days.

TITLE OF INVENTION INTERFERENCE CANCELING METHOD AND APPARATUS

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
2 670025-7007	379-407.000	G01	UTILITY	YES	$605.00	03/21/00

THE APPLICATION IDENTIFIED ABOVE HAS BEEN EXAMINED AND IS ALLOWED FOR ISSUANCE AS A PATENT. PROSECUTION ON THE MERITS IS CLOSED.

THE ISSUE FEE MUST BE PAID WITHIN THREE MONTHS FROM THE MAILING DATE OF THIS NOTICE OR THIS APPLICATION SHALL BE REGARDED AS ABANDONED. THIS STATUTORY PERIOD CANNOT BE EXTENDED.

HOW TO RESPOND TO THIS NOTICE:

I. Review the SMALL ENTITY status shown above.
If the SMALL ENTITY is shown as YES, verify your current SMALL ENTITY status:

A. If the status is changed, pay twice the amount of the FEE DUE shown above and notify the Patent and Trademark Office of the change in status, or

B. If the status is the same, pay the FEE DUE shown above.

If the SMALL ENTITY is shown as NO:

A. Pay FEE DUE shown above, or

B. File verified statement of Small Entity Status before, or with, payment of 1/2 the FEE DUE shown above.

II. Part B-Issue Fee Transmittal should be completed and returned to the Patent and Trademark Office (PTO) with your ISSUE FEE. Even if the ISSUE FEE has already been paid by charge to deposit account, Part B Issue Fee Transmittal should be completed and returned. If you are charging the ISSUE FEE to your deposit account, section "4b" of Part B-Issue Fee Transmittal should be completed and an extra copy of the form should be submitted.

III. All communications regarding this application must give application number and batch number.
Please direct all communications prior to issuance to Box ISSUE FEE unless advised to the contrary.

IMPORTANT REMINDER: Utility patents issuing on applications filed on or after Dec. 12, 1980 may require payment of maintenance fees. It is patentee's responsibility to ensure timely payment of maintenance fees when due.

PATENT AND TRADEMARK OFFICE COPY

PTOL-85 (REV. 10-96) Approved for use through 06/30/99. (0651-0033)

*U.S. GPO: 1999-454-457/24601

B4

PART B—ISSUE FEE TRANSMITTAL

Complete and mail this form, together with applicable fees, to:

Box ISSUE FEE
Assistant Commissioner for Patents
Washington, D.C. 20231

MAILING INSTRUCTIONS: This form should be used for transmitting the ISSUE FEE. Blocks 1 through 4 should be completed where appropriate. All further correspondence including the Issue Fee Receipt, the Patent, advance orders and notification of maintenance fees will be mailed to the current correspondence address as indicated unless corrected below or directed otherwise in Block 1, by (a) specifying a new correspondence address; and/or (b) indicating a separate "FEE ADDRESS" for maintenance fee notifications.

CURRENT CORRESPONDENCE ADDRESS (Note: Legibly mark-up with any corrections or use Block 1)

M4J/1221

THOMAS J KOWALSKI
FROMMER LAWRENCE & HAUG
745 FIFTH AVENUE
NEW YORK NY 10151

OIPE JC3
FEB 1 5 2000
PATENT & TRADEMARK OFFICE

Note: The certificate of mailing below can only be used for domestic mailings of the Issue Fee Transmittal. This certificate cannot be used for any other accompanying papers. Each additional paper, such as an assignment or formal drawing, must have its own certificate of mailing.

Certificate of Mailing

I hereby certify that this Issue Fee Transmittal is being deposited with the United States Postal Service with sufficient postage for first class mail in an envelope addressed to the Box Issue Fee address above on the date indicated below.

THOMAS J. KOWALSKI Reg. No. 32,147 (Depositor's name)

Thomas J. Kowalski (Signature)

(Date)

APPLICATION NO.	FILING DATE	TOTAL CLAIMS	EXAMINER AND GROUP ART UNIT	DATE MAILED
09/157,035	09/18/98	037	SAINT SURIN, J 2747	12/21/99

First Named Applicant MARASH, 35 USC 154(b) term ext. = 0 Days.

TITLE OF INVENTION INTERFERENCE CANCELING METHOD AND APPARATUS

ATTY'S DOCKET NO.	CLASS-SUBCLASS	BATCH NO.	APPLN. TYPE	SMALL ENTITY	FEE DUE	DATE DUE
2 670025-7007	379-407.000	G01	UTILITY	YES	$605.00	03/21/00

1. Change of correspondence address or indication of "Fee Address" (37 CFR 1.363). Use of PTO form(s) and Customer Number are recommended, but not required.

☐ Change of correspondence address (or Change of Correspondence Address form PTO/SB/122) attached.

☐ "Fee Address" indication (or "Fee Address" Indication form PTO/SB/47) attached.

2. For printing on the patent front page, list (1) the names of up to 3 registered patent attorneys or agents OR, alternatively, (2) the name of a single firm (having as a member a registered attorney or agent) and the names of up to 2 registered patent attorneys or agents. If no name is listed, no name will be printed.

1 FROMMER LAWRENCE & HAUG LLP
2 THOMAS J. KOWALSKI
3 ____

3. ASSIGNEE NAME AND RESIDENCE DATA TO BE PRINTED ON THE PATENT (print or type)
PLEASE NOTE: Unless an assignee is identified below, no assignee data will appear on the patent. Inclusion of assignee data is only appopriate when an assignment has been previously submitted to the PTO or is being submitted under separate cover. Completion of this form is NOT a subsititue for filing an assignment.

(A) NAME OF ASSIGNEE LAMAR SIGNAL PROCESSING, a wholly owned subsidiary of ANDREA ELECTRONICS CORPORATION

(B) RESIDENCE: (CITY & STATE OR COUNTRY) Yokneam, Isreal

Please check the appropriate assignee category indicated below (will not be printed on the patent)

☐ individual ☒ corporation or other private group entity ☐ government

4a. The following fees are enclosed (make check payable to Commissioner of Patents and Trademarks):

☒ Issue Fee
☒ Advance Order - # of Copies 10

4b. The following fees or deficiency in these fees should be charged to:

DEPOSIT ACCOUNT NUMBER 05-0320
(ENCLOSE AN EXTRA COPY OF THIS FORM)

☑ Issue Fee
☒ Advance Order - # of Copies 10

The COMMISSIONER OF PATENTS AND TRADEMARKS IS requested to apply the Issue Fee to the application identified above.

(Authorized Signature)		(Date)
Thomas J. Kowalski	Thomas J. Kowalski Reg. No. 32,147	8 Feb 2000

NOTE; The Issue Fee will not be accepted from anyone other than the applicant; a registered attorney or agent; or the assignee or other party in interest as shown by the records of the Patent and Trademark Office.

Burden Hour Statement: This form is estimated to take 0.2 hours to complete. Time will vary depending on the needs of the individual case. Any comments on the amount of time required to complete this form should be sent to the Chief Information Officer, Patent and Trademark Office, Washington, D.C. 20231. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. SEND FEES AND THIS FORM TO: Box Issue Fee, Assistant Commissioner for Patents, Washington D.C. 20231

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

BERHE1 00000148 09157035
605.00 OP
30.00 OP

TRANSMIT THIS FORM WITH FEE

PTOL-85B (REV.10-96) Approved for use through 06/30/99. OMB 0651-0033

Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

UNITED STATES DEPARTMENT OF COMMERCE
Patent and Trademark Office
Address: COMMISSIONER OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.
[illegible]	1[illegible]/98	MA[illegible]	[illegible]07

LM41/1221

THOMAS J KOWALSKI
FROMMER LAWRENCE & HAUG
745 FIFTH AVENUE
NEW YORK NY 10151

EXAMINER
SAI[illegible]RIN, J

ART UNIT	PAPER NUMBER
[illegible]47	4

DATE MAILED: 12/21/99

Please find below and/or attached an Office communication concerning this application or proceeding.

Commissioner of Patents and Trademarks

PTO-90C (Rev. 2/95)

1- File Copy

EXAMINER'S AMENDMENT

1. An examiner's amendment to the record appears below. Should the changes and/or additions be unacceptable to applicant, an amendment may be filed as provided by 37 CFR 1.312. To ensure consideration of such an amendment, it MUST be submitted no later than the payment of the issue fee.

Authorization for this examiner's amendment was given in a telephone interview with Thomas J. Kowalski on 12/3/99.

2. The application has been amended as follows:

In the Claims:

In claim 1, line 8, after "signals", delete "equals", insert --equal--.

In claim 13, line 8, after "signals", delete "equals", insert --equal--.

In claim 25, line 8, after "signals", delete "equals", insert --equal--.

REASONS FOR ALLOWANCE

3. The following is an examiner's statement of reasons for allowance: The prior art of record taken alone and in combination does not disclose "a beam splitter for beam-splitting said target signal into a plurality of band-limited target signals and beam-splitting said interference signal into band-limited interference signals, wherein the amount and frequency of band-limited target signals equal the amount and frequency of band-limited interference signals, whereby for each band-limited target signal there is a corresponding band-limited interference signal" recited in claim 1.

4. Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled "Comments on Statement of Reasons for Allowance."

5. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Jacques M. Saint-Surin whose telephone number is (703) 305-4760. The examiner can normally be reached on Mondays through Thursdays from 8:30 A.M. to 6:00 P.M.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Forester W. Isen, can be reached on (703) 305-4386. The fax phone number for this Group is (703) 308-5403.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group receptionist whose telephone number is (703) 305-3900 .

FORESTER W. ISEN
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2700

JACQUES SAINT-SURIN
PATENT EXAMINER

Jacques M. Saint-Surin
December 3, 1999

Interview Summary

Application No. 09/157,035	Applicant(s) **Joseph Marash et al.**	
Examiner **Jacques M. Saint-Surin**	Group Art Unit **2747**	

All participants (applicant, applicant's representative, PTO personnel):

(1) *Jacques M. Saint-Surin*

(2) *Thomas P. Kowalski*

(3)

(4)

Date of Interview *Dec 3, 1999*

Type: ☒ Telephonic ☐ Personal (copy is given to ☐ applicant ☐ applicant's representative).

Exhibit shown or demonstration conducted: ☐ Yes ☒ No. If yes, brief description:

Agreement ☒ was reached. ☐ was not reached.

Claim(s) discussed: *1, 13, and 25*

Identification of prior art discussed:

Description of the general nature of what was agreed to if an agreement was reached, or any other comments:
The Examiner called Applicant's representative to discuss some minor informalities in claims 1, 13 and 25. It was agreed to overcome the matters upon an examiner's amendment.

(A fuller description, if necessary, and a copy of the amendments, if available, which the examiner agreed would render the claims allowable must be attached. Also, where no copy of the amendents which would render the claims allowable is available, a summary thereof must be attached.)

1. ☒ It is not necessary for applicant to provide a separate record of the substance of the interview.

Unless the paragraph above has been checked to indicate to the contrary, A FORMAL WRITTEN RESPONSE TO THE LAST OFFICE ACTION IS NOT WAIVED AND MUST INCLUDE THE SUBSTANCE OF THE INTERVIEW. (See MPEP Section 713.04). If a response to the last Office action has already been filed, APPLICANT IS GIVEN ONE MONTH FROM THIS INTERVIEW DATE TO FILE A STATEMENT OF THE SUBSTANCE OF THE INTERVIEW.

2. ☐ Since the Examiner's interview summary above (including any attachments) reflects a complete response to each of the objections, rejections and requirements that may be present in the last Office action, and since the claims are now allowable, this completed form is considered to fulfill the response requirements of the last Office action. Applicant is not relieved from providing a separate record of the interview unless box 1 above is also checked.

Examiner Note: You must sign and stamp this form unless it is an attachment to a signed Office action.

U. S. Patent and Trademark Office
PTO-413 (Rev. 10-95)

Interview Summary

Paper No. 4

FORM PTO 948 (REV. 11-97) U.S. DEPARTMENT OF COMMERCE-Patent and Trademark Office Application No. 157035

NOTICE OF DRAFTPERSON'S PATENT DRAWING REVIEW

The drawing filed (insert date) 9/18/98 are:

A. ______ not objected to by the Draftperson under 37 CFR 1.84 or 1.152.

B. X objected to by the Draftperson under 37 CFR 1.84 or 1.152 as indicated below. The Examiner will require submission of new, corrected drawings when necessary. Corrected drawings must be submitted according to the instructions on the back of this notice.

1. DRAWINGS. 37 CFR 1.84(a): Acceptable categories of drawings: Black ink. Color.
   - ______ Color drawing are not acceptable until petition is granted. Fig.(s) ______
   - ______ Pencil and non black ink is not permitted. Fig(s) ______
2. PHOTOGRAPHS. 37 CFR 1.84(b)
   - ______ Photographs are not acceptable until petition is granted.
   - ______ 3 full-tone sets are required. Fig(s) ______
   - ______ Photographs not properly mounted (must brystol board or photographic double-weight paper). Fig(s) ______
   - ______ Poor quailty (half-tone). Fig(s) ______
3. TYPE OF PAPER. 37 CFR 1.84(e)
   - ______ Paper not flexible, strong, white and durable. Fig.(s) ______
   - ______ Erasures, alterations, overwritings, interlineations, folds, copy machine marks not acceptable. (too thin)
   - ______ Mylar, vellum paper is not acceptable (too thin). Fig(s) ______
4. SIZE OF PAPER. 37 CFR 1.84(F): Acceptable sizes:
   - ______ 21.0 cm by 29.7 cm (DIN size A4)
   - ______ 21.6 cm by 27.9 cm (8 1/2 x 11 inches)
   - ______ All drawings sheets not the same size. Sheet(s) ______
5. MARGINS. 37 CFR 18.4(g): Acceptable margins:
   - Top 2.5 cm Left 2.5 cm Right 1.5 cm Bottom 1.0 cm SIZE: A4 Size
   - Top 2.5 cm Left 2.5 cm Right 1.5 cm Bottom 1.0 cm SIZE: 8 1/2 x 11
   - ______ Margins not acceptable. Fig(s) ______
   - ______ Top (T) ______ Left (L)
   - ______ Right (R) ______ Bottom (B)
6. VIEWS. CFR 1.84(h)
   REMINDER: Specification may require revision to correspond to drawing changes.
   - ______ Views connected by projection lines or lead lines. Fig.(s) ______

   Partial views. 37 CFR 1.84(h)(2)
   - ______ Brackets needed to show figure as one entity. Fig.(s) ______
   - ______ Views not labeled separately or properly. Fig.(s) ______
   - ______ Enlarged view not labeled separately or properly. Fig.(s) ______
7. SECTIONAL VIEWS. 37 CFR 1.84(h)(3)
   - ______ Hatching not indicated for sectional portions of an object. Fig.(s) ______
   - ______ Sectional designation should be noted with Arabic or Roman numbers. Fig.(s) ______
8. ARRANGEMENT OF VIEWS. 37 CFR 1.84(i)
   - ______ Words do not appear on a horizontal, left-to-right fashion when page is either upright or turned, so that the top becomes the right side, except for graphs. Fig.(s) ______
   - ______ Views not on the same plane on drawing sheet. Fig.(s) ______
9. SCALE. 37 CFR 1.84(k)
   - ______ Scale not large enough to show mechansim without crowding when drawing is reduced in size to two-thirds in reproduction. Fig.(s) ______
10. CHARACTER OF LINES, NUMBERS, & LETTERS. 37 CFR 1.84(l)
    - X Lines, numbers & letters not uniformly thick and well defined, clean, durable and black (poor line quality). Fig.(s) 1 – 7
11. SHADING. 37 CFR 1.84(m)
    - ______ Solid black areas pale. Fig.(s) ______
    - ______ Solid black shading not permitted. Fig.(s) ______
    - ______ Shade lines, pale, rough and blurred. Fig.(s)
12. NUMBERS, LETTERS, & REFERENCE CHARACTERS. 37 CFR 1.48(p)
    - ______ Numbers and reference characters not plain and legible. Fig.(s) ______
    - ______ Figure legends are poor. Fig.(s) ______
    - ______ Numbers and reference characters not oriented in the same direction as the view. 37 CFR 1.84(p)(3) Fig.(s) ______
    - ______ English alphabet not used. 37 CFR 1.84(p)(3) Fig.(s) ______
    - ______ Numbers, letters and reference characters must be at least .32 cm (1/8 inch) in height. 37 CFR 1.84(p)(3) Fig.(s) ______
13. LEAD LINES. 37 CFR 1.84(q)
    - ______ Lead lines cross each other. Fig.(s) ______
    - ______ Lead lines missing. Fig.(s) ______
14. NUMBERING OF SHEETS OF DRAWINGS. 37 CFR 1.48(t)
    - ______ Sheets not numbered consecutively, and in Ababic numerals beginning with number 1. Fig.(s) ______
15. NUMBERING OF VIEWS. 37 CFR 1.84(u)
    - ______ Views not numbered consecutively, and in Abrabic numerals, beginning with number 1. Fig.(s) ______
16. CORRECTIONS. 37 CFR 1.84(w)
    - ______ Corrections not made from PTO-948 dated ______
17. DESIGN DRAWINGS. 37 CFR 1.152
    - ______ Surface shading shown not appropriate. Fig.(s) ______
    - ______ Solid black shading not used for color contrast. Fig.(s) ______

COMMENTS

REVIEWER ______ DATE 5/13/99 TELEPHONE NO. 703 308 0895

ATTACHMENT TO PAPER NO. 4

PTO COPY

#4/ A
Lu [illegible]
07-21-00

Notice of Allowability	Application No. **09/157,035**	Applicant(s) **Joseph Marash et al.**	
	Examiner **Jacques M. Saint-Surin**	Group Art Unit **2747**	

All claims being allowable, PROSECUTION ON THE MERITS IS (OR REMAINS) CLOSED in this application. If not included herewith (or previously mailed), a Notice of Allowance and Issue Fee Due or other appropriate communication will be mailed in due course.

☒ This communication is responsive to *9/18/96*

☒ The allowed claim(s) is/are *1-37*

☐ The drawings filed on ____________ are acceptable.

☐ Acknowledgement is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d).

☐ All ☐ Some* ☐ None of the CERTIFIED copies of the priority documents have been
- ☐ received.
- ☐ received in Application No. (Series Code/Serial Number) ____________.
- ☐ received in this national stage application from the International Bureau (PCT Rule 17.2(a)).

*Certified copies not received: ____________

☐ Acknowledgement is made of a claim for domestic priority under 35 U.S.C. § 119(e).

A SHORTENED STATUTORY PERIOD FOR RESPONSE to comply with the requirements noted below is set to EXPIRE **THREE MONTHS** FROM THE "DATE MAILED" of this Office action. Failure to timely comply will result in ABANDONMENT of this application. Extensions of time may be obtained under the provisions of 37 CFR 1.136(a).

☐ Note the attached EXAMINER'S AMENDMENT or NOTICE OF INFORMAL APPLICATION, PTO-152, which discloses that the oath or declaration is deficient. A SUBSTITUTE OATH OR DECLARATION IS REQUIRED.

☒ Applicant MUST submit NEW FORMAL DRAWINGS
- ☐ because the originally filed drawings were declared by applicant to be informal.
- ☒ including changes required by the Notice of Draftsperson's Patent Drawing Review, PTO-948, attached hereto or to Paper No. *4*.
- ☐ including changes required by the proposed drawing correction filed on ____________, which has been approved by the examiner.
- ☐ including changes required by the attached Examiner's Amendment/Comment.

Identifying indicia such as the application number (see 37 CFR 1.84(c)) should be written on the reverse side of the drawings. The drawings should be filed as a separate paper with a transmittal lettter addressed to the Official Draftsperson.

☐ Note the attached Examiner's comment regarding REQUIREMENT FOR THE DEPOSIT OF BIOLOGICAL MATERIAL.

Any response to this letter should include, in the upper right hand corner, the APPLICATION NUMBER (SERIES CODE/SERIAL NUMBER). If applicant has received a Notice of Allowance and Issue Fee Due, the ISSUE BATCH NUMBER and DATE of the NOTICE OF ALLOWANCE should also be included.

Attachment(s)
- ☒ Notice of References Cited, PTO-892
- ☐ Information Disclosure Statement(s), PTO-1449, Paper No(s). ____________
- ☒ Notice of Draftsperson's Patent Drawing Review, PTO-948
- ☐ Notice of Informal Patent Application, PTO-152
- ☒ Interview Summary, PTO-413
- ☒ Examiner's Amendment/Comment
- ☐ Examiner's Comment Regarding Requirement for Deposit of Biological Material
- ☒ Examiner's Statement of Reasons for Allowance

U. S. Patent and Trademark Office
PTO-37 (Rev. 9-95)

Notice of Allowability

Part of Paper No. 5

Notice of References Cited

Application No. 09/157,035	Applica	Joseph Marash et al.	
Examiner Jacques M. Saint-Surin		Group Art Unit 2747	Page 1 of 1

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*		DOCUMENT NO.	DATE	NAME	CLASS	SUBCLASS
X	A	5,825,898	10/98	Marash	381	92
X	B	5,627,799	5/97	Hoshuyama	367	121
X	C	4,965,834	10/90	Miller	381	94.1
X	D	5,226,016	7/93	Christman	367	135
	E				----	
	F					
	G					
	H					
	I					
	J					
	K					
	L					
	M					

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*		DOCUMENT NO.	DATE	COUNTRY	NAME	CLASS	SUBCLASS
	N						
	O						
	P						
	Q						
	R						
	S						
	T						

NON-PATENT DOCUMENTS

*		DOCUMENT (Including Author, Title, Source, and Pertinent Pages)	DATE
	U		
	V		
	W		
	X		

* A copy of this reference is not being furnished with this Office action.
(See Manual of Patent Examining Procedure, Section 707.05(a).)

Notice of References Cited

Part of Paper No. 5

US005825898A

United States Patent [19]

Marash

[11] **Patent Number: 5,825,898**

[45] **Date of Patent: Oct. 20, 1998**

[54] **SYSTEM AND METHOD FOR ADAPTIVE INTERFERENCE CANCELLING**

[75] Inventor: **Joseph Marash**, Haifa, Israel

[73] Assignee: **Lamar Signal Processing Ltd.**, Haifa, Israel

[21] Appl. No.: **672,899**

[22] Filed: **Jun. 27, 1996**

[51] **Int. Cl.**^{6} **H04R 3/00**

[52] **U.S. Cl.** **381/92**; 381/94.1; 381/91.2; 381/94.7; 367/121; 367/119

[58] **Field of Search** 381/92, 94, 155, 381/94.1, 94.2–94.4, 94.7, 26, 122, 111; 367/121–122, 123–126, 119, 118

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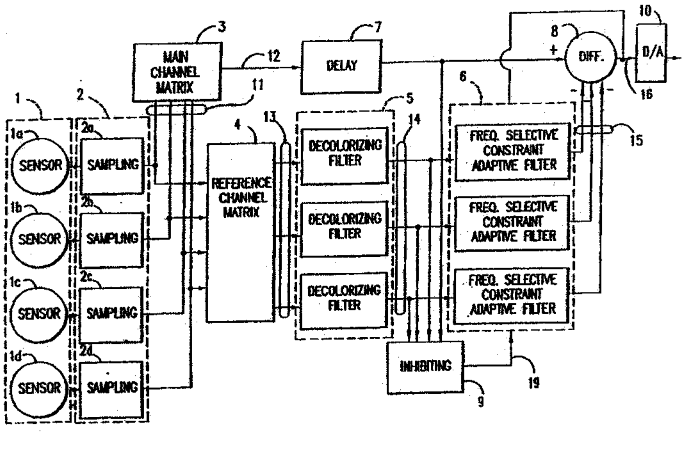
Primary Examiner—Curtis A. Kuntz
Assistant Examiner—Duc Nguyen
Attorney, Agent, or Firm—Frommer, Lawrence & Haug, LLP; Thomas J. Kowalski; I. Marc Asperas

[57] ABSTRACT

An adaptive system and method for reducing interference in a signal received from an array of sensors. Adaptive filters are used to generate cancelling signals that closely approximate the interference present in the received signal. The adaptive filter weights are converted into the frequency domain where the frequency representation values in a selected frequency range are truncated to avoid signal leakage involving narrow band signals. Decolorizing filters are used to produce the cancelling signals having a flat frequency spectrum. Normalized power difference is used limit the operation of the adaptive filters to the case where there is some directional interference to be eliminated.

28 Claims, 15 Drawing Sheets



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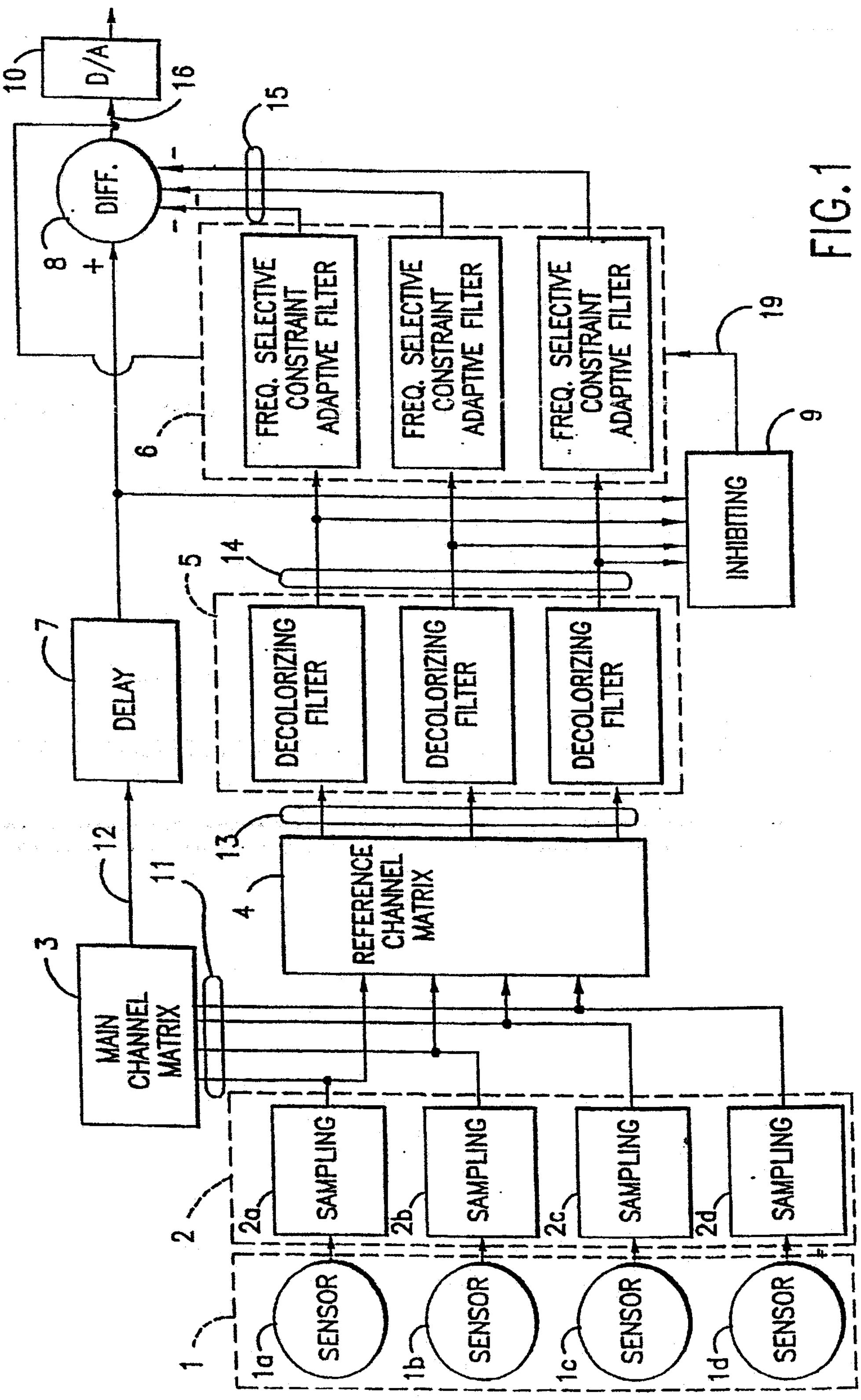

D/A
10
16
DIFF.
8
+
-
15
FREQ. SELECTIVE CONSTRAINT ADAPTIVE FILTER
FREQ. SELECTIVE CONSTRAINT ADAPTIVE FILTER
FREQ. SELECTIVE CONSTRAINT ADAPTIVE FILTER
6
19
9
INHIBITING
DELAY
7
5
14
DECOLORIZING FILTER
DECOLORIZING FILTER
DECOLORIZING FILTER
12
11
13
REFERENCE CHANNEL MATRIX
4
MAIN CHANNEL MATRIX
3
SAMPLING
SAMPLING
SAMPLING
SAMPLING
2
2a
2b
2c
2d
SENSOR
SENSOR
SENSOR
SENSOR
1
1a
1b
1c
1d

FIG. 1

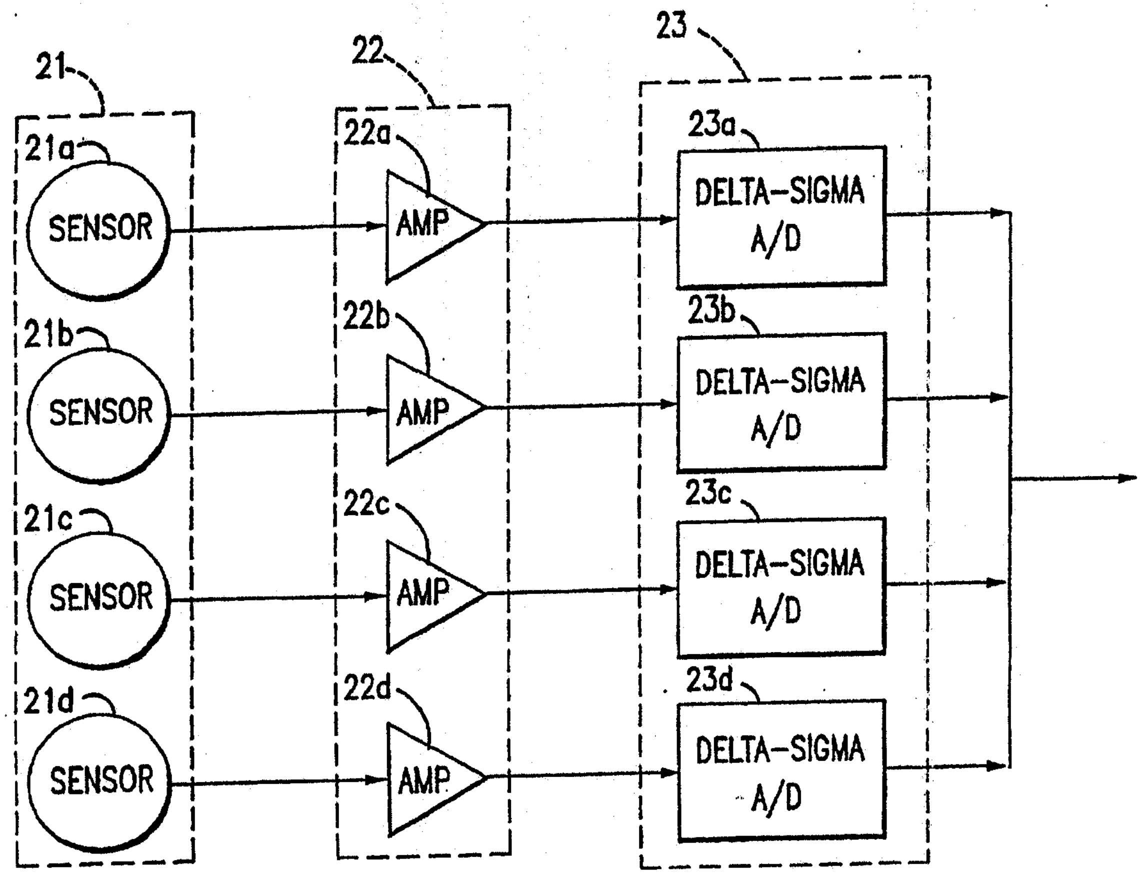

21
21a
SENSOR
21b
SENSOR
21c
SENSOR
21d
SENSOR
22
22a
AMP
22b
AMP
22c
AMP
22d
AMP
23
23a
DELTA-SIGMA A/D
23b
DELTA-SIGMA A/D
23c
DELTA-SIGMA A/D
23d
DELTA-SIGMA A/D

FIG.2

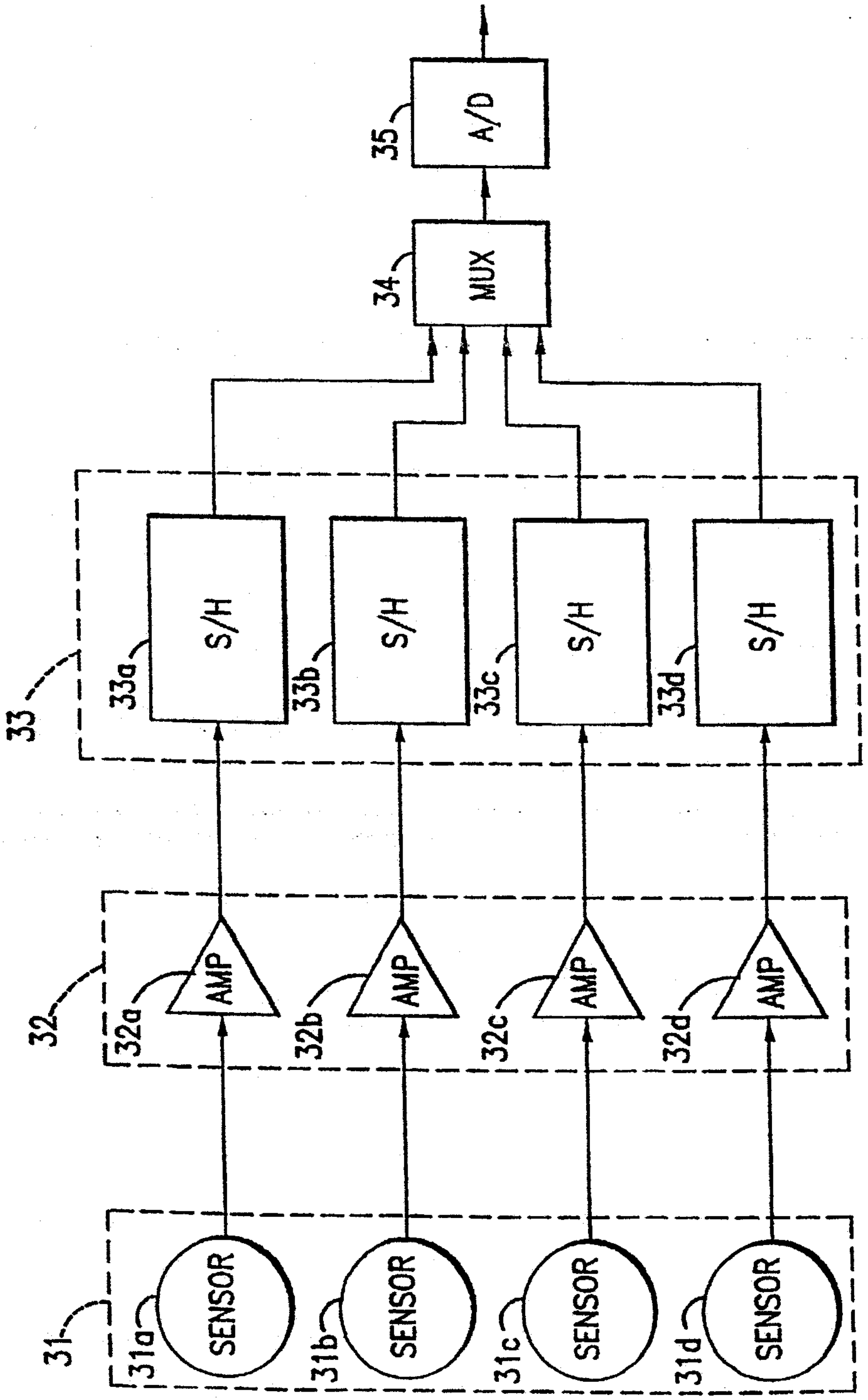

35
A/D
34
MUX
33
33a
S/H
33b
S/H
33c
S/H
33d
S/H
32
32a
AMP
32b
AMP
32c
AMP
32d
AMP
31
31a
SENSOR
31b
SENSOR
31c
SENSOR
31d
SENSOR

FIG.3

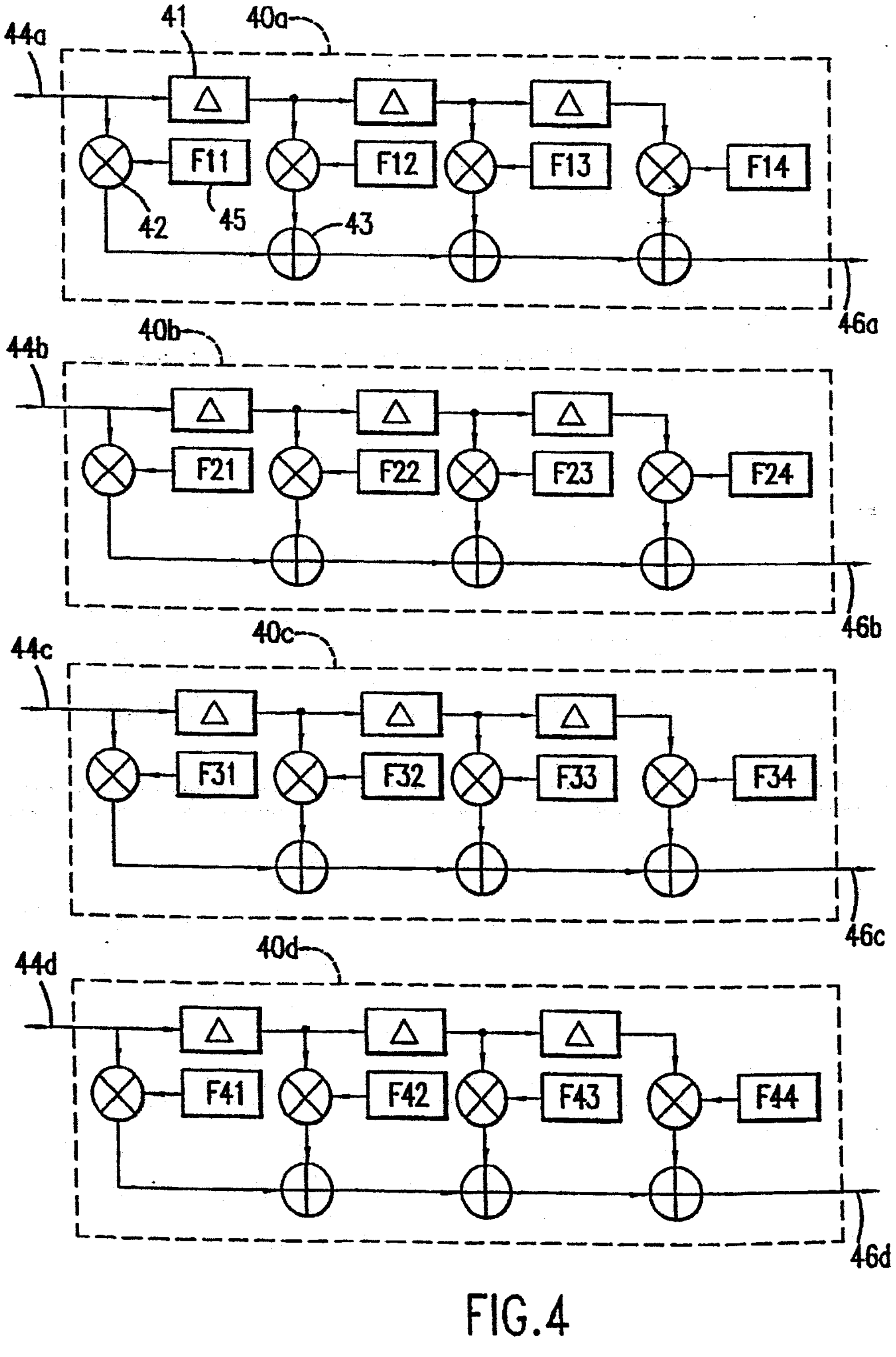

44a
41
40a
Δ
F11
F12
F13
F14
42
45
43
46a
44b
40b
F21
F22
F23
F24
46b
44c
40c
F31
F32
F33
F34
46c
44d
40d
F41
F42
F43
F44
46d

FIG.4

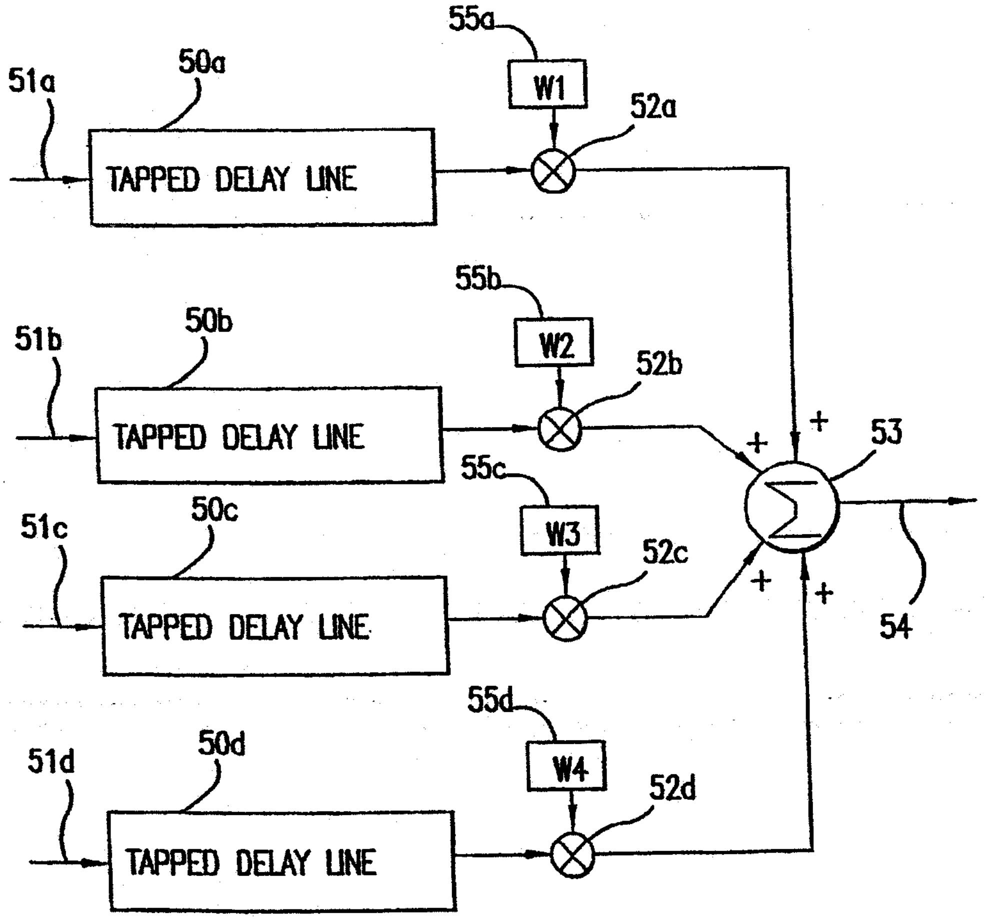

55a
W1
50a
51a
52a
TAPPED DELAY LINE
55b
W2
50b
51b
52b
TAPPED DELAY LINE
+
+
53
55c
W3
50c
51c
52c
TAPPED DELAY LINE
+
+
54
55d
W4
50d
51d
52d
TAPPED DELAY LINE

FIG.5

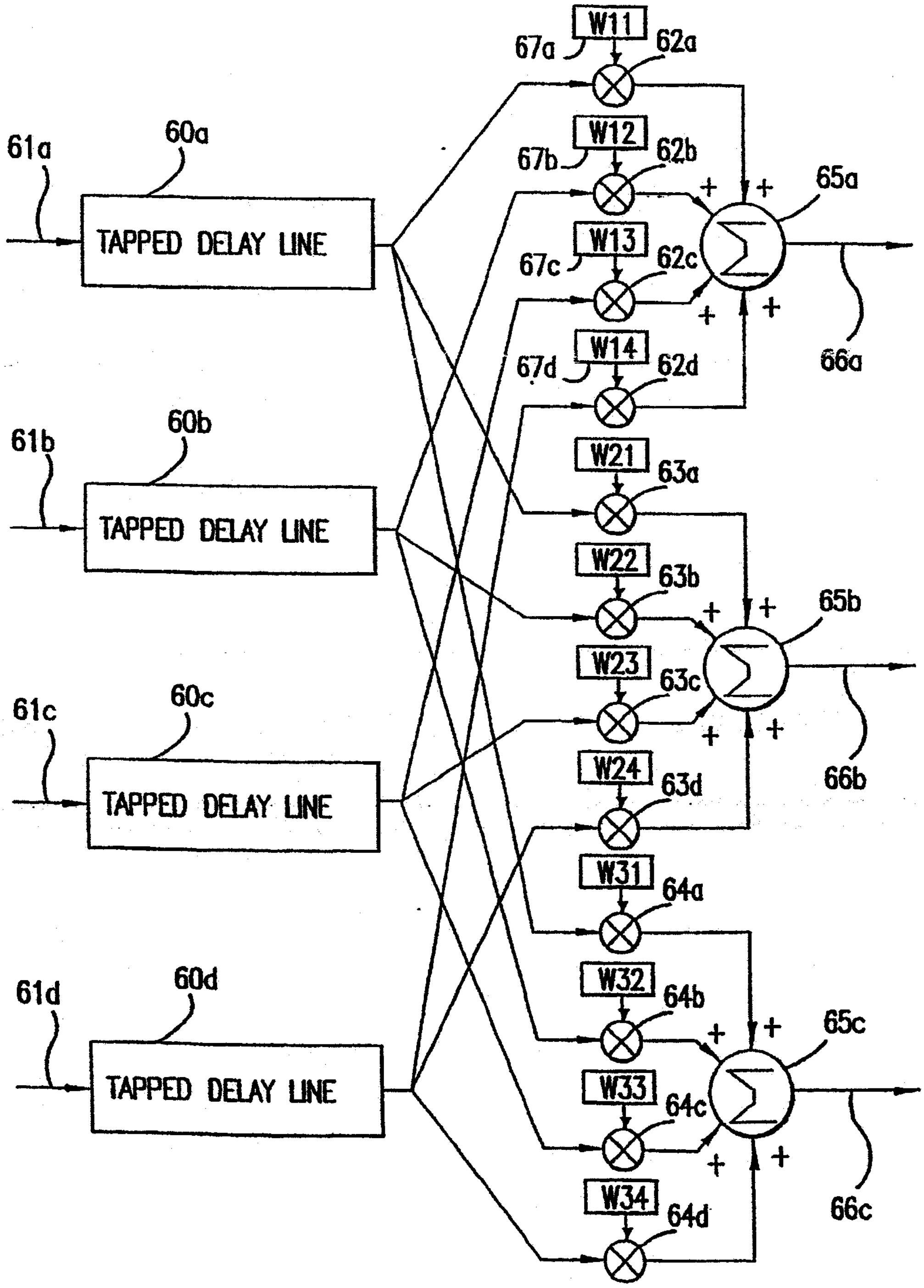

W11
67a
62a
61a
60a
W12
67b
62b
65a
TAPPED DELAY LINE
W13
67c
62c
66a
W14
67d
62d
W21
63a
61b
60b
TAPPED DELAY LINE
W22
63b
65b
W23
63c
61c
60c
TAPPED DELAY LINE
W24
63d
66b
W31
64a
61d
60d
W32
64b
65c
TAPPED DELAY LINE
W33
64c
W34
64d
66c
+

FIG.6

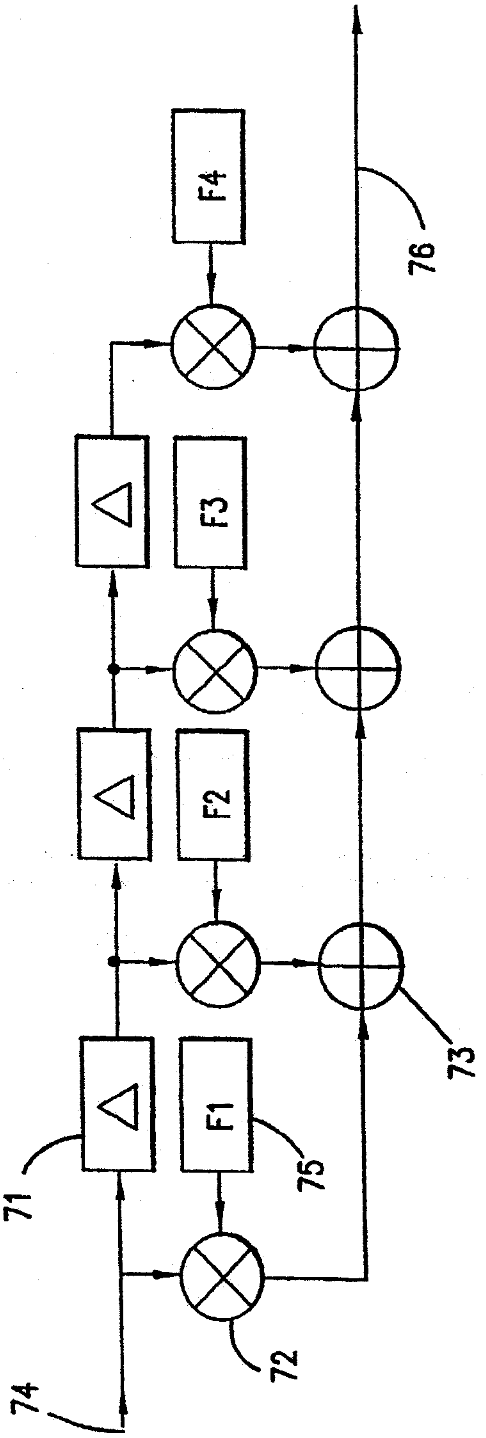


FIG.7

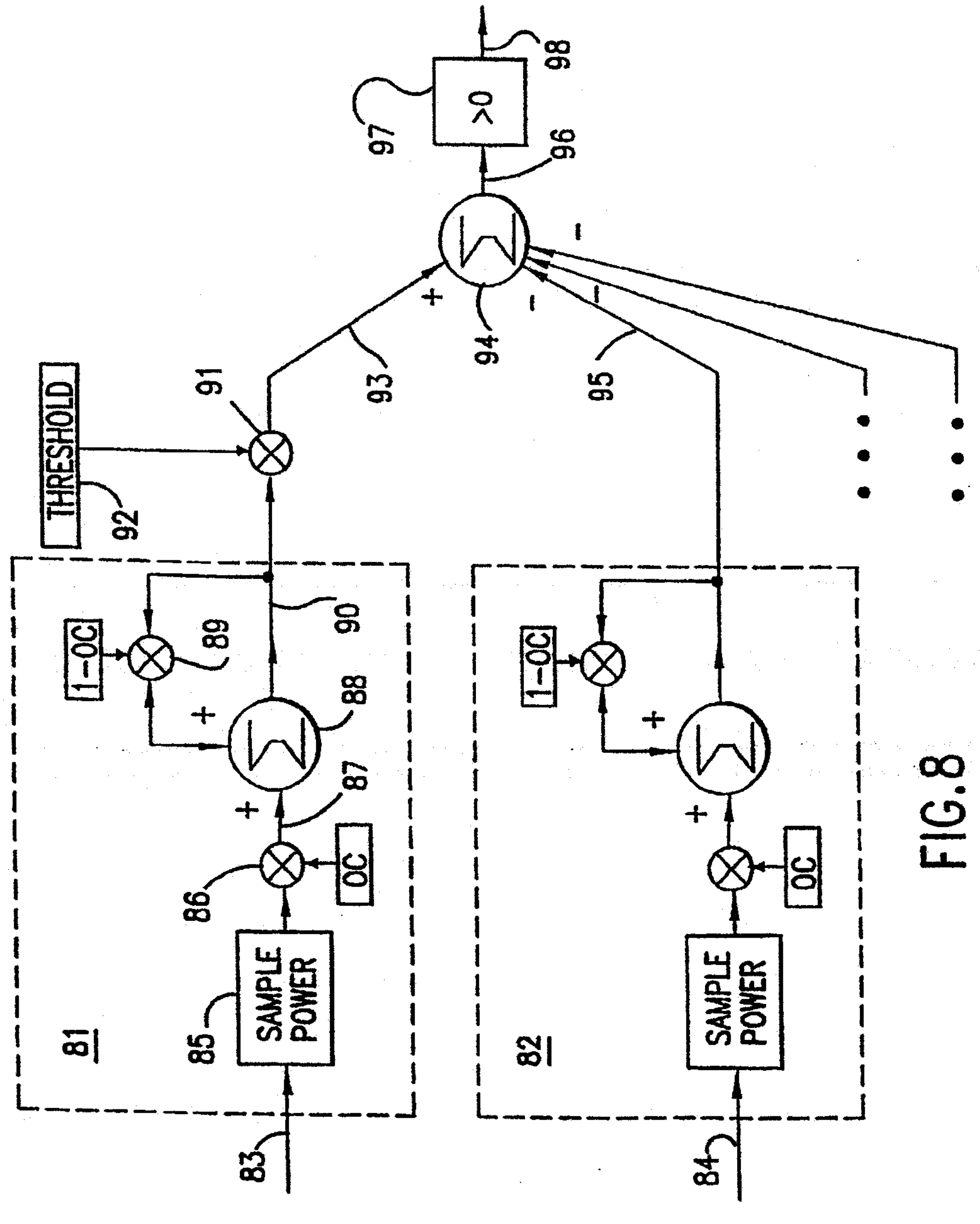

83
81
85
SAMPLE POWER
86
OC
87
+
88
+
1-OC
89
90
THRESHOLD
92
91
93
+
94
-
-
95
-
96
>0
97
98
82
84
SAMPLE POWER
OC
+
+
1-OC

FIG.8

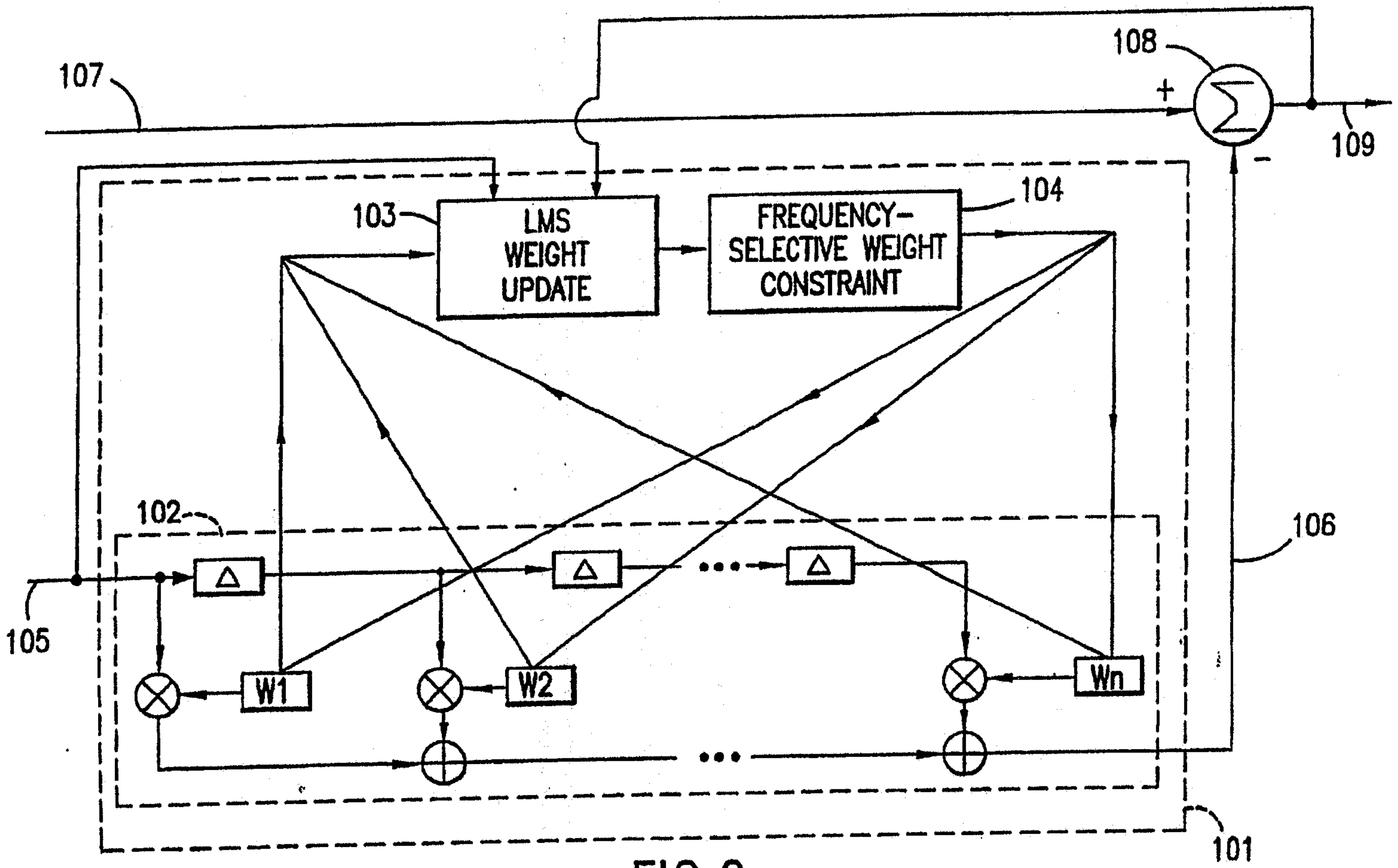

107
108
+
-
109
103
LMS WEIGHT UPDATE
FREQUENCY-SELECTIVE WEIGHT CONSTRAINT
104
102
105
Δ
Δ
Δ
W1
W2
Wn
106
101

FIG.9

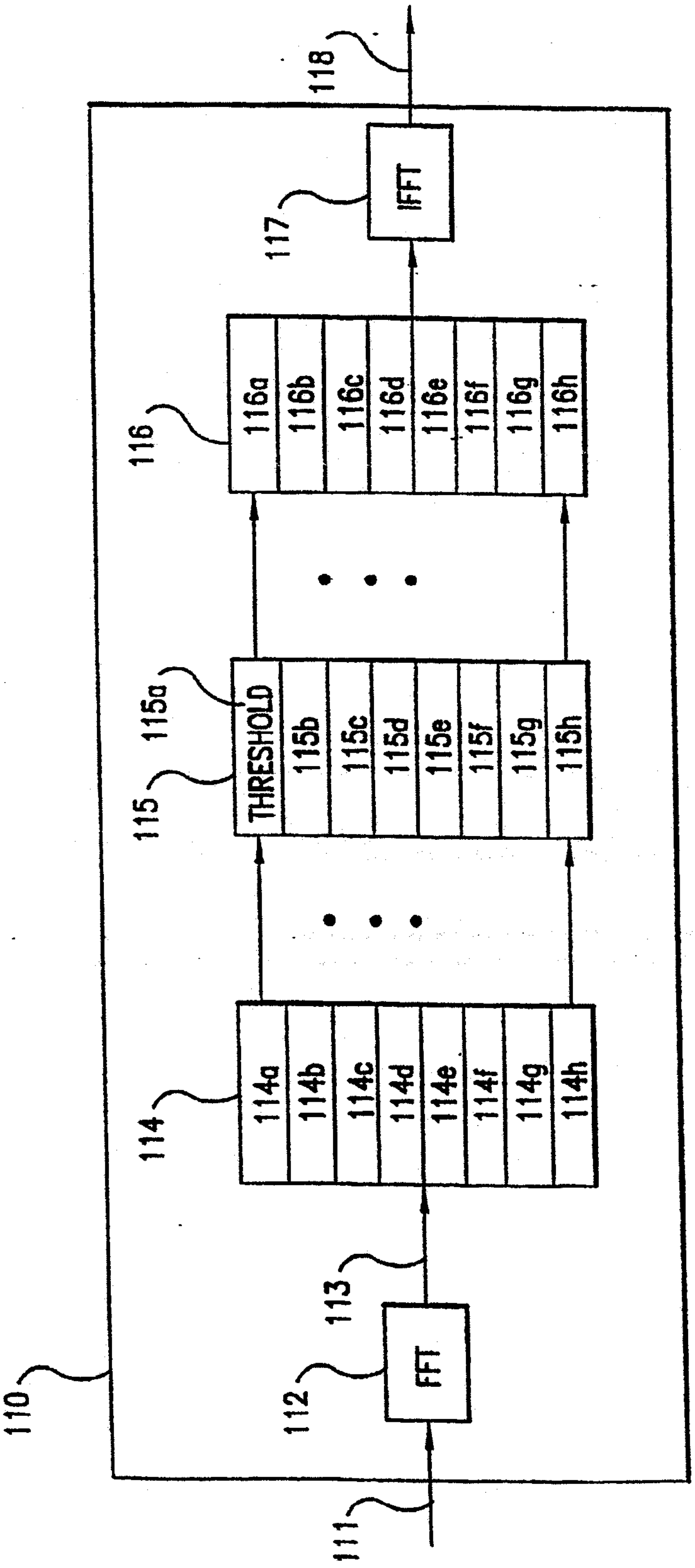

110
111
112
FFT
113
114
114a
114b
114c
114d
114e
114f
114g
114h
115
115a
THRESHOLD
115b
115c
115d
115e
115f
115g
115h
116
116a
116b
116c
116d
116e
116f
116g
116h
117
IFFT
118

FIG. 10

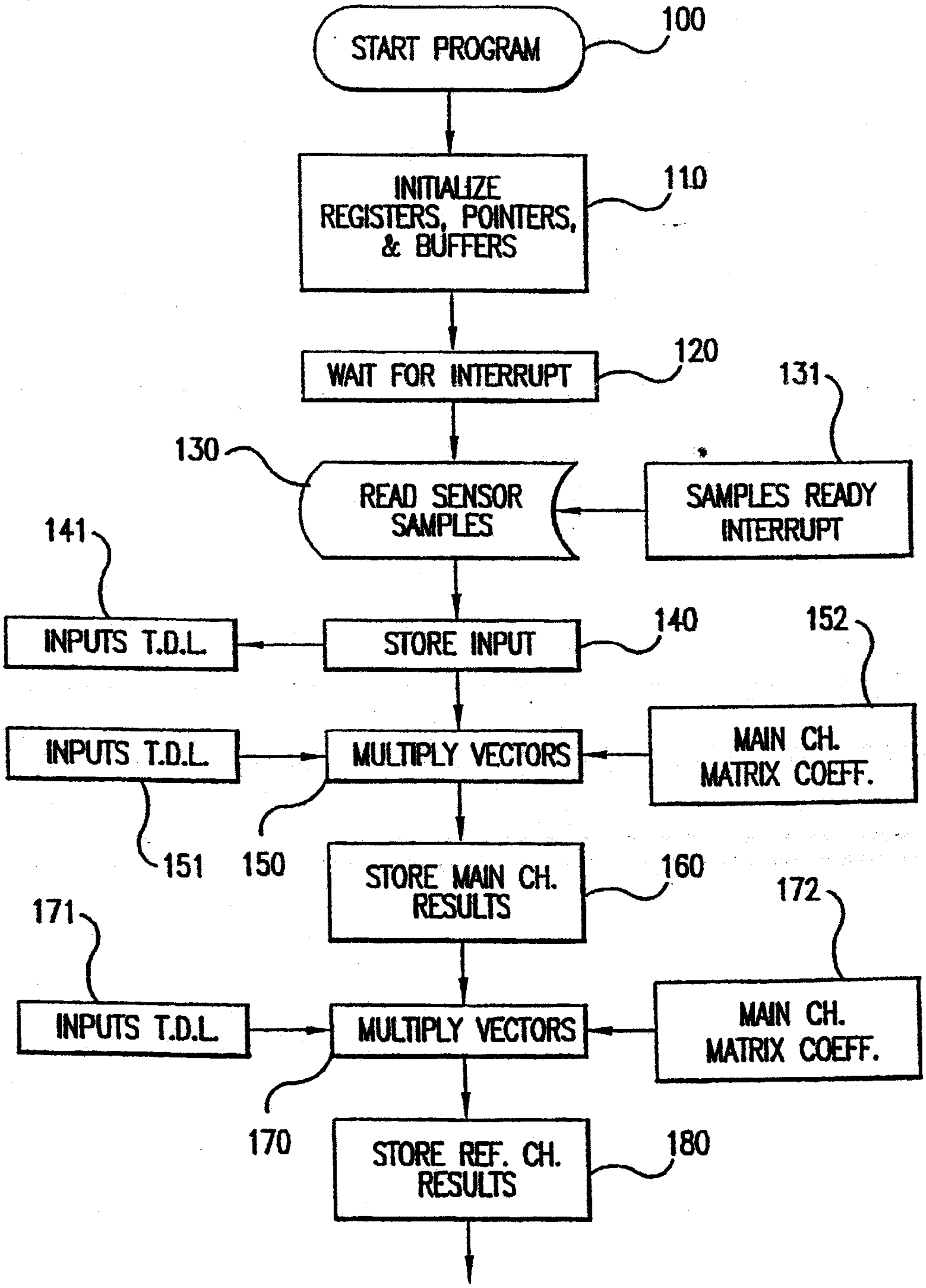
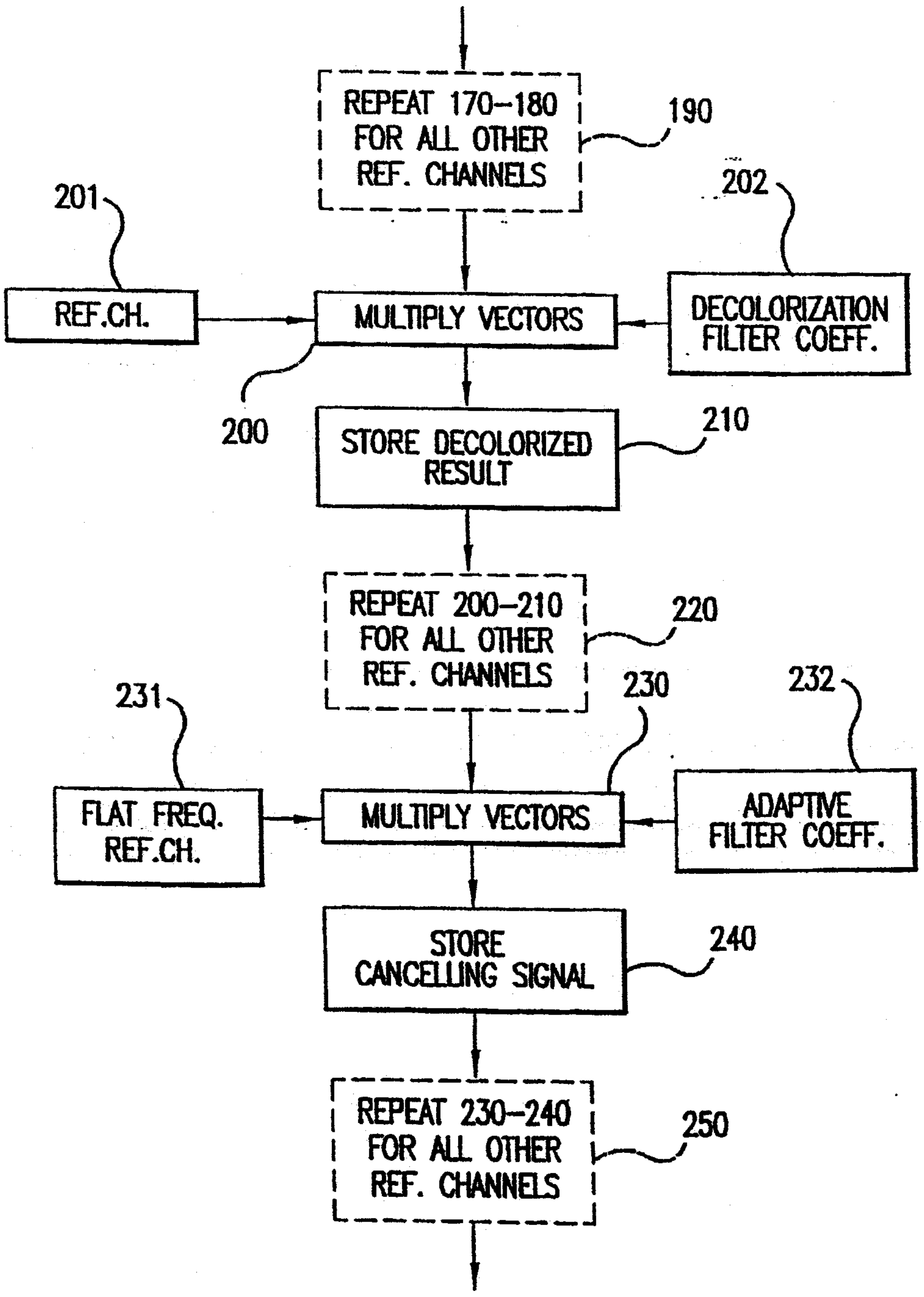
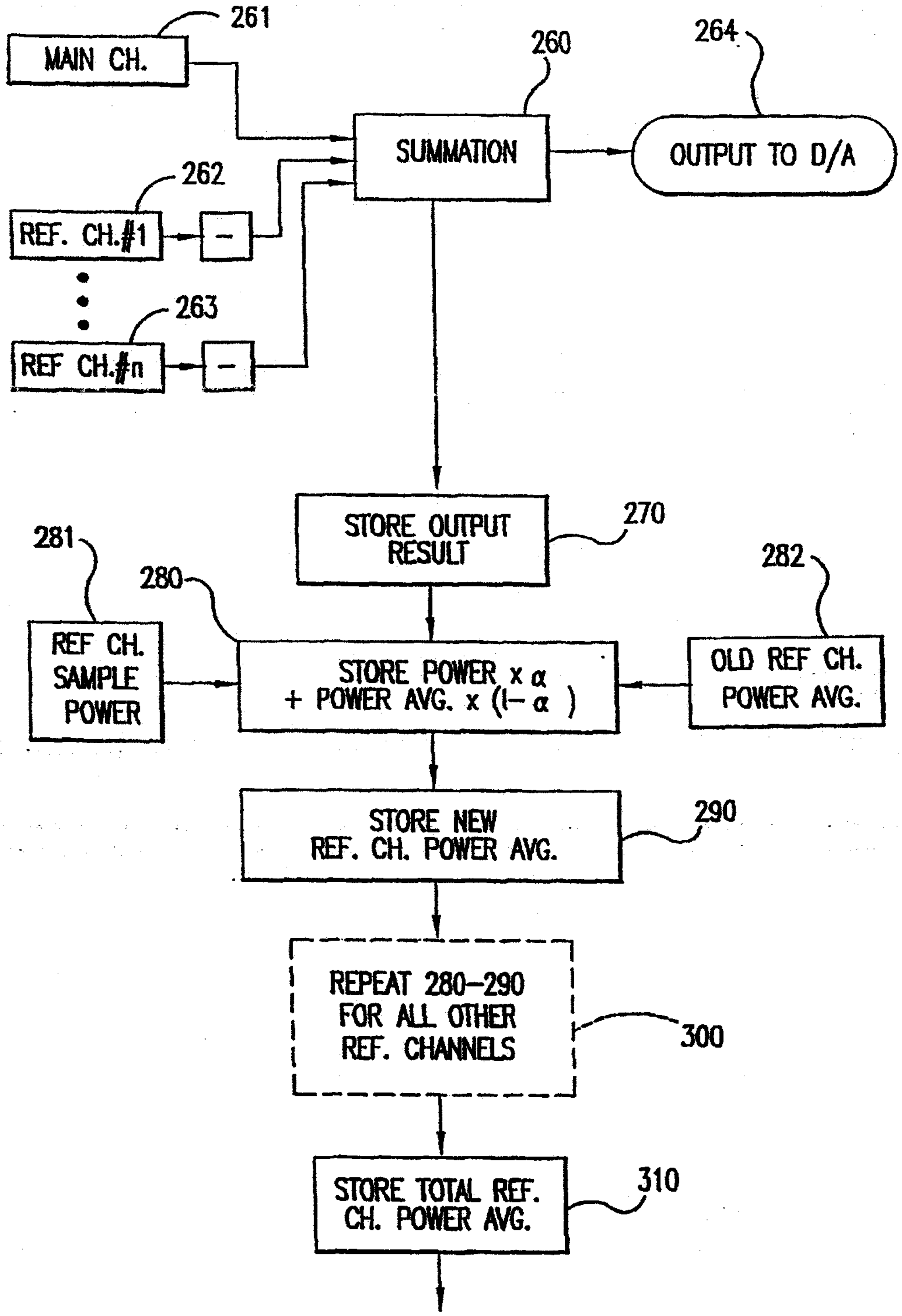

START PROGRAM
100
INITIALIZE REGISTERS, POINTERS, & BUFFERS
110
WAIT FOR INTERRUPT
120
131
130
READ SENSOR SAMPLES
SAMPLES READY INTERRUPT
141
INPUTS T.D.L.
STORE INPUT
140
152
INPUTS T.D.L.
MULTIPLY VECTORS
MAIN CH. MATRIX COEFF.
151
150
STORE MAIN CH. RESULTS
160
172
171
INPUTS T.D.L.
MULTIPLY VECTORS
MAIN CH. MATRIX COEFF.
170
STORE REF. CH. RESULTS
180

FIG.11A



REPEAT 170-180 FOR ALL OTHER REF. CHANNELS
190
201
202
REF.CH.
MULTIPLY VECTORS
DECOLORIZATION FILTER COEFF.
200
STORE DECOLORIZED RESULT
210
REPEAT 200-210 FOR ALL OTHER REF. CHANNELS
220
231
230
232
FLAT FREQ. REF.CH.
MULTIPLY VECTORS
ADAPTIVE FILTER COEFF.
STORE CANCELLING SIGNAL
240
REPEAT 230-240 FOR ALL OTHER REF. CHANNELS
250

FIG.11B



261
MAIN CH.
260
264
SUMMATION
OUTPUT TO D/A
262
REF. CH.#1
–
263
REF CH.#n
–
STORE OUTPUT RESULT
270
281
280
282
REF CH. SAMPLE POWER
STORE POWER x α + POWER AVG. x (1– α)
OLD REF CH. POWER AVG.
STORE NEW REF. CH. POWER AVG.
290
REPEAT 280–290 FOR ALL OTHER REF. CHANNELS
300
STORE TOTAL REF. CH. POWER AVG.
310

FIG.11C

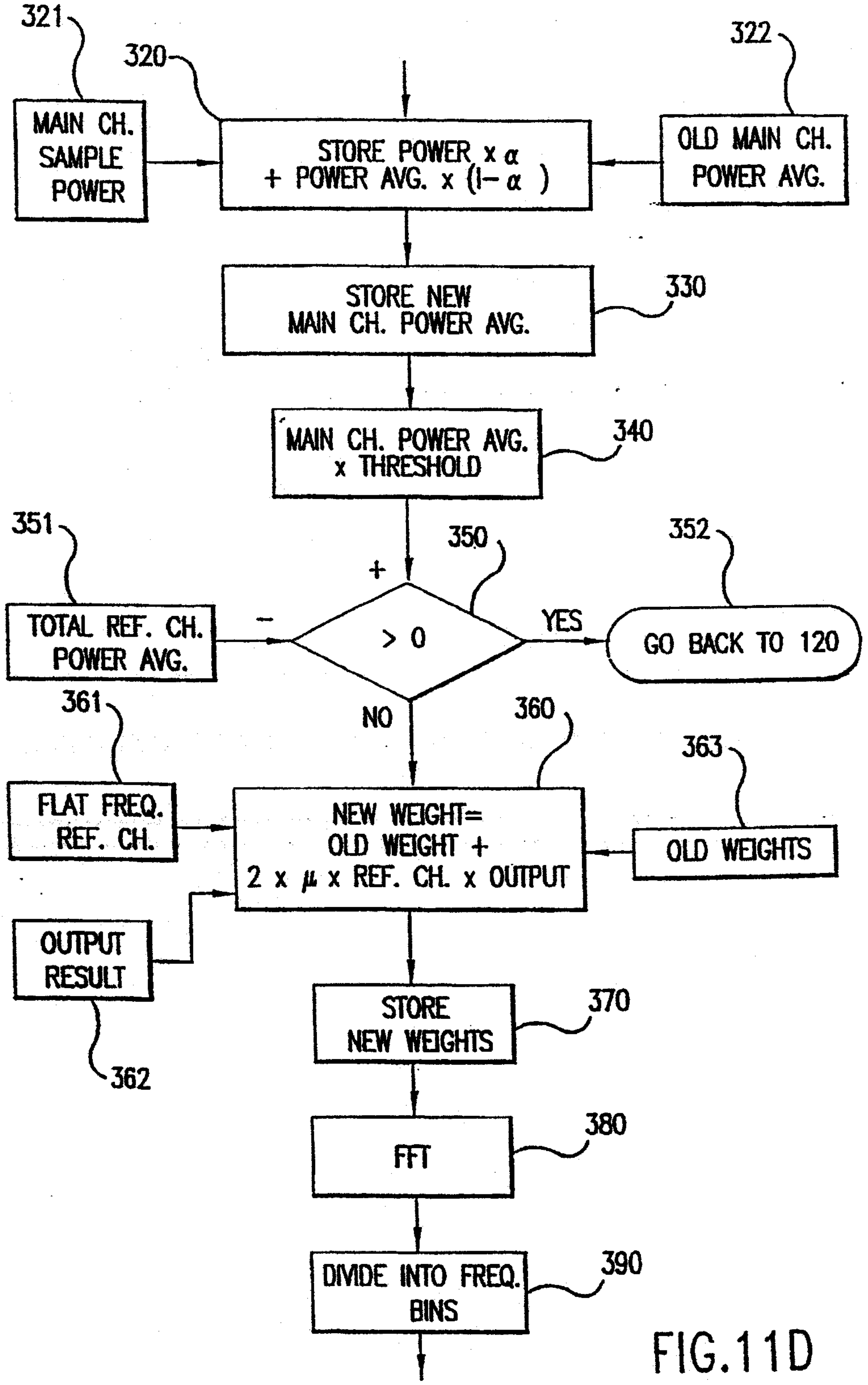

321
320
322
MAIN CH. SAMPLE POWER
STORE POWER x α + POWER AVG. x (1− α)
OLD MAIN CH. POWER AVG.
STORE NEW MAIN CH. POWER AVG.
330
MAIN CH. POWER AVG. x THRESHOLD
340
351
350
352
+
−
TOTAL REF. CH. POWER AVG.
> 0
YES
GO BACK TO 120
NO
361
360
363
FLAT FREQ. REF. CH.
NEW WEIGHT= OLD WEIGHT + 2 x μ x REF. CH. x OUTPUT
OLD WEIGHTS
OUTPUT RESULT
362
STORE NEW WEIGHTS
370
FFT
380
DIVIDE INTO FREQ. BINS
390

FIG.11D

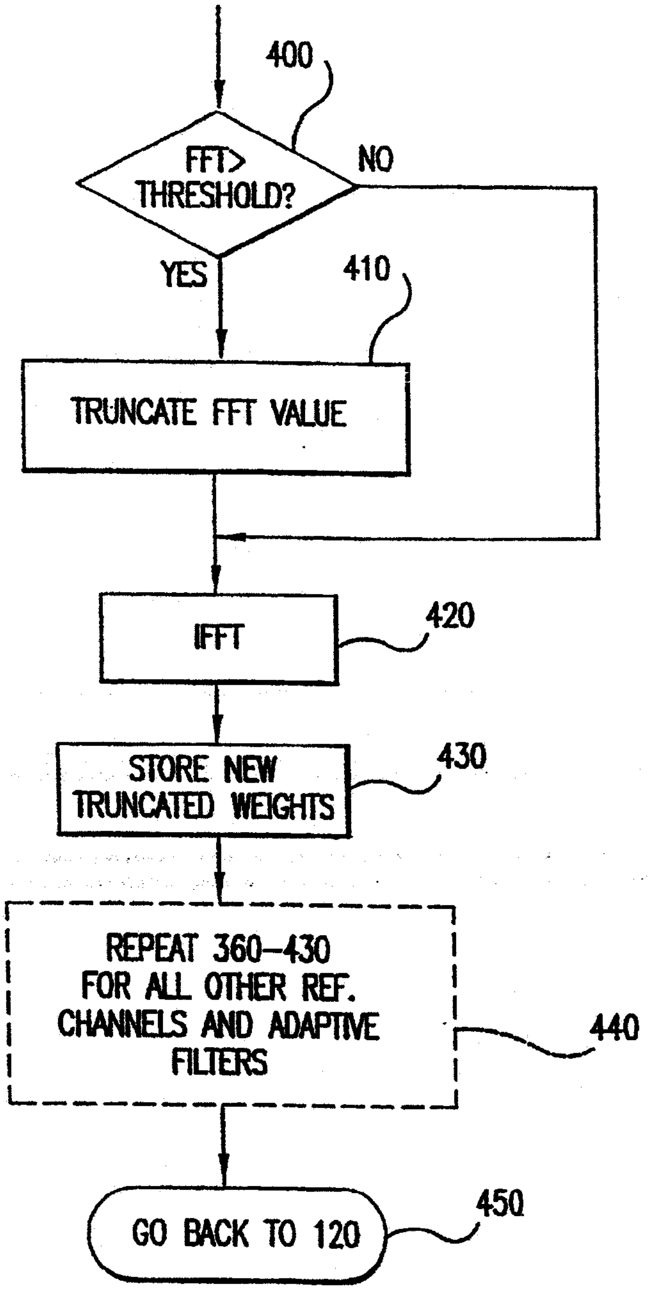

400
FFT> THRESHOLD?
NO
YES
410
TRUNCATE FFT VALUE
IFFT
420
STORE NEW TRUNCATED WEIGHTS
430
REPEAT 360-430 FOR ALL OTHER REF. CHANNELS AND ADAPTIVE FILTERS
440
GO BACK TO 120
450

FIG.11E

SYSTEM AND METHOD FOR ADAPTIVE INTERFERENCE CANCELLING

BACKGROUND OF THE INVENTION

The present invention relates generally to signal processing, and more specifically to an adaptive signal processing system and method for reducing interference in a received signal.

There are many instances where it is desirable to have a sensor capable of receiving an information signal from a particular signal source where the environment includes sources of interference signals at locations different from that of the signal source. One such instance is the use of microphones to record a particular party's speech in a room where there are other parties speaking simultaneously, causing interference in the received signals.

If one knows the exact characteristics of the interference, one can use a fixed-weight filter to suppress it. But it is often difficult to predict the exact characteristics of the interference because they may vary according to changes in the interference sources, the background noise, acoustic environment, orientation of the sensor with respect to the signal source, the transmission paths from the signal source to the sensor, and many other factors. Therefore, in order to suppress such interference, an adaptive system that can change its own parameters in response to a changing environment is needed.

An adaptive filter is an adaptive system that can change its own filtering characteristics in order to produce a desired response. Typically, the filter weights defining the characteristics of an adaptive filter are continuously updated so that the difference between a signal representing a desired response and an output signal of the adaptive filter is minimized.

The use of adaptive filters for reducing interference in a received signal has been known in the art as adaptive noise cancelling. It is based on the idea of cancelling a noise component of a received signal from the direction of a signal source by sampling the noise independently of the source signal and modifying the sampled noise to approximate the noise component in the received signal using an adaptive filter. For a seminal article on adaptive noise cancelling, see B. Widrow et al., Adaptive Noise Cancelling: Principles and Applications, Proc. IEEE 63:1692–1716, 1975.

A basic configuration for adaptive noise cancelling has a primary input received by a microphone directed to a desired signal source and a reference input received independently by another microphone directed to a noise source. The primary input contains both a source signal component originating from the signal source and a noise component originating from the noise source. The noise component is different from the reference input representing the noise source itself because the noise signal must travel from the noise source to the signal source in order to be included as the noise component.

The noise component, however, is likely to have some correlation with the reference input because both of them originate from the same noise source. Thus, a filter can be used to filter the reference input to generate a cancelling signal approximating the noise component. The adaptive filter does this dynamically by generating an output signal which is the difference between the primary input and the cancelling signal, and by adjusting its filter weights to minimize the mean-square value of the output signal. When the filter weights settle, the output signal effectively replicates the source signal substantially free of the noise component because the cancelling signal closely tracks the noise component.

Adaptive noise cancelling can be combined with beamforming, a known technique of using an array of sensors to improve reception of signals coming from a specific direction. A beamformer is a spatial filter that generates a single channel from multiple channels received through multiple sensors by filtering the individual multiple channels and combining them in such a way as to extract signals coming from a specific direction. Thus, a beamformer can change the direction of receiving sensitivity without physically moving the array of sensors. For details on beamforming, see B. D. Van Veen and K. M. Buckley, Beamforming:

A Versatile Approach to Spatial Filtering, IEEE ASSP Mag. 5(2), 4–24.

Since the beamformer can effectively be pointed in many directions without physically moving its sensors, the beamformer can be combined with adaptive noise cancelling to form an adaptive beamformer that can suppress specific directional interference rather than general background noise. The beamformer can provide the primary input by spatially filtering input signals from an array of sensors so that its output represents a signal received in the direction of a signal source. Similarly, the beamformer can provide the reference input by spatially filtering the sensor signals so that the output represents a signal received in the direction of interference sources. For a seminal article on adaptive beamformers, see L. J. Griffiths & C. W. Jim, An Alternative Approach to Linearly Constrained Adaptive Beamforming, IEEE Trans. Ant. Prop. AP-30:27–34, 1982.

One problem with a conventional adaptive beamformer is that its output characteristics change depending on input frequencies and sensor directions with respect to interference sources. This is due to the sensitivity of a beamformer to different input frequencies and sensor directions. A uniform output behavior of a system over all input frequencies of interest and over all sensor directions is clearly desirable in a directional microphone system where faithful reproduction of a sound signal is required regardless of where the microphones are located.

Another problem with adaptive beamforming is "signal leakage". Adaptive noise cancelling is based on an assumption that the reference input representing noise sources is uncorrelated with the source signal component in the primary input, meaning that the reference input should not contain the source signal. But this "signal free" reference input assumption is violated in any real environment. Any mismatch in the microphones (amplitude or phase) or their related analog front end, any reverberation caused by the surroundings or a mechanical structure, and even any mechanical coupling in the physical microphone structure will likely cause "signal leakage" from the signal source into the reference input. If there is any correlation between the reference input and the source signal component in the primary input, the adaptation process by the adaptive filter causes cancellation of the source signal component, resulting in distortion and degradation in performance.

It is also important to confine the adaptation process to the case where there is at least some directional interference to be eliminated. Since nondirectional noise, such as wind noise or vibration noise induced by the mechanical structure of the system, is typically uncorrelated with the noise component of the received signal, the adaptive filter cannot generate a cancelling signal approximating the noise component.

Prior art suggests inhibiting the adaptation process of an adaptive filter when the signal-to-noise ratio (SNR) is high based on the observation that a strong source signal tends to leak into the reference input. For example, U.S. Pat. No. 4,956,867 describes the use of cross-correlation between two sensors to inhibit the adaptation process when the SNR is high.

But the prior art approach fails to consider the effect of directional interference because the SNR-based approach considers only nondirectional noise. Since nondirectional noise is not correlated to the noise component of the received signal, the adaptation process searches in vain for new filter weights, which often results in cancelling the source signal component of the received signal.

The prior art approach also fails to consider signal leakage when the source signal is of a narrow bandwidth. In a directional microphone application, the source signal often contains a narrow band signal, such as speech signal, with its power spectral density concentrated in a narrow frequency range. When signal leakage occurs due to a strong narrow band signal, the prior art approach may not inhibit the adaptation process because the overall signal strength of such narrow band signal may not high enough. The source signal component of the received signal is cancelled as a result, and if the source signal is a speech signal, degradation in speech intelligibility occurs.

Therefore, there exists a need for an adaptive system that can suppress directional interference in a received signal with a uniform frequency behavior over a wide angular distribution of interference sources.

SUMMARY OF THE INVENTION

Accordingly, it is an object of the present invention to suppress interference in a received signal using an adaptive filter for processing inputs from an array of sensors.

Another object of the invention is to limit the adaptation process of such adaptive filter to the case where there is at least some directional interference to be eliminated.

A further object of the invention is to control the adaptation process to prevent signal leakage for narrow band signals.

Another object is to produce an output with a uniform frequency behavior in all directions from the sensor array.

These and other objects are achieved in accordance with the present invention, which uses a system for processing digital data representing signals received from an array of sensors. The system includes a main channel matrix unit for generating a main channel representing signals received in the direction of a signal source where the main channel has a source signal component and an interference signal component. The system includes a reference channel matrix unit for generating at least one reference channel where each reference channel represents signals received in directions other than that of the signal source. The system uses adaptive filters for generating cancelling signals approximating the interference signal component of the main channel and a difference unit for generating a digital output signal by subtracting the cancelling signals from the main channel. Each adaptive filter has weight updating means for finding new filter weights based on the output signal. The system includes weight constraining means for truncating the new filter weight values to predetermined threshold values when each of the new filter weight value exceeds the corresponding threshold value.

The system may further include at least one decolorizing filter for generating a flat-frequency reference channel. The system may further include inhibiting means for estimating the power of the main channel and the power of the reference channels and for generating an inhibit signal to the weight updating means based on normalized power difference between the main channel and the reference channels.

The system produces an output substantially free of directional interference with a uniform frequency behavior in all directions from the system.

The objects are also achieved in accordance with the present invention using a method, which can readily be implemented in a program controlling a commercially available DSP processor.

BRIEF DESCRIPTION OF THE DRAWINGS

The objects, features and advantages of the present invention will be more readily apparent from the following detailed description of the invention in which:

FIG. 1 is a block diagram of an overall system;

FIG. 2 is a block diagram of a sampling unit;

FIG. 3 is a block diagram of an alternative embodiment of a sampling unit;

FIG. 4 is a schematic depiction of tapped delay lines used in a main channel matrix and a reference matrix unit;

FIG. 5 is a schematic depiction of a main channel matrix unit;

FIG. 6 is a schematic depiction of a reference channel matrix unit;

FIG. 7 is a schematic depiction of a decolorizing filter;

FIG. 8 is a schematic depiction of an inhibiting unit based on directional interference;

FIG. 9 is a schematic depiction of a frequency-selective constraint adaptive filter;

FIG. 10 is a block diagram of a frequency-selective weight-constraint unit;

FIG. 11 is a flow chart depicting the operation of a program that can be used to implement the invention.

DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 is a block diagram of a system in accordance with a preferred embodiment of the present invention. The system illustrated has a sensor array **1**, a sampling unit **2**, a main channel matrix unit **3**, a reference channel matrix unit **4**, a set of decolorizing filters **5**, a set of frequency-selective constrained adaptive filters **6**, a delay **7**, a difference unit **8**, an inhibiting unit **9**, and an output D/A unit **10**.

Sensor array **1**, having individual sensors **1***a*–**1***d*, receives signals from a signal source on-axis from the system and from interference sources located off-axis from the system. The sensor array is connected to sampling unit **2** for sampling the received signals, having individual sampling elements, **2***a*–**2***d*, where each element is connected to the corresponding individual sensor to produce digital signals **11**.

The outputs of sampling unit **2** are connected to main channel matrix unit **3** producing a main channel **12** representing signals received in the direction of a source. The main channel contains both a source signal component and an interference signal component.

The outputs of sampling unit **2** are also connected reference channel matrix unit **4**, which generates reference channels **13** representing signals received from directions other that of the signal source. Thus, the reference channels represent interference signals.

The reference channels are filtered through decolorizing filters **5**, which generate flat-frequency reference channels **14** having a frequency spectrum whose magnitude is substantially flat over a frequency range of interest. Flat-frequency reference channels **14** are fed into the set of frequency-selective constraint adaptive filters **6**, which generate cancelling signals **15**.

In the mean time, main channel **12** is delayed through delay **7** so that it is synchronized with cancelling signals **15**. Difference unit **8** then subtracts cancelling signals **15** from the delayed main channel to generate an digital output signal **16**, which is converted by D/A unit **10** into analog form. Digital output signal **15** is fed back to the adaptive filters to update the filter weights of the adaptive filters.

Flat-frequency reference channels **14** are fed to inhibiting unit **9**, which estimates the power of each flat-frequency reference channel as well as the power of the main channel and generates an inhibit signal **19** to prevent signal leakage.

FIG. **2** depicts a preferred embodiment of the sampling unit. A sensor array **21**, having sensor elements **21*a*–21*d***, is connected to an analog front end **22**, having amplifier elements **22*a*–22*d***, where each amplifier element is connected to the output of the corresponding sensor element. In a directional microphone application, each sensor can be either a directional or omnidirectional microphone. The analog front end amplifies the received analog sensor signals to match the input requirement of the sampling elements. The outputs from the analog front ends are connected to a set of delta-sigma A/D converters, **23**, where each converter samples and digitizes the amplified analog signals. The delta-sigma sampling is a well-known A/D technique using both oversampling and digital filtering. For details on delta-sigma A/D sampling, see Crystal Semiconductor Corporation, Application Note: Delta-Sigma Techniques, 1989.

FIG. **3** shows an alternative embodiment of the sampling unit. A sensor array **31**, having sensor elements **31*a*–31*d***, is connected to an amplifier **32**, having amplifier elements **32*a*–32*d***, where each amplifier element amplifies the received signals from the corresponding sensor element. The outputs of the amplifier are connected to a sample & hold (S/H) unit **33** having sample & hold elements **33*a*–33*d***, where each S/H element samples the amplified analog signal from the corresponding amplifier element to produce a discrete signal. The outputs from the S/H unit are multiplexed into a single signal through a multiplexor **34**. The output of the multiplexor is connected to a conventional A/D converter **35** to produce a digital signal.

FIG. **4** is a schematic depiction of tapped delay lines used in the main channel matrix unit and the reference channel matrix in accordance with a preferred embodiment of the present invention. The tapped delay line used here is defined as a nonrecursive digital filter, also known in the art as a transversal filter, a finite impulse response filter or an FIR filter. The illustrated embodiment has 4 tapped delay lines, **40*a*–40*d***. Each tapped delay line includes delay elements **41**, multipliers **42** and adders **43**. Digital signals, **44*a*–44*d***, are fed into the set of tapped delay lines **40*a*–40*d***. Delayed signals through delay elements **41** are multiplied by filter coefficients, F_{ij}, **45** and added to produce outputs, **46*a*–46*d***.

The n-th sample of an output from the i-th tapped delay line, $Y_i(n)$, can then be expressed as:

$$Y_i(n)=\Sigma^k{}_{j=0} F_{i,j} X_i(n-j),$$

where k is the length of the filter, and $X_i(n)$ is the n-th sample of an input to the i-th tapped delay line.

FIG. **5** depicts the main channel matrix unit for generating a main channel in accordance with a preferred embodiment of the present invention. The unit has tapped delay lines, **50*a*–50*d***, as an input section taking inputs **51*a*–51*d*** from the sampling unit. Its output section includes multipliers, **52*a*–52*d***, where each multiplier is connected to the corresponding tapped delay line and an adder **53**, which sums all output signals from the multipliers. The unit generates a main channel **54**, as a weighted sum of outputs from all multipliers. The filter weights **55*a*–55*d*** can be any combination of fractions as long as their sum is **1**. For example, if 4 microphones are used, the embodiment may use the filter weights of ¼ in order to take into account of the contribution of each microphone.

The unit acts as a beamformer, a spatial filter which filters a signal coming in all directions to produce a signal coming in a specific direction without physically moving the sensor array. The coefficients of the tapped delay lines and the filter weights are set in such a way that the received signals are spatially filtered to maximize the sensitivity toward the signal source.

Since some interference signals find their way to reach the signal source due to many factors such as the reverberation of a room, main channel **54** representing the received signal in the direction of the signal source contains not only a source signal component, but also an interference signal component.

FIG. **6** depicts the reference channel matrix unit for generating reference matrix channels in accordance with a preferred embodiment of the present invention. It has tapped delay lines, **60*a*–60*d***, as an input section taking inputs **61*a*–61*d*** from the sampling unit. The same tapped delay lines as that of FIG. **4** may be used, in which case the tapped delay lines may be shared by the main and reference channel matrix units.

Its output section includes multipliers, **62*a*–62*d*, 63*a*–63*d*, 64*a*–64*d*** and adders **65*a*–65*c***, where each multiplier is connected to the corresponding tapped delay line and adder. The unit acts as a beamformer which generates the reference channels **66*a*–66*c*** representing signals arriving off-axis from the signal source by obtaining the weighted differences of certain combinations of outputs from the tapped delay lines. The filter weight combinations can be any numbers as long as their sum of filter weights for combining a given reference channel is 0. For example, the illustrated embodiment may use a filter weight combination, **(W11, W12, W13, W14) =**(0.25, 0.25, 0.25, –0.75), in order to combine signals **61*a*–61*d*** to produce reference channel **66*a***.

The net effect is placing a null (low sensitivity) in the receiving gain of the beamformer toward the signal source. As a result, the reference channels represent interference signals in directions other than that of the signal source. In other words, the unit "steers" the input digital data to obtain interference signals without physically moving the sensor array.

FIG. **7** is a schematic depiction of the decolorizing filter in accordance with a preferred embodiment of the present invention. It is a tapped delay line including delay elements **71**, multipliers **72** and adders **73**. A reference channel **74** is fed into the tapped delay line. Delayed signals are multiplied by filter coefficients, F_i, **75** and added to produce an output **76**. The filter coefficients are set in such a way that the filter amplifies the low-magnitude frequency components of an input signal to obtain an output signal having a substantially flat frequency spectrum.

As mentioned before in the background section, the output of a conventional adaptive beamformer suffers a non-uniform frequency behavior. This is because the reference channels do not have a flat frequency spectrum. The receiving sensitivity of a beamformer toward a particular angular direction is often described in terms of a gain curve. As mentioned before, the reference channel is obtained by placing a null in the gain curve (making the sensor array insensitive) in the direction of the signal source. The resulting gain curve has a lower gain for lower frequency signals than higher frequency signals. Since the reference channel is modified to generate a cancelling signal, a non-flat frequency spectrum of the reference channel is translated to a non-uniform frequency behavior in the system output.

The decolorizing filter is a fixed-coefficient filter which flattens the frequency spectrum of the reference channel (thus "decolorizing" the reference channel) by boosting the low frequency portion of the reference channel. By adding the decolorizing filters to all outputs of the reference channel matrix unit, a substantially flat frequency response in all directions is obtained.

The decolorizing filter in the illustrated embodiment uses a tapped delay line filter which is the same as a finite impulse response (FIR) filter, but other kinds of filters such as an infinite impulse response (IIR) filter can also be used for the decolorizing filter in an alternative embodiment.

FIG. **8** depicts schematically the inhibiting unit in accordance with a preferred embodiment of the present invention. It includes power estimation units **81**, **82** which estimate the power of a main channel **83** and each reference channel **84**, respectively. A sample power estimation unit **85** calculates the power of each sample. A multiplier **86** multiplies the power of each sample by a fraction, α, which is the reciprocal of the number of samples for a given averaging period to obtain an average sample power **87**. An adder **88** adds the average sample power to the output of another multiplier **89** which multiplies a previously calculated main channel power average 90 by (1−α). A new main channel power average is obtained by (new sample power)×α+(old power average)×(1−α). For example, if a 100-sample average is used, α=0.01. The updated power average will be (new sample power)×0.01 +(old power average)×0.99. In this way, the updated power average will be available at each sampling instant rather than after an averaging period. Although the illustrated embodiment shows an on-the-fly estimation method of the power average, other kinds of power estimation methods can also be used in an alternative embodiment.

A multiplier **91** multiplies the main channel power **89** with a threshold **92** to obtain a normalized main channel power average **93**. An adder **94** subtracts reference channel power averages **95** from the normalized main channel power average **93** to produce a difference **96**. If the difference is positive, a comparator **97** generates an inhibit signal **98**. The inhibit signal is provided to the adaptive filters to stop the adaptation process to prevent signal leakage.

Although the illustrated embodiment normalizes the main channel power average, an alternative embodiment may normalize the reference channel power average instead of the main channel power average. For example, if the threshold **92** in the illustrated embodiment is 0.25, the same effect can be obtained in the alternative embodiment by normalizing each reference channel power average by multiplying it by 4.

This inhibition approach is different from the prior art SNR-based inhibition approach mentioned in the background section in that it detects the presence of significant directional interference which the prior art approach does not consider. As a result, the directional-interference-based inhibition approach stops the adaptation process when there is no significant directional interference to be eliminated, whereas the prior art approach does not.

For example, where there is a weak source signal (e.g. during speech intermission) and there is almost no directional interference except some uncorrelated noise (such as noise due to wind or mechanical vibrations on the sensor structure), the SNR-based approach would allow the adaptive filter to continue adapting due to the small SNR. The continued adaptation process is not desirable because there is very little directional interference to be eliminated in the first place, and the adaptation process searches in vain for new filter weights to eliminate the uncorrelated noise, which often results in cancelling the source signal component of the received signal.

By contrast, the directional-interference-based inhibition mechanism will inhibit the adaptation process in such a case because the strength of directional interference as reflected in the reference channel power average will be smaller than the normalized main channel power average, producing a positive normalized power difference. The adaptive process is inhibited as a result until there is some directional interference to be eliminated.

FIG. **9** shows the frequency-selective constraint adaptive filter together with the difference unit in accordance with a preferred embodiment of the present invention. The frequency-selective constraint adaptive filter **101** includes a finite impulse response (FIR) filter **102**, an LMS weight updating unit **103** and a frequency-selective weight-constraint unit **104**. In an alternative embodiment, an infinite impulse response (IIR) filter can be,used instead of the FIR filter.

A flat-frequency reference channel **105** passes through FIR filter **102** whose filter weights are adjusted to produce a cancelling signal **106** which closely approximates the actual interference signal component present in a main channel **107**. In a preferred embodiment, the main channel is obtained from the main channel matrix unit after a delay in order to synchronize the main channel with the cancelling signal. In general, there is a delay between the main channel and the cancelling signal because the cancelling signal is obtained by processing reference channels through extra stages of delay, i.e., the decolorization filters and adaptive filters. In an alternative embodiment, the main channel directly from the main channel matrix unit may be used if the delay is not significant.

A difference unit **108** subtracts cancelling signal **106** from main channel **107** to generates an output signal **109**. Adaptive filter **101** adjusts filter weights, W_1–W_n, to minimize the power of the output signal. When the filter weights settle, output signal **109** generates the source signal substantially free of the actual interference signal component because cancelling signal **106** closely tracks the interference signal component. Output signal **109** is sent to the output D/A unit to produce an analog output signal. Output signal **109** is also used to adjust the adaptive filter weights to further reduce the interference signal component.

There are many techniques to continuously update the values of the filter weights. The preferred embodiment uses the Least Mean-Square (LMS) algorithm which minimize the mean-square value of the difference between the main channel and the cancelling signal, but in an alternative embodiment, other algorithms such as Recursive Least Square (RLS) can also be used.

Under the LMS algorithm, the adaptive filter weights are updated according to the following:

$$W_p(n+1)=W_p(n)+2\ \mu r(n-p)\ e(n)$$

where n is a discrete time index; W_p is a p-th filter weight of the adaptive filter; e(n) is a difference signal between the main channel signal and the cancelling signal; r(n) is a reference channel; and μ is an adaptation constant that controls the speed of adaptation.

FIG. **10** depicts a preferred embodiment of the frequency-selective weight-constraint unit. The frequency-selective weight-control unit **110** includes a Fast Fourier Transform (FFT) unit **112**, a set of frequency bins **114**, a set of truncating units **115**, a set of storage cells **116**, and an Inverse Fast Fourier Transform (IFFT) unit **117**, connected in series.

The FFT unit **112** receives adaptive filter weights **111** and performs the FFT of the filter weights **111** to obtain frequency representation values **113**. The frequency representation values are then divided into a set of frequency bands and stored into the frequency bins **114*a*–114*h***. Each frequency bin stores the frequency representation values within a specific bandwidth assigned to each bin. The values represent the operation of the adaptive filter with respect to a specific frequency component of the source signal. Each of the truncating units **115*a*–115*h*** compares the frequency representation values with a threshold assigned to each bin, and truncates the values if they exceeds the threshold. The truncated frequency representation values are temporarily stored in **116*a*–116*h*** before the IFFT unit **117** converts them back to new filter weight values **118**.

In addition to the inhibiting mechanism based on directional interference, the frequency-selective weight-constraint unit further controls the adaptation process based on the frequency spectrum of the received source signal. Once the adaptive filter starts working, the performance change in the output of the filter, better or worse, becomes drastic. Uncontrolled adaptation can quickly lead to a drastic performance degradation.

The weight-constraint mechanism is based on the observation that a large increase in the adaptive filter weight values hints signal leakage. If the adaptive filter works properly, there is no need for the filter to increase the filter weights to large values. But, if the filter is not working properly, the filter weights tend to grow to large values.

One way to curve the growth is to use a simple truncating mechanism to truncate the values of filter weights to predetermined threshold values. In this way, even if the overall signal power may be high enough to trigger the inhibition mechanism, the weight-constraint mechanism can still prevent the signal leakage.

For narrow band signals, such as a speech signal or a tonal signal, having their power spectral density concentrated in a narrow frequency range, signal leakage may not be manifested in a large growth of the filter weight values in the time domain. However, the filter weight values in the frequency domain will indicate some increase because they represent the operation of the adaptive filter in response to a specific frequency component of the source signal. The frequency-selective weight-constraint unit detects that condition by sensing a large increase in the frequency representation values of the filter weights. By truncating the frequency representation values in the narrow frequency band of interest and inverse-transforming them back to the time domain, the unit acts to prevent the signal leakage involving narrow band signals.

The system described herein may be implemented using commercially available digital signal processing (DSP) systems such as Analog Device 2100 series.

FIG. **11** shows a flow chart depicting the operation of a program for a DSP processor in accordance with a preferred embodiment of the present invention.

After the program starts at step **100**, the program initializes registers and pointers as well as buffers (step **110**). The program then waits for an interrupt from a sampling unit requesting for processing of samples received from the array of sensors (step **120**). When the sampling unit sends an interrupt (step **131**) that the samples are ready, the program reads the sample values (step **130**) and stores the values (step **140**). The program filters the stored values using a routine implementing a tapped delay line and stores the filtered input values (step **141**).

The program then retrieves the filtered input values (step **151**) and main channel matrix coefficients (step **152**) to generate a main channel (step **150**) by multiplying the two and to store the result (step **160**).

The program retrieves the filtered input values (step **171**) and reference channel matrix coefficients (step **172**) to generate a reference channel (reference channel #1) by multiplying the two (step **170**) and to store the result (step **180**). Steps **170** and **180** are repeated to generate all other reference channels (step **190**).

The program retrieves one of the reference channels (step **201**) and decolorization filter coefficients for the corresponding reference channel (step **202**) to generate a flat-frequency reference channel by multiplying the two (step **200**) and stores the result (step **210**). Steps **200** and **210** are repeated for all other reference channels (step **220**).

The program retrieves one of the flat-frequency reference channels (step **231**) and adaptive filter coefficients (step **232**) to generate cancelling signal (step **230**) by multiplying the two and to store the result (step **240**). Steps **230** and **240** are repeated for all other reference channels to generate more cancelling signals (step **250**).

The program retrieves cancelling signals (steps **262–263**) to subtract them from the main channel (retrieved at step **261**) to cancel the interference signal component in the main channel (step **260**). The output is send to a D/A unit to reproduce the signal without interference in analog form (step **264**). The output is also stored (step **270**).

The program calculates the power of a reference channel sample (step **281**) and retrieves an old reference channel power average (step **282**). The program multiplies the sample power by a and the old power average by (1–α), and sums them (step **280**), and stores the result as a new power average (step **290**). This process is repeated for all other reference channels (step **300**) and the total sum of power averages of all reference channels is stored (step **310**).

The program multiplies the power of a main channel sample (retrieved at step **321**) by α and an old main channel power average (retrieved at step **322**) by (1–α), sums them (step **320**) and stores them as a new main channel power average (step **330**).

The program then multiplies the main channel power with a threshold to obtain a normalized main channel power average (step **340**). The program subtracts the total reference channel power average (retrieved at step **341**) from the normalized main channel power average to produce a difference (step **350**). If the difference is positive, the program goes back to step **120** where it simply waits for another samples.

If the difference is negative, the program enters a weight-updating routine. The program calculates a new filter weight by adding [2×adaptation constant×reference channel sample (retrieved at step **361**)×output (retrieved at step **362**)] to an old filter weight (retrieved at step **363**) to update the weight (step **360**) and stores the result (step **370**).

The program performs the FFT of the new filter weights to obtain their frequency representation (step **380**). The frequency representation values are divided into several frequency bands and stored into a set of frequency bins (step **390**). The frequency representation values in each bin are compared with a threshold associated with each frequency bin (step **400**). If the values exceed the threshold, the values are truncated to the threshold (step **410**). The program performs the IFFT to convert the truncated frequency representation values back to filter weight values (step **420**) and stores them (step **430**). The program repeats the weight-updating routine, steps **360**–**430**, for all other reference channels and associated adaptive filters (step **440**). The program then goes back to step **120** to wait for an interrupt for a new round of processing samples (step **450**).

While the invention has been described with reference to preferred embodiments, it is not limited to those embodiments. It will be appreciated by those of ordinary skill in the art that modifications can be made to the structure and form of the invention without departing from its spirit and scope which is defined in the following claims.

What is claimed is:

1. An adaptive system for processing digital input data representing signals containing a source signal from a signal source on-axis relative to an array of sensors as well as interference signals from interference sources located off-axis from the signal source and for producing digital output data representing the source signal with reduced interference signals relative to the source signal, comprising:

a main channel matrix unit for generating a main channel from the digital input data, the main channel representing signals received in the direction of the signal source and having a source signal component and an interference signal component;

a reference channel matrix unit for generating at least one reference channel from the digital input data, each reference channel representing signals received in directions other than that of the signal source;

at least one adaptive filter having adaptive filter weights, connected to receive signals from the reference channel matrix unit, for generating a cancelling signal approximating the interference signal component of the main channel;

a difference unit, connected to receive signals from the main channel matrix unit and said at least one adaptive filter, for generating the digital output data by subtracting the cancelling signal from the main channel;

said at least one adaptive filter also being connected to receive the digital output data and including weight updating means for finding new filter weight values of said at least one adaptive filter such that the difference between the main channel and the cancelling signal is minimized; and

weight constraining means for truncating said new filter weight values to predetermined threshold values when each of the new filter weight values exceeds the corresponding threshold value.

2. The system of claim **1**, further comprising at least one decolorizing filter for filtering said at least one reference channel so that it has a frequency spectrum whose magnitude is substantially flat over a predetermined frequency range.

3. The system of claim **1**, further comprising inhibiting means, connected to receive signals from the main channel matrix unit and the reference channel matrix unit, for estimating the power of the main channel and the power of said at least one reference channel and for generating an inhibit signal to said weight updating means when a normalized power difference between the main channel and said at least one reference channel is positive.

4. The system of claim **1** wherein the sensors are microphones.

5. An adaptive system for processing digital input data representing signals containing a source signal from a signal source on-axis relative to an array of sensors as well as interference signals from interference sources located off-axis from the signal source and for producing digital output data representing the source signal with reduced interference signals relative to the source signal, comprising:

a main channel matrix unit for generating a main channel from the digital input data, the main channel representing signals received in the direction of the signal source and having a source signal component and an interference signal component;

a reference channel matrix unit for generating at least one reference channel from the digital input data, each reference channel representing signals received in directions other than that of the signal source;

at least one adaptive filter having adaptive filter weights, connected to receive signals from the reference channel matrix unit, for generating a cancelling signal approximating the interference signal component of the main channel;

a difference unit, connected to receive signals from the main channel matrix unit and said at least one adaptive filter, for generating digital output data by subtracting the cancelling signal from the main channel;

said at least one adaptive filter also being connected to receive the digital output data and including weight updating means for finding new filter weight values of said at least one adaptive filter such that the difference between the main channel and the cancelling signal is minimized; and

weight constraining means for converting the new filter weight values to frequency representation values, truncating the frequency representation values to predetermined threshold values, and converting them back to adaptive filter weights.

6. The system of claim **5**, further comprising at least one decolorizing filter for filtering said at least one reference channel so that it has a frequency spectrum whose magnitude is substantially flat over a predetermined frequency range.

7. The system of claim **5**, further comprising inhibiting means, connected to receive signals from the main channel matrix unit and the reference channel matrix unit, for estimating the power of the main channel and the power of said at least one reference channel and for generating an inhibit signal to said weight updating means when a normalized power difference between the main channel and said at least one reference channel is positive.

8. The system of claim **5** wherein the sensors are microphones.

9. An adaptive system for receiving a source signal from a signal source on-axis relative to the system as well as interference signals from interference sources located off-axis from the signal source and for producing an output signal with reduced interference signals relative to the source signal, comprising:

a sensor array of spatially distributed sensors, each for receiving such source and interference signals;

a sampling unit, connected to receive signals from the sensor array, for converting such signals to digital form;

a main channel matrix unit, connected to receive signals from the sampling unit, for generating a main channel

- representing signals received in the direction of the signal source, the main channel having a source signal component and an interference signal component;
- a reference channel matrix unit, connected to receive signals from the sampling unit, for generating at least one reference channel, each reference channel representing signals received in directions other than that of the signal source;
- at least one adaptive filter having adaptive filter weights, connected to receive signals from the reference channel matrix unit, for generating a cancelling signal approximating the interference signal component of the main channel;
- a difference unit, connected to receive signals from the main channel matrix unit and said at least one adaptive filter, for subtracting the cancelling signal from the main channel to generate a digital output signal;
- an output digital-to-analog converter for converting said digital output signal to analog form;
- said at least one adaptive filter also being connected to receive the digital output signal of the difference unit and including weight updating means for finding new filter weight values of said at least one adaptive filter such that the difference between the main channel and the cancelling signal is minimized; and
- weight constraining means for truncating said new filter weight values to predetermined threshold values when each of the new filter weight value exceeds the corresponding threshold value.

10. The system of claim **9**, further comprising at least one decolorizing filter for filtering said at least one reference channel so that it has a frequency spectrum whose magnitude is substantially flat over a predetermined frequency range.

11. The system of claim **9**, further comprising inhibiting means, connected to receive signals from the main channel matrix unit and the reference channel matrix unit, for estimating the power of the main channel and the power of said at least one reference channel and for generating an inhibit signal to said weight updating means when a normalized power difference between the main channel and said at least one reference channel is positive.

12. The system of claim **9**, further comprising delay means for delaying the main channel so that the main channel is synchronized with the cancelling signal before the difference unit subtracts the cancelling signal from the main channel.

13. The system of claim **9** wherein the sensors are microphones.

14. The system of claim **13** wherein the microphones are omnidirectional microphones.

15. The system of claim **13** wherein the microphones are unidirectional microphones.

16. The system of claim **9** wherein the main channel matrix unit includes beamforming means for spatially filtering signals from the sampling unit to exhibit the highest sensitivity toward the signal source.

17. The system of claim **9** wherein the reference channel matrix unit includes beamforming means for spatially filtering signals from the sampling unit to exhibit the lowest sensitivity toward the signal source.

18. The system of claim **9** wherein said at least one adaptive filter comprises a finite-impulse-response filter for generating the cancelling signal.

19. The system of claim **9** wherein said at least one adaptive filter comprises an infinite-impulse-response filter for generating the cancelling signal.

20. The system of claim **9** wherein the weight updating means uses the least-mean-square algorithm where the mean-square value of the difference between the main channel and the cancelling signal is minimized.

21. An adaptive system for receiving a source signal from a signal source on-axis relative to the system as well as interference signals from interference sources located off-axis from the signal source and for producing an output signal with reduced interference signals relative to the source signal, comprising:

- a sensor array of spatially distributed sensors, each for receiving such source and interference signals;
- a sampling unit, connected to receive signals from the sensor array, for converting such signals to digital form;
- a main channel matrix unit, connected to receive signals from the sampling unit, for generating a main channel representing signals received in the direction of the signal source, the main channel having a source signal component and an interference signal component;
- a reference channel matrix unit, connected to receive signals from the sampling unit, for generating at least one reference channel, each reference channel representing signals received in directions other than that of the signal source;
- at least one adaptive filter having adaptive filter weights, connected to receive signals from the reference channel matrix unit, for generating a cancelling signal approximating the interference signal component of the main channel;
- a difference unit, connected to receive signals from the main channel matrix unit and said at least one adaptive filter, for subtracting the cancelling signal from the main channel to generate a digital output signal;
- an output digital-to-analog converter for converting the digital output signal to analog form;
- said at least one adaptive filter also being connected to receive the digital output signal of the difference unit and including weight updating means for finding new filter weight values of said at least one adaptive filter such that the difference between the main channel and the cancelling signal is minimized; and
- weight constraining means for constraining the operation of the adaptive filter by converting the new filter weight values to frequency representation values, truncating the frequency representation values to predetermined threshold values, and converting them back to adaptive filter weights.

22. The system of claim **21** wherein the weight constraining means comprises:

- a Fast Fourier Transform unit for generating frequency representation values of the new filter weight values;
- a set of frequency bins, each frequency bin for storing the frequency representation values for a frequency band assigned to each frequency bin;
- a set of truncating means, each connected to the corresponding frequency bin, for truncating the frequency representation values stored in each frequency bin to a predetermined threshold value if the frequency representation values exceed the threshold value associated with each frequency bin; and
- an Inverse Fast Fourier Transform unit, connected to the set of truncating means, for converting values from the set of truncating means back to adaptive filter weights.

23. The system of claim **21**, further comprising at least one decolorizing filter for filtering said at least one reference channel so that it has a frequency spectrum whose magnitude is substantially flat over a predetermined frequency range.

24. The system of claim **21**, further comprising inhibiting means, connected to receive signals from the main channel matrix unit and the reference channel matrix unit, for estimating the power of the main channel and the power of said at least one reference channel and for generating an inhibit signal to said weight updating means when a normalized power difference between the main channel and said at least one reference channel is positive.

25. The system of claim **21** wherein the sensors are microphones.

26. The system of claim **21** wherein the main channel matrix unit includes beamforming means for spatially filtering signals from the sampling unit to exhibit the highest sensitivity toward the signal source.

27. The system of claim **21** wherein the reference channel matrix unit includes beamforming means for spatially filtering signals from the sampling unit to exhibit the lowest sensitivity toward the signal source.

28. The system of claim **21** wherein the weight updating means uses the least-mean-square algorithm where the mean-square value of the difference between the main channel and the cancelling signal is minimized.

* * * * *

US005627799A

United States Patent [19]

Hoshuyama

[11] **Patent Number: 5,627,799**

[45] **Date of Patent: May 6, 1997**

[54] **BEAMFORMER USING COEFFICIENT RESTRAINED ADAPTIVE FILTERS FOR DETECTING INTERFERENCE SIGNALS**

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[73] Assignee: **NEC Corporation**, Tokyo, Japan

[21] Appl. No.: **523,059**

[22] Filed: **Sep. 1, 1995**

[30] **Foreign Application Priority Data**

Sep. 1, 1994 [JP] Japan 6-208635

[51] **Int. Cl.^{6}** **G01S 15/00**

[52] **U.S. Cl.** **367/121**; 367/901; 367/119; 367/905; 381/94

[58] **Field of Search** 367/12, 119, 121, 367/123, 129, 901, 905, 103; 128/661.01; 381/94

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Primary Examiner—Ian J. Lobo

Attorney, Agent, or Firm—Sughrue, Mion, Zinn, Macpeak & Seas

[57] **ABSTRACT**

In an adaptive array beamformer, a spatial beamforming filter is connected to a sensor array for respectively filtering and summing array signals to produce a first filter output containing a target signal that arrives in a specified direction. First adaptive filters provide transversal-filtering the first filter output to produce a second filter output not containing the target signal, using a first error signal by restraining their tap weight coefficients. The array signals are further coupled to subtractors. Each subtractor detects a difference between the second filter output of the corresponding first adaptive filter and the corresponding sensor signal to derive the first error signal. Second adaptive filters provide transversal-filtering the first error signals of the subtractors to produce third filter outputs, using a second error signal, by restraining their tap weight coefficients. The third filter outputs are summed and subtracted from the first filter output to produce an output of the beamformer, which is supplied as the second error signal to the second adaptive filters

10 Claims, 11 Drawing Sheets

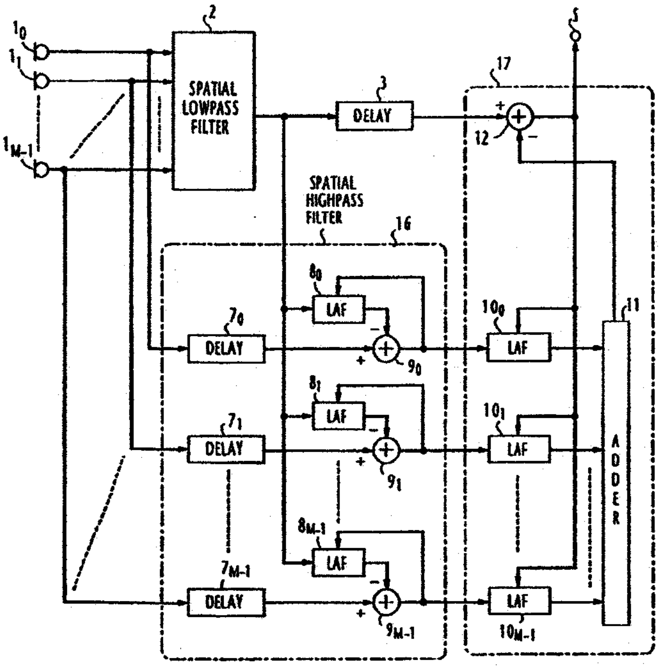
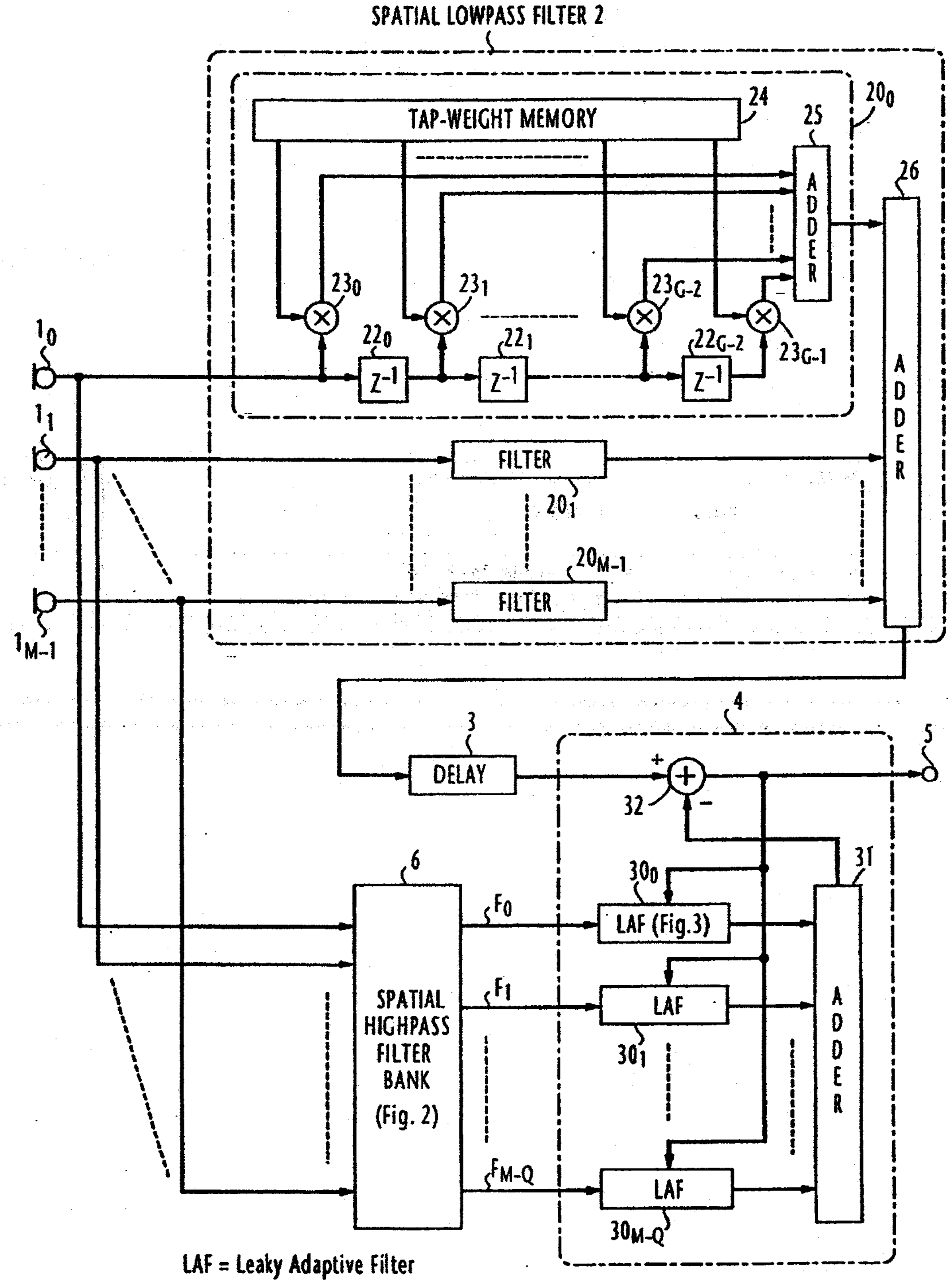


FIG. 1
PRIOR ART



LAF = Leaky Adaptive Filter

FIG. 2 PRIOR ART

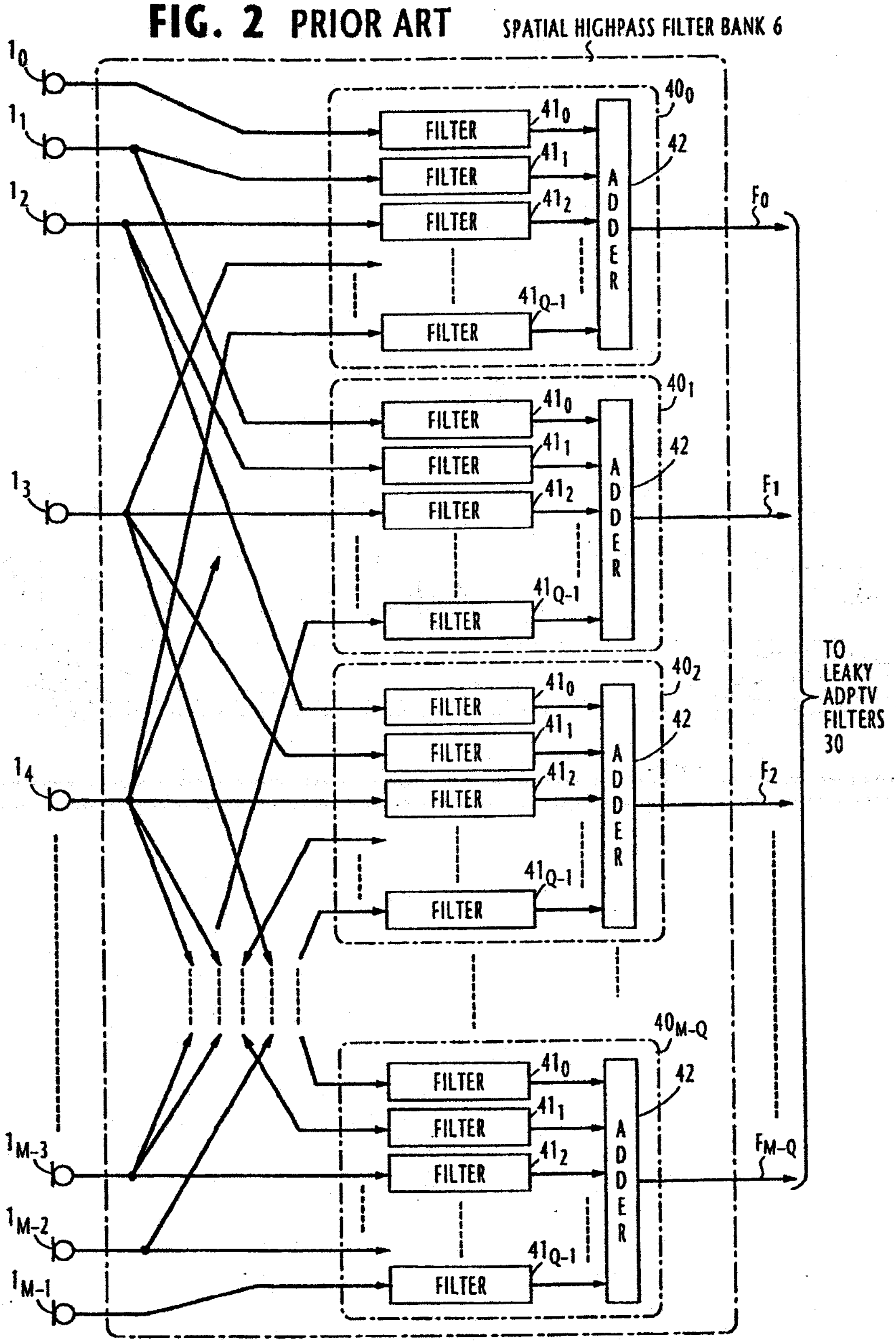


FIG. 3
PRIOR ART

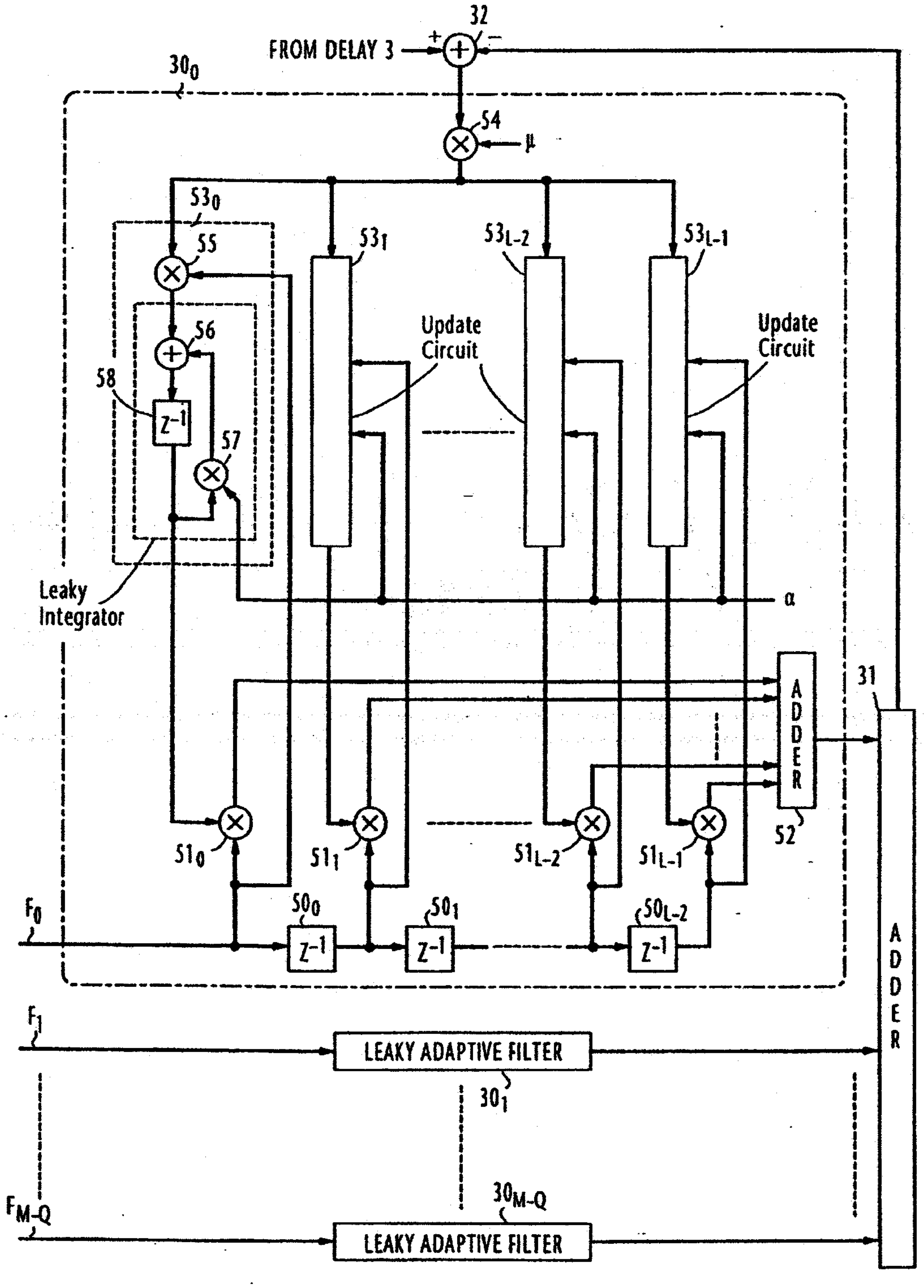


FIG. 4

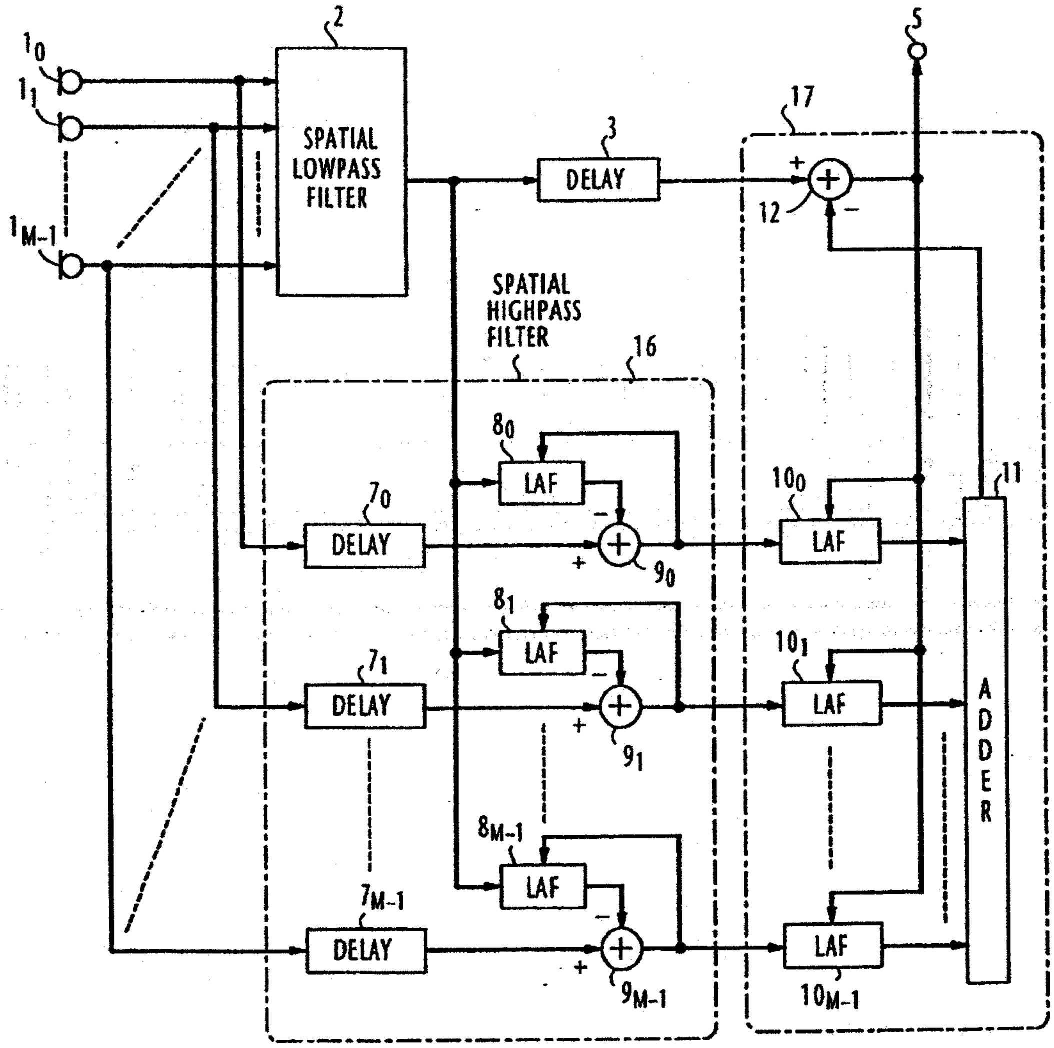
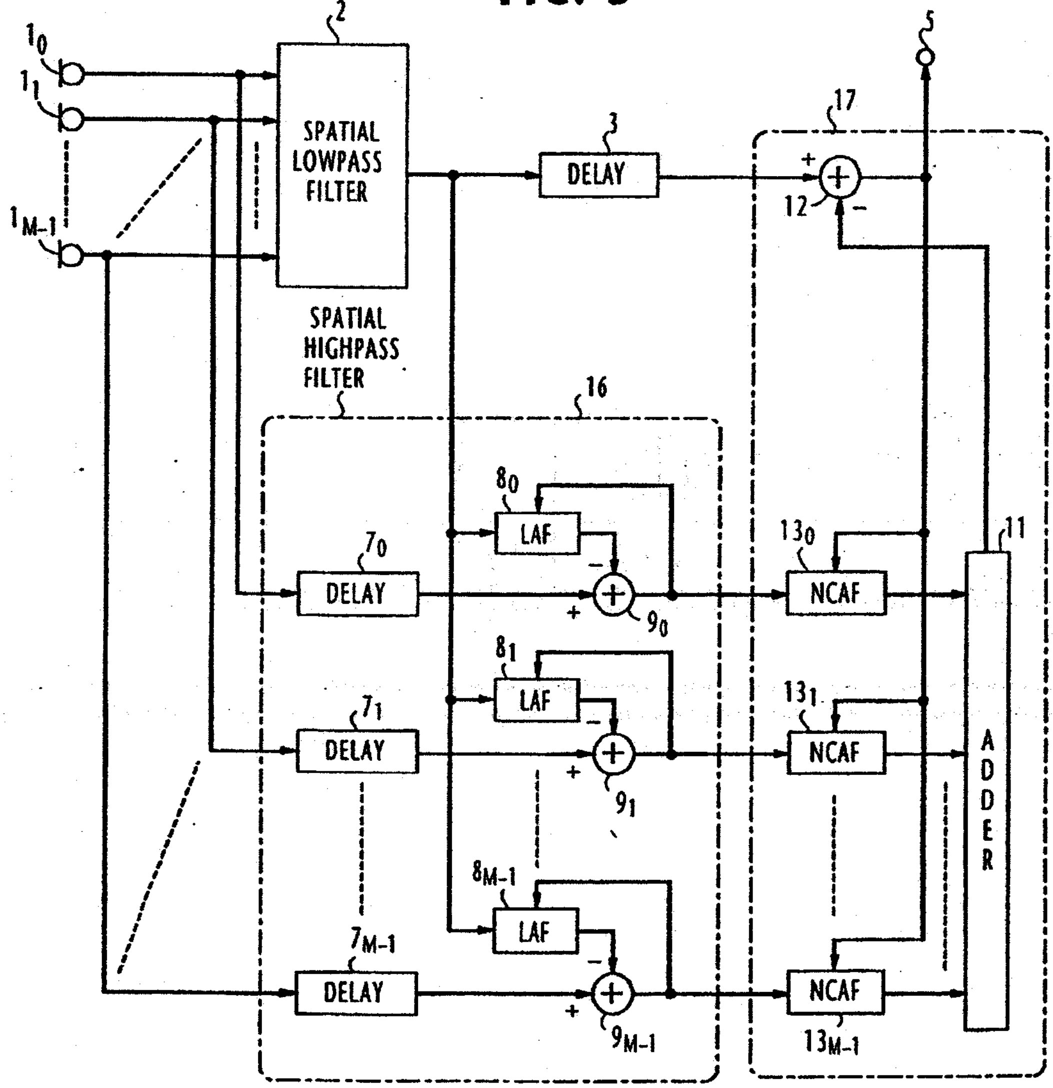

1_0
1_1
1_{M-1}
2
SPATIAL LOWPASS FILTER
3
DELAY
5
17
12
+
−
SPATIAL HIGHPASS FILTER
16
8_0
LAF
7_0
DELAY
9_0
8_1
LAF
7_1
DELAY
9_1
8_{M-1}
LAF
7_{M-1}
DELAY
9_{M-1}
10_0
LAF
10_1
LAF
LAF
10_{M-1}
11
ADDER

FIG. 5



NCAF = Norm-Constrained Adaptive Filter

FIG. 6

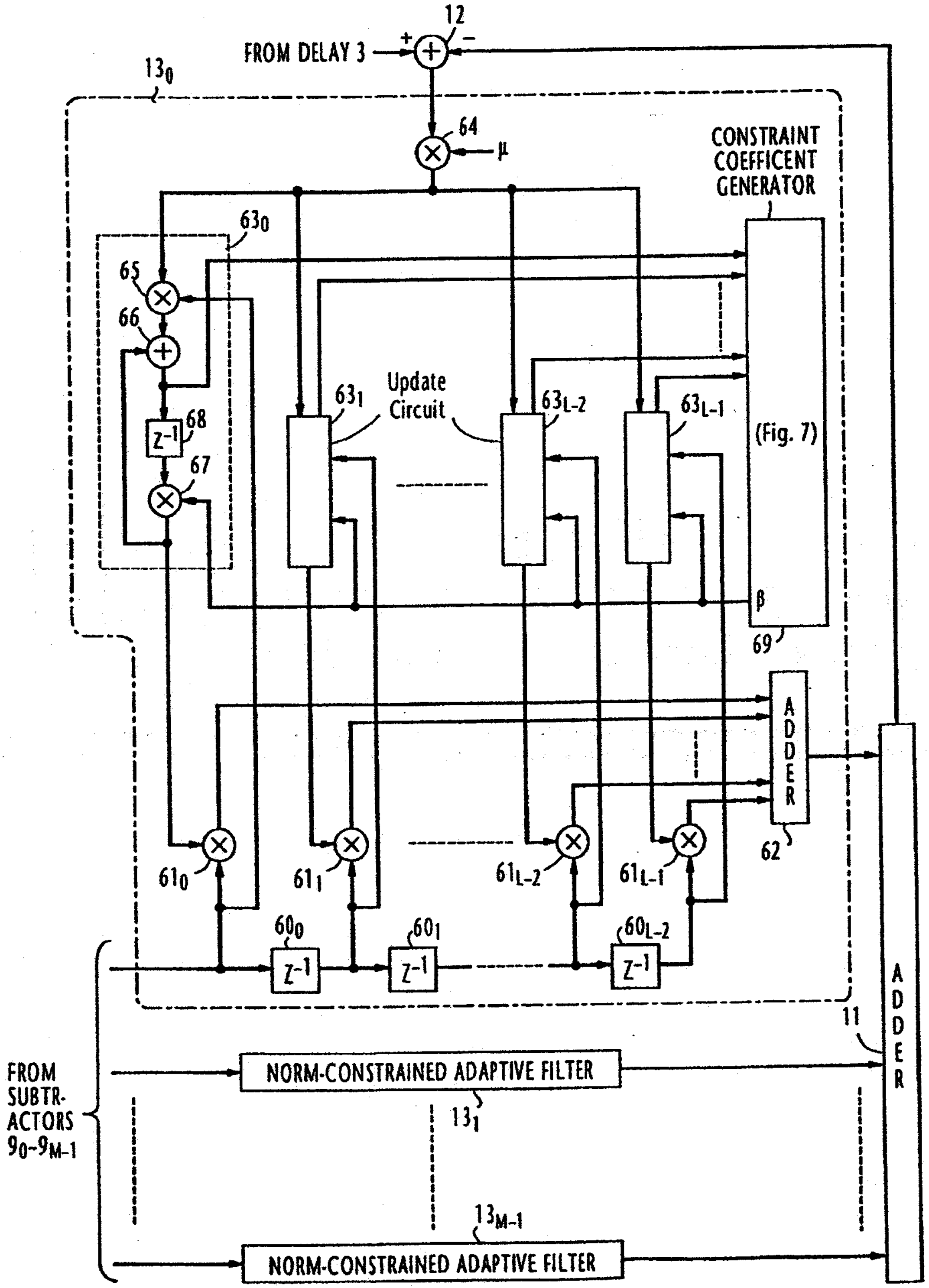

12
FROM DELAY 3
+
−
13₀
64
μ
CONSTRAINT COEFFICENT GENERATOR
63₀
65
66
68
Z⁻¹
67
63₁
Update Circuit
63L-2
63L-1
(Fig. 7)
β
69
ADDER
62
61₀
61₁
61L-2
61L-1
60₀
60₁
60L-2
Z⁻¹
Z⁻¹
Z⁻¹
11
ADDER
FROM SUBTR-ACTORS 9₀~9M-1
NORM-CONSTRAINED ADAPTIVE FILTER
13₁
13M-1
NORM-CONSTRAINED ADAPTIVE FILTER

FIG. 7

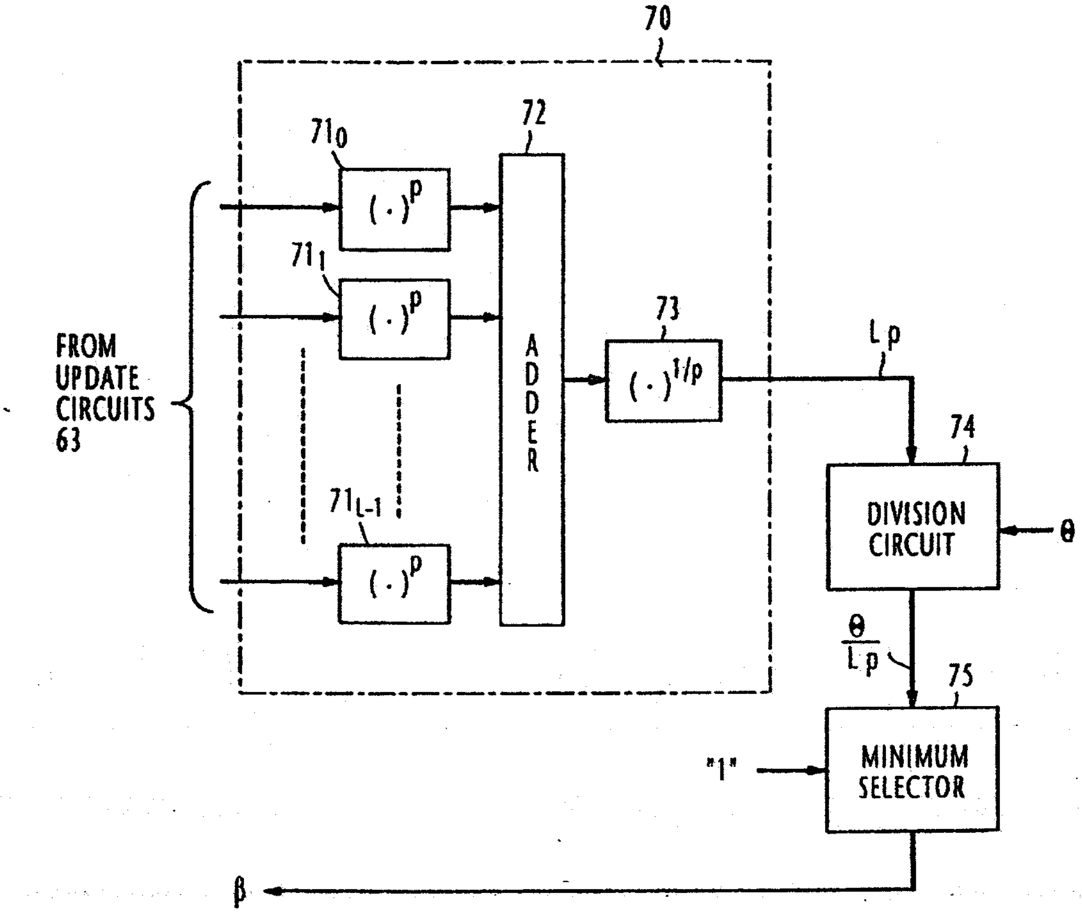

70
72
71_0
$(\cdot)^p$
71_1
$(\cdot)^p$
71_{L-1}
$(\cdot)^p$
FROM UPDATE CIRCUITS 63
ADDER
73
$(\cdot)^{1/p}$
Lp
74
DIVISION CIRCUIT
θ
$\frac{\theta}{Lp}$
75
"1"
MINIMUM SELECTOR
β

FIG. 10

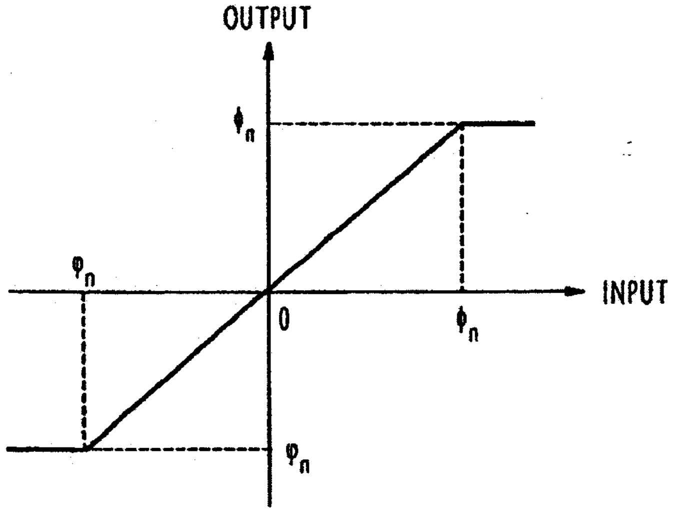
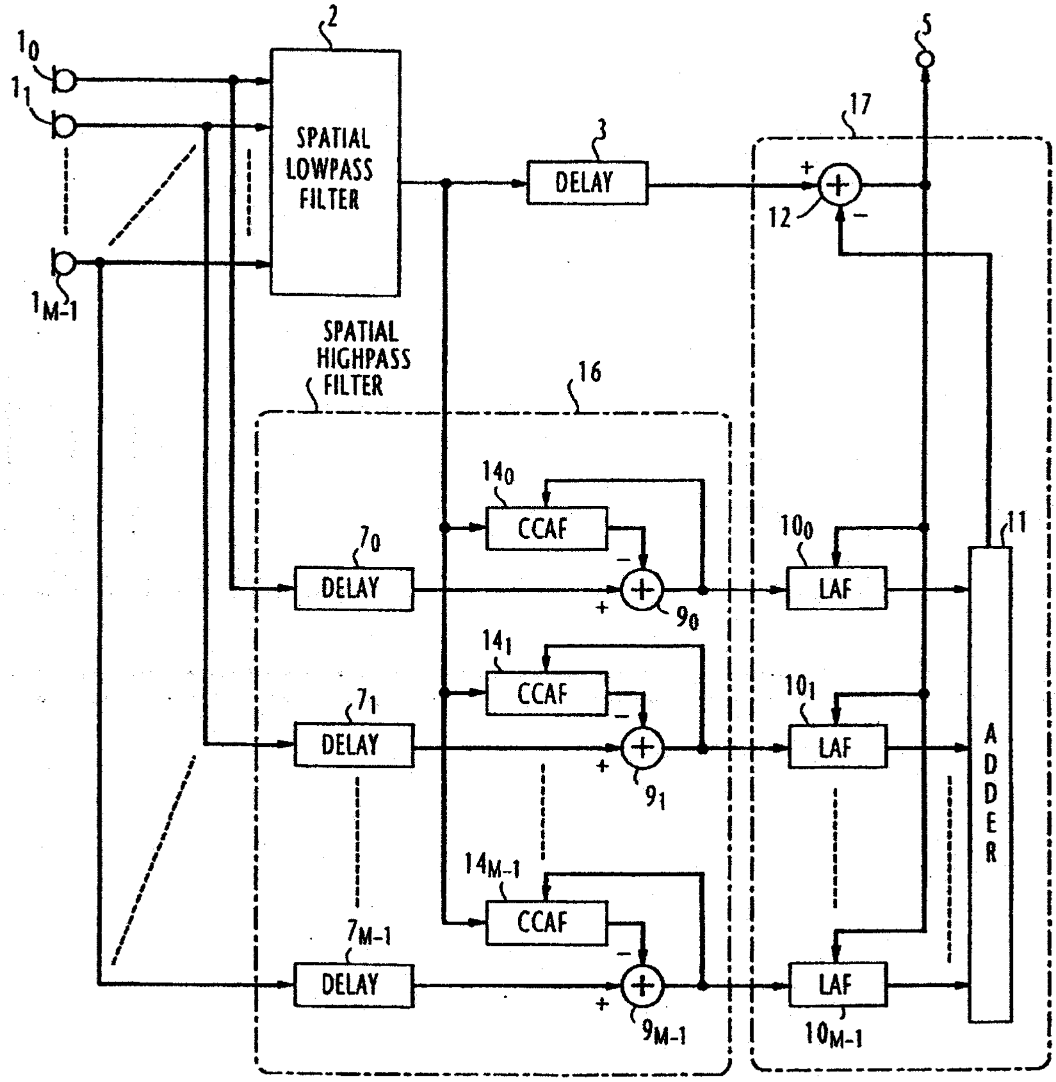
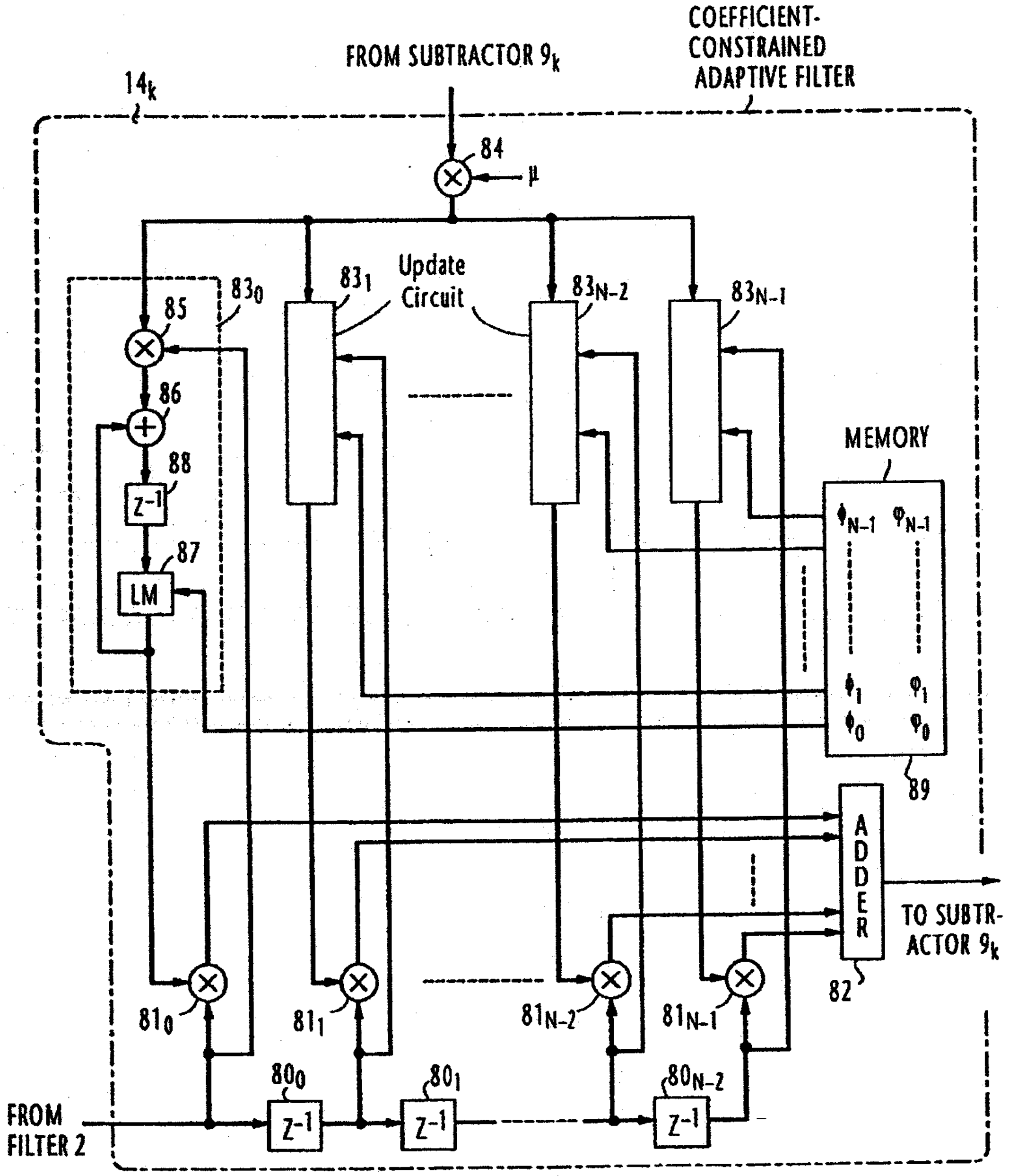

OUTPUT
INPUT
ϕ_n
φ_n
0
ϕ_n
φ_n

FIG. 8



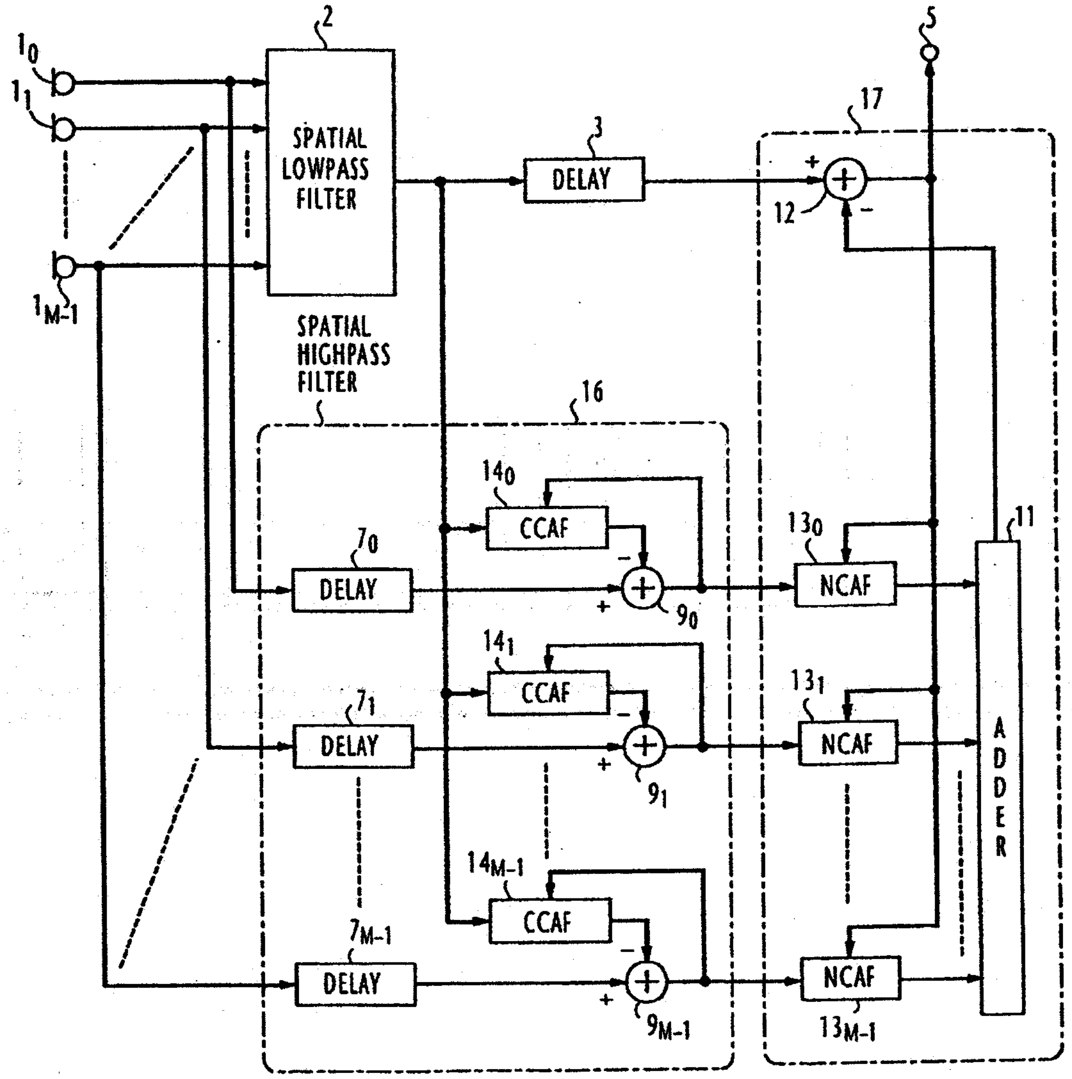
CCAF = Coefficient-Constrained Adaptive Filter

FIG. 9



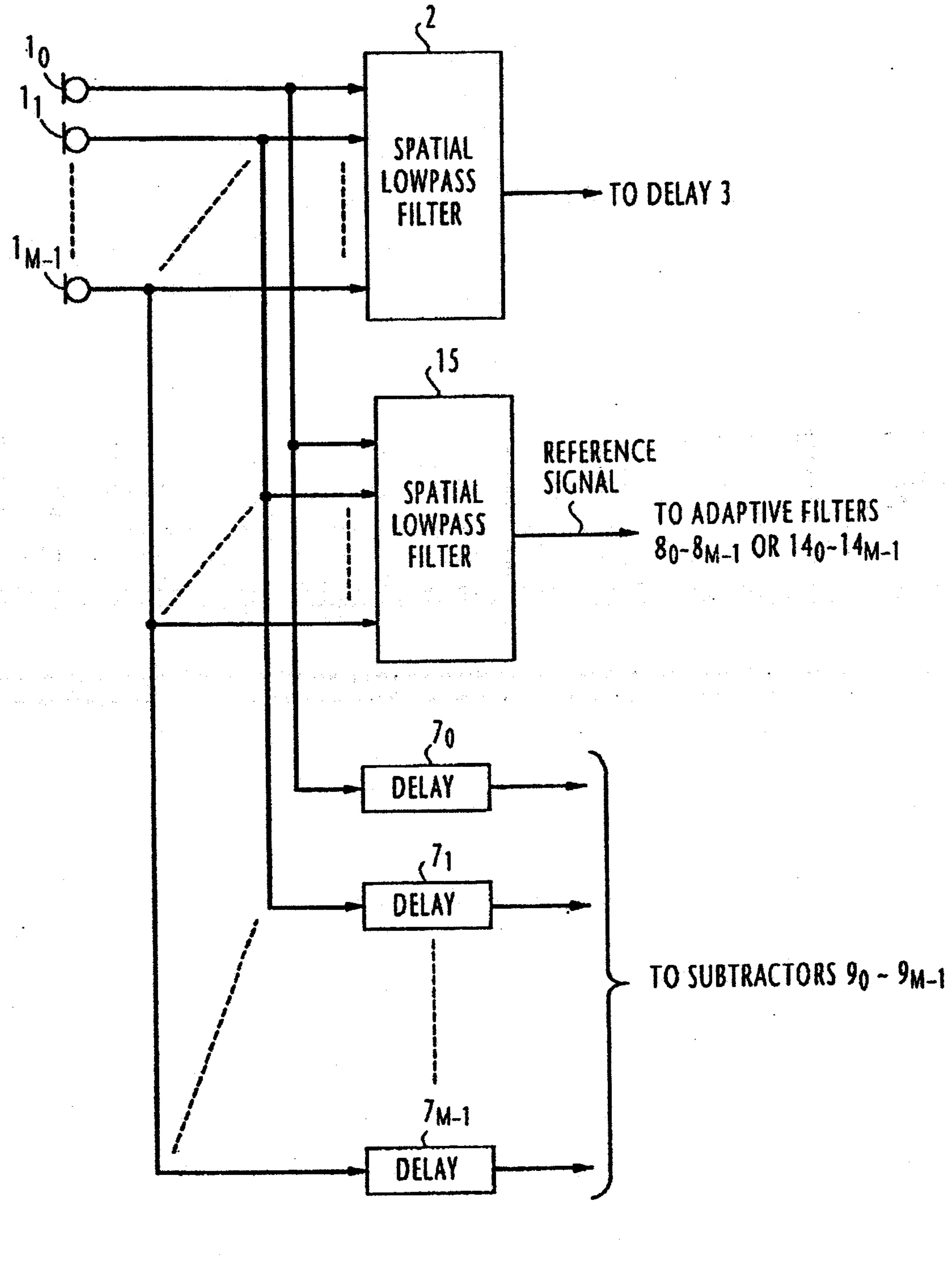
COEFFICIENT-CONSTRAINED ADAPTIVE FILTER
FROM SUBTRACTOR 9_k
14_k
84
μ
83_0
83_1
Update Circuit
83_{N-2}
83_{N-1}
85
86
88
Z^{-1}
87
LM
MEMORY
ϕ_{N-1}
φ_{N-1}
ϕ_1
φ_1
ϕ_0
φ_0
89
ADDER
82
TO SUBTRACTOR 9_k
81_0
81_1
81_{N-2}
81_{N-1}
80_0
80_1
80_{N-2}
Z^{-1}
Z^{-1}
Z^{-1}
FROM FILTER 2

FIG. 11



CCAF = Coefficient-Constrained Adaptive Filter
NCAF = Norm-Constrained Adaptive Filter

FIG. 12



2
1_0
1_1
1_{M-1}
SPATIAL LOWPASS FILTER
TO DELAY 3
15
SPATIAL LOWPASS FILTER
REFERENCE SIGNAL
TO ADAPTIVE FILTERS 8_0-8_{M-1} OR 14_0-14_{M-1}
7_0
DELAY
7_1
DELAY
7_{M-1}
DELAY
TO SUBTRACTORS 9_0 - 9_{M-1}

BEAMFORMER USING COEFFICIENT RESTRAINED ADAPTIVE FILTERS FOR DETECTING INTERFERENCE SIGNALS

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates generally to interference cancelers, and more particularly to a generalized sidelobe canceler, or adaptive beamformer for an array of sensors such as microphones and the like.

2. Description of the Related Art

It is known that wideband signals propagating across an array of sensors in directions that are different than the beam steering direction of the array suffer a distortion that is similar to lowpass filtering.

According to a prior art microphone array, signals detected by an array of microphones are lowpass filtered and summed together to detect a target signal that arrives in a particular direction. The adaptive microphone array beamformer is one form of the generalized sidelobe canceler as described in an article "An alternative Approach to Linearly Constrained Adaptive Beamforming", Lloyd J. Griffiths and Charles W. Jim, the IEEE Transactions on Antenna and Propagation, Vol. AP-30, No. 1, January 1982, pages 27–34. As described in an article "The Broad-Band Wiener Solution for Griffiths-Jim Beamformers", S. Nordholm, I. Claesson and P. Eriksson, the IEEE Transactions on signal Processing, Vol. 40, No. 2, February 1992, pages 474–478 (hereinafter referred to as Document 1), the generalized sidelobe canceler comprises, a spatial lowpass filter connected to an array of microphones for filtering signals from the array and summing the filtered signals so that only the desired signal is contained in the summed signal. A plurality of spatial highpass filters are provided to form a spatial highpass filter bank. Each spatial highpass filer is connected to a selected pair of microphones for filtering and summing the sensor signals to detect the interference signals. A plurality of adaptive filters are provided for using the interference signals as reference signals to detect those components having high correlation with the interference signals contained in the detected target signal.

Since the spatial highpass filters of Document 1 are of nonadaptive type and each uses two microphone outputs, the range of signals which must be rejected is very narrow. As a result, a slight departure from the intended direction causes a leakage of the desired signal into the interference path of the beamformer.

To overcome the prior art shortcoming, a proposal has been made to implement a spatial highpass filter for receiving more than two microphone outputs as described in an article "A Spatial Filtering Approach to Robust Adaptive Beaming", I. Claesson et al, the IEEE Transactions on Antennas and Propagation, Vol. 40, No. 9, September 1992, pages 1093 to 1096 (hereinafter referred to as Document 2). According to Document 2, each of the highpass filters that comprise the spatial highpass filter broadens the range of arrival angles by receiving multiple spatial samples from a selected set of microphone outputs using a plurality of leaky adaptive filters.

However, a large number of microphones (the Q value) are required to implement a beamformer having a wide range of rejection angles, for each group of spatial highpass filters in the filter bank. If a sufficient number of microphones is not provided, the degree of design freedom must be sacrificed, resulting in a beamformer having a low noise canceling capability. The difference between the assumed direction and the actual arrival direction of the target signal, or a look-direction error, is of another concern because it degrades the target signal, or a look-direction error, is of another concern because it degrades the target signal. In order to compensate for this shortcoming, the spatial highpass filter bank of the prior art needs as many spatial highpass filters as is necessary to provide a wide range of angles to reject the target signal to prevent its leakage into the interference path of the beamformer.

SUMMARY OF THE INVENTION

It is therefore an object of the present invention to provide an adaptive array beamformer with a reduced number of sensors while allowing a look-direction error.

According to the present invention, there is provided an adaptive array beamformer comprising an array of spatially distributed sensors, and a spatial beamforming filter connected to the sensors for respectively filtering output signals of the sensors and summing the filtered output signals to produce a first filter output containing a target signal arriving at the array in a specified direction. A plurality of first adaptive filters are provided, each having a tapped-delay line connected to receive the first filter output, a coefficient update circuit for producing tap weight coefficients indicating correlations between tap signals from the tapped-delay line and a first error signal applied thereto, a plurality of multipliers for weighting the tap signals with the coefficients, respectively, and means for summing the weighted tap signals to produce a second filter output not containing the target signal. The coefficient update means includes restraining means for preventing the coefficients from increasing indefinitely. A plurality of first subtractors are provided, each detecting a difference between a corresponding sensor signal and the second filter output of the corresponding first adaptive filter and supplying the difference to the coefficient update circuit of the corresponding first adaptive filter as the first error signal. A plurality of second adaptive filters are provided, each having a tapped-delay line connected to receive the error signal from a corresponding one of the first subtractors, a coefficient update circuit for producing tap weight coefficients indicating correlations between tap signals from the tapped-delay line and a second error signal applied thereto, a multiply-and-sum circuit for weighting the tap signals with the coefficients respectively and summing the weighted tap signals to produce a third filter output. The coefficient update circuit includes restraining means for preventing the coefficients from increasing indefinitely. An adder is provided for summing the third filter outputs from the second adaptive filters. A second subtractor detects a difference between the first filter output and the output of the adder and supplying the difference to the coefficient update circuit of the second adaptive filters as the second error signal.

In a preferred embodiment, a second spatial beamforming filter is connected to the sensors for respectively filtering output signals of the sensors and summing the filtered output signals to produce a second filter output containing the target signal, the second spatial beamforming filter having a greater beam width than a beam width of the first spatial beamforming filter. The first adaptive filters are connected to the output of the second spatial beamforming filter, instead of to the output of the first-named spatial beamforming filter.

BRIEF DESCRIPTION OF THE DRAWINGS

The present invention will be described in further detail with reference to the accompanying drawings, in which:

FIG. 1 is a block diagram of a prior art adaptive array beamformer;

FIG. 2 is a block diagram of the spatial highpass filter of the FIG. 1 prior art;

FIG. 3 is a block diagram of the leaky adaptive filters of the FIG. 1 prior art;

FIG. 4 is a block diagram of an adaptive array beamformer according to a first embodiment of the present invention;

FIG. 5 is a block diagram of an adaptive array beamformer according to a second embodiment of the present invention;

FIG. 6 is a block diagram of the norm constraint adaptive filters of the second embodiment;

FIG. 7 is a block diagram of the constraint coefficient generator used in FIG. 6;

FIG. 8 is a block diagram of an adaptive array beamformer according to a third embodiment of the present invention;

FIG. 9 is a block diagram of the coefficient-constrained adaptive filters of the third embodiment;

FIG. 10 is a graphic representation of the input/output characteristic of the limiters of FIG. 9;

FIG. 11 is a block diagram of an adaptive array beamformer according to a fourth embodiment of the present invention; and

FIG. 12 is a block diagram of a modification of the present invention.

DETAILED DESCRIPTION

Before proceeding with the detailed description of the present invention, it may provide helpful to provide an explanation of the prior art with reference to FIGS. 1 to 3. In FIG. 1, a linear array of microphones 1_0~1_{M-1} of identical operating characteristics are located at sufficient distances from signal sources of interest so that the wavefront of each signal at the microphones is considered to be linear. The microphones are connected to FIR transversal filters 20_0~20_{M-1} of a spatial lowpass filter 2, the outputs of the filters 20 being summed by an adder 26 to produce an output containing the target signal from a particular (assumed) direction and signals from other directions which are uncorrelated with the target signal. The outputs of filters 20 are applied through a timing adjustment delay circuit 3 to a subtractor 32 of a canceler 4.

The outputs of the M microphones are further connected to a spatial highpass filter bank 6 to produce (M−Q−1) output signals. The filter bank 6 operates so that the signals including the target signal as well as signals in the neighborhood of the assumed direction are rejected. The outputs of the filter bank 6 thus contain the undesired signals as dominant components. The outputs of filter bank 6 are fed through leads F_0~F_{M-Q} to leaky adaptive filters 30_0~30_{M-Q} of the canceler 4. Leaky adaptive filters 30 of the canceler detect undesired signals contained in the output signal of the beamformer at terminal 5 having a high correlation with the undesired signals detected by the spatial highpass filter 6 by adaptively updating their tap weight values using the output of the beamformer as a signal indicating the amount of correction error. The high correlation signals detected by the leaky adaptive filters 30 are combined by an adder 31 and fed to the subtractor 32 where it is subtracted from the time-coincident signal from spatial lowpass filter 2, whereby the undesired signals are canceled at the output terminal 5 of the beamformer.

Each of the filters 20 has a tapped delay line formed by delay elements 22_0~22_{G-2} forming (G−1) delay-line taps which are connected to corresponding tap weight multipliers 23 for respectively weighting the tap signals with particular tap weight coefficients supplied from a tap weight memory 24 (where G is equal to or greater than 2), the weighted tap signals being summed by an adder 25 and fed to the adder 26. The tap weight memory 24 of each filter 20 stores a set of tap weight coefficients whose values are determined so that filters 20 exhibit particular characteristics which result in an output containing the target signal. If the assumed direction is normal to the length of the microphone array, the integer G=2 is used and the tap weight coefficient of the multiplier 23_0 is set equal to "1". Other design approaches are described in "Multidimensional Digital Signal Processing", Prentice-Hall, Inc. pages 289–315, 1984 and IEEE, Proceedings of International Conference on Acoustics, Speech and Signal Processing 93, pages 169–172.

Spatial highpass filter bank 6 of the type described in Document 2 is shown in FIG. 2. Filter bank 6 is made up of(M−Q−1) groups 40 of Q highpass filters 41 each, and an adder 42, which each group forming a spatial highpass filter, where Q is equal to or greater than "3". Each spatial filter 40 receives a selected set of the microphone outputs such that the signals from the microphones positioned closer to the center of the array are coupled to an increasing number of filters 41. Thus, the signals incident on the center area of the microphone array fare filtered through a greater number of filters 41 than the signals incident on the edges of the array are. Highpass filters 41 are basically of the same transversal filter configuration as the filters 20, but with different delay line lengths (G) and different filter characteristics.

The characteristics of the highpass filters 41 of filter bank 6 are those of a rejection filter wherein a group of signals propagating in the assumed direction are rejected at the output of adder 42 of each spatial highpass filter 40. A basic design method for this type of spatial filter is described in Document 2. One important consideration is the degree of design freedom which is determined by the number of microphones used. For an M-microphone array, it is represented by M−Q+1. With the use of a large number of microphones a beamformer having a wide rejection angle with high attenuation can be implemented. Advantageously, the target signal can be rejected in the interference path of such beamformers even though the assumed direction differs from its actual arrival direction.

In each of the leaky adaptive filters 30 (FIG. 3), a corresponding output signal from the filter bank 5 is successively shifted through delay-line taps formed by delay elements 50_0~50_{L-2} and the tap signals are weighted respectively by (L−1) multipliers 51 with tap weight coefficients supplied from update circuits 53_0~53_{L-1} and then summed by adder 52 for coupling to the adder 31. Each update circuit 53 operates in accordance with the least mean square (LMS) algorithm. The output of beamformer from subtractor 32, representing a correction error, is weighted by a stepsize μ in a multiplier 54 and applied to a multiplier 55 of each update circuit 53 for detecting a correlation between the weighted error and a corresponding tap signal. Each update circuit 53 includes a leaky integrator formed by an adder 56, a multiplier 57 and a delay element 58. The correlation output of multiplier 55 is summed with a feedback signal from multiplier 57 and delayed by a symbol interval by delay element 58. The delayed symbol is applied to the corresponding tap weight multiplier 51 as an updated tap weight coefficient as well as to the multiplier 57 where it is scaled down by a factor α (equal to less than unity) and fed back to the adder **56**. Because of this scale-down feedback, the integrator operates as a leaky integrator which differs from normal integrators where the scale factor is unity. The leaky integration prevents the tap weight coefficient from growing indefinitely when the target signal, when there is a leakage of the target signal to the interference path (i.e., the outputs of filter bank **6**) of the beamformer due to the inherent variability of microphone characteristics and positional errors of the microphones. Otherwise, the interference signals produced by the adaptive filters would become identical to the components of the signal in the main path of the beamformer, and the resulting cancellation would substantially remove the target signal.

However, in order to implement a beamformer having a wide range of rejection angles, a large number of microphones (the Q value) are required for each group of spatial highpass filters in the filter bank. If a sufficient number of microphones is not provided, the degree of design freedom must be sacrificed, resulting in a beamformer having a low noise cancelling capability.

Referring now to FIG. **4**, there is shown an adaptive array beamformer according to a first embodiment of the present invention in which parts corresponding to those of FIG. **1** are marked by the same numerals as those used in FIG. **1**, the description thereof being omitted for simplicity. The adaptive array beamformer of this embodiment comprises a spatial highpass filter **16** and a canceler **17**. Spatial highpass filters **16** includes M delay circuits 7_0~7_{M-1} connected respectively to the microphones 1_0~1_{M-1}, M leaky adaptive filters 8_0~8_{M-1} and M subtractors 9_0~9_{M-1} connected respectively to the outputs of the M delay circuits **7**.

The spatial lowpass filter **2**, connected to the microphone array, provides spatial lowpass filtering of the individual microphone signals and summing the lowpass-filtered signals in the same manner as in the prior art beamformer to detect the target signal. The output of the spatial lowpass filter **2** is applied to all the leaky adaptive filters **8** as a reference signal as well as to the delay **3**. The outputs the microphone array are passed through corresponding delay circuits **7** to subtractors **9** to which the outputs of leaky adaptive filters **8** are also supplied to be subtracted from the corresponding microphone outputs. The output of each subtractor **9** is coupled to the corresponding leaky adaptive filter **8** as an error signal to update their tap weight values. The M delay circuits **7** provide a delay to the microphone outputs so that they are time coincident at the inputs of corresponding subtractors **9** with the output signals of leaky adaptive filters **8**.

Each of the leaky adaptive filters **8** is identical in structure to that shown in FIG. **3**. Correlations between the reference signal and each of the error signals are detected by the leaky adaptive filters **8**. As described previously in connection with the prior art, the strength of a leaky adaptive filter for restraining the growth of tap weight is proportional to the magnitude of the tap weight value itself. As a result, if the optimum value for the tap weight coefficient (which minimizes the error input of the leaky adaptive filter) is relatively large, the tap weight value cannot converge to the optimum value, resulting in a substantial amount of error from the optimum value. This implies that depending on the tap weight value the correlation capability of the leaky adaptive filters **8** differs significantly. Therefore, those signal components, which require a greater tap weight value for enabling their correlation to be detected, cannot sufficiently be removed, while those signals requiring a lower tap weight value can be removed sufficiently.

With respect to the signal arriving in the assumed direction as well as to those arriving in near-assumed directions, the output of spatial lowpass filter **2** contains the same amount of such signal components as those detected by the microphone array, and the maximum tap weight value necessary for removing them from the interference path of the beamformer is as small as "1". The leaky adaptive filters **8** are therefore designed with a low maximum tap weight value so that the target signal components are completely removed at the outputs of subtractors **9**.

With respect to the interference signals, on the other hand, the output of the spatial lowpass filter **2** contains a smaller amount of interference signals than those detected by the microphone array. Therefore, the tap weight value necessary for the leaky adaptive filters **8** to remove the interference signals is much higher than "1". Thus, the amount of removal at the outputs of subtractors **9** is much less in the case of the interference signals than in the case of the target signal components. If normal adaptive filters are used instead of the leaky adaptive filters **9**, their tap weight values would be allowed to grow indefinitely, and as a result, not only the interference signals but the target signal components are removed.

Canceler **17** includes M leaky adaptive filters 10_0~10_{M-1} connected respectively to the outputs of corresponding subtractors **9** to receive the interference signals detected in a manner just described. Each of the leaky adaptive filters **10** is identical in characteristic to the prior art leaky adaptive filters. Although most of the target signal components are removed, there is still a small amount of their leakage at the outputs of subtractors **9**. Due to the adaptive leaky integration of filters **10**, the growth of their tap weight values due to the presence of such small amount of leakage of the target signal are restrained. The outputs of leaky adaptive filters **10** are summed by adder **11** and supplied to subtractor **12** for canceling the interference signals contained in the main path of the beamformer.

Since the output of each subtractor **9** contains only a small amount of the target signal, the latter is not canceled in the subtractor **12** even though there is a look-direction error.

the leaky adaptive filters **8** of the spatial highpass filter **16** operate in effect as variable spatial highpass filters. The degree of design freedom of the present embodiment is not less than that of Document 2 and a large look-direction error is allowed using a smaller number of microphones than in the case of Document 2.

A second embodiment of the present invention is shown in FIG. **5** in which parts corresponding to those in FIG. **4** are marked with the same numeral as those used in FIG. **4**. The beamformer of FIG. **5** differs from the first embodiment in that the leaky adaptive filters **10** of FIG. **4** are replaced with norm-constrained adaptive filters 13_0~13_{M-1}.

As shown in detail in FIG. **6**, each norm-constrained adaptive filter **13** comprises a tapped-delay line formed by delay elements 60_0~60_{L-2}, tap weight multipliers 61_0~61_{L-1} connected to the delay-line taps, and adder **62** for summing the weighted tap signals. Update circuits 63_0~63_{L-1} are provided which are connected to a constraint coefficient generator **69**. Each update circuit **63** receives an error signal from the output of the beamformer from subtractor **12** via multiplier **64** where it weighted by the stepsize μ. Correlation between the corresponding tap signal and the weighted error signal is taken by a multiplier **65** and summed by adder **66** with a tap weight value of a previous sample supplied from multiplier **67**. The output of delay element **68** is scaled down by multiplier **67** with a constraint control parameter β from the constraint coefficient generator **69**. To the constraint coefficient generator **69** is connected the output of multiplier **65** of each update circuit **63**. The output of the adder **66** is supplied to the constraint coefficient generator **69** as the output of the update circuit **63**.

Constraint coefficient generator **69** controls the constraint control parameter β such that the p-th power of norm L_p (where p is an integer equal to or greater than unity) of the tap weight coefficients does not exceed a positive integer Θ using the following Equation:

$$L_p = \sqrt[p]{\sum_{i=0}^{L-1} |w_i|^p} \tag{1}$$

where w_i is the tap weight coefficient at the i-th delay-line tap. By constraining the Lp value below the Θ-value, the growth of tap weights is restrained.

As shown in detail in FIG. 7, the constraint coefficient generator **69** includes a calculator **70** for calculating the p-th power of norms. This calculator is formed by a plurality of circuits 71_0~71_{L-1} for raising the corresponding outputs of the update circuits **63** to the p-th power. The outputs of the p-th power-raising circuits **71** are summed by an adder **72** and supplied to a circuit **73** where an Lp value is obtained by taking the inverse root of p-th power of the output of adder **72**. The value Lp is supplied to a division circuit **74** where it is used to divide the threshold value Θ. The output of the division circuit **74** is fed to a minimum selector **75** which compares it with the unity value and selects the smaller of the two and supplies it as a constraint control parameter β to all the update circuits **63**. When the Lp value exceeds the constant Θ, all the tap weight values are decreased so that Lp becomes smaller than Θ.

A third embodiment of the present invention is shown in FIG. 8 which is a further modification of the first embodiment. In this modification, a coefficient-constrained adaptive filter **14** is used instead of each leaky adaptive filter **8** of FIG. 4. As illustrated in detail in FIG. 9, each coefficient-constrained adaptive filter **14** has a memory **89** in which maximum tap weight values $ø_0$~$ø_{N-1}$ and minimum tap weight values ϕ_0~ϕ_{N-1} are stored for update circuits **83**. The reference signal from spatial lowpass filter **2** successively appears as tap signals along taps formed by delay elements **80** and multiplied in corresponding multipliers **81** with a tap weight coefficient supplied from corresponding update circuits **83** and summed by adder **82** where it is coupled to the corresponding subtractor **9**. The output of this subtractor **9** is weighted by the stepsize μ in multiplier **84** and supplied to multiplier **85** where it is multiplied with the corresponding tap signal, the output of multiplier **85** being summed in adder **86** with a previous tap weight value form a limiter **87** and supplied through delay element **88** to the limiter **87**.

Corresponding maximum and minimum tap weight values $ø_i$ and ϕi (i=0, 1, . . . N−1) form a pair and each maximum/minimum pair is supplied to corresponding update circuit $\mathbf{83}_i$ from memory **89**. As illustrated graphically in FIG. 10, the limiter **87** of each update circuit has a linear input/output characteristic for input values varying in the range between ϕ_n and $ø_n$ and a flat characteristic outside the range. The output of the limiter **87** varies linearly with its input as long as it is within the limit values and clamped to one of the limit values when the input falls outside of the range.

As shown in FIG. 11, a fourth embodiment of the present invention is implemented by the combination of the norm-constrained adaptive filters **13** of FIG. 5 and the coefficient-constrained adaptive filters **14** of FIG. 8.

A second spatial lowpass filter **15** may be provided as shown in FIG. 12. This second filter is connected to the microphone array to produce a signal which can be used as a reference signal, instead of using the output of the first spatial lowpass filter **2**, for the leaky adaptive filters **8** as well as for the coefficient-constrained adaptive filters **14** of the previous embodiments. In this embodiment, the first spatial lowpass filter **2** is designed to form a mainlobe of greater width in the assumed direction in comparison with the width of the mainlobe formed by spatial lowpass filter **15**. With the wider mainlobe of the first spatial lowpass filter **2**, the overall characteristic of the beamformer is fit to the characteristic of this filter. This arrangement is particularly useful when there is a large look-direction error.

What is claimed is:

1. An adaptive array beamformer comprising:

an array of spatially distributed sensors;

a spatial beamforming filter connected to said sensors for respectively filtering output signals of the sensors and summing the filtered output signals to produce a first filter output containing a target signal arriving at said array in a specified direction;

a plurality of first adaptive filters, each having a tapped-delay line connected to receive said first filter output, coefficient update means for producing tap weight coefficients indicating correlations between tap signals from the tapped-delay line and a first error signal applied thereto, a multiply-and-sum circuit for weighting said tap signals with said coefficients respectively and summing the weighted tap signals to produce a second filter output not containing said target signal, said coefficient update means including restraining means for preventing said coefficients from increasing indefinitely;

a plurality of first subtractors, each detecting a difference between an output signal of a corresponding one of said sensors and the second filter output of a corresponding one of said first adaptive filters and supplying the difference to the coefficient update means of the corresponding first adaptive filter as said first error signal;

a plurality of second adaptive filters, each having a tapped-delay line connected to receive said first error signal from a corresponding one of said first subtractors, coefficient update means for producing tap weight coefficients indicating correlations between tap signals from the tapped-delay line and a second error signal applied thereto, a multiply-and-sum circuit for weighting said tap signals with said coefficients respectively and summing the weighted tap signals to produce a third filter output, said coefficient update means including restraining means for preventing said coefficients from increasing indefinitely;

an adder for summing the third filter outputs from the second adaptive filters; and

a second subtractor for detecting a difference between the first filter output and a summed signal from said adder and supplying the difference to the coefficient update means of said second adaptive filters as said second error signal.

2. An adaptive array beamformer comprising:

an array of spatially distributed sensors;

a first spatial beamforming filter connected to said sensors for respectively filtering output signals of the sensors and summing the filtered output signals to produce a first filter output containing a target signal arriving at said array in a specified direction;

a second spatial beamforming filter connected to said sensors for respectively filtering output signals of the sensors and summing the filtered output signals to produce a second filter output containing said target signal, said second spatial beamforming filter having a greater beam width than a beam width of the first spatial beamforming filter;

a plurality of first adaptive filters, each having a tapped-delay line connected to receive said second filter output coefficient update means for producing tap weight coefficients indicating correlations between tap signals form the tapped-delay line and a first error signal applied thereto, a multiply-and-sum circuit for weighting said tap signals with said coefficients respectively and summing the weighted tap signals to produce a third filter output not containing said target signal, said coefficient update means including restraining means for preventing said coefficients from increasing indefinitely;

a plurality of first subtractors, each detecting a difference between an output signal of a corresponding one of said sensors and the third filter output of a corresponding one of said first adaptive filters and supplying the difference to the coefficient update means of the corresponding first adaptive filter as said first error signal;

a plurality of second adaptive filters, each having a tapped-delay line connected to receive said first error signal from a corresponding one of said first subtractors, coefficient update means for producing a tap weight coefficients indicating correlations between tap signals from the tapped-delay line and a second error signal applied thereto, a multiply-and-sum circuit for weighting said tap signals with said coefficients respectively and summing the weighted tap signals to produce a fourth filter output, said coefficient update means including restraining means for preventing said coefficients from increasing indefinitely;

an adder for summing the third filter outputs from the second adaptive filters; and

a second subtractor for detecting a difference between the first filter output and a summed signal from said adder and supplying the difference to the coefficient update means of said second adaptive filters as said second error signal.

3. An adaptive array beamformer as claimed in claim 1 or 2, wherein the restraining means of said first adaptive filters comprises a leaky integrator.

4. An adaptive array beamformer as claimed in claim 1 or 2, wherein the restraining means of said first adaptive filters comprises a limiter having a linear input/output characteristic between predetermined maximum and minimum values.

5. An adaptive array beamformer as claimed in claim 1 or 2, wherein the restraining means of said second adaptive filters comprises a leaky integrator.

6. An adaptive array beamformer as claimed in claim 1 or 2, wherein the restraining means of said second adaptive filters comprises a norm constraining means.

7. An adaptive array beamformer as claimed in claim 3, wherein the restraining means of said second adaptive filters comprises a leaky integrator.

8. An adaptive array beamformer as claimed in claim 4, wherein the restraining means of said second adaptive filters comprises a leaky integrator.

9. An adaptive array beamformer as claimed in claim 3, wherein the restraining means of said second adaptive filters comprises a norm constraining means.

10. An adaptive array beamformer as claimed in claim 4, wherein the restraining means of said second adaptive filters comprises a norm constraining means.

* * * * *

United States Patent [19]
Miller

[11] **Patent Number:** **4,965,834**
[45] **Date of Patent:** **Oct. 23, 1990**

[54] **MULTI-STAGE NOISE-REDUCING SYSTEM**

[75] Inventor: **Harry B. Miller**, Niantic, Conn.

[73] Assignee: **The United States of America as represented by the Secretary of the Navy**, Washington, D.C.

[21] Appl. No.: **328,651**

[22] Filed: **Mar. 20, 1989**

[51] **Int. Cl.**^{5} **H04B 15/00**
[52] **U.S. Cl.** **381/94**
[58] **Field of Search** 381/94, 71; 367/123

[56] **References Cited**

U.S. PATENT DOCUMENTS

4,589,137	5/1986	Miller	381/94
4,649,505	3/1987	Zinser, Jr. et al.	381/71

OTHER PUBLICATIONS

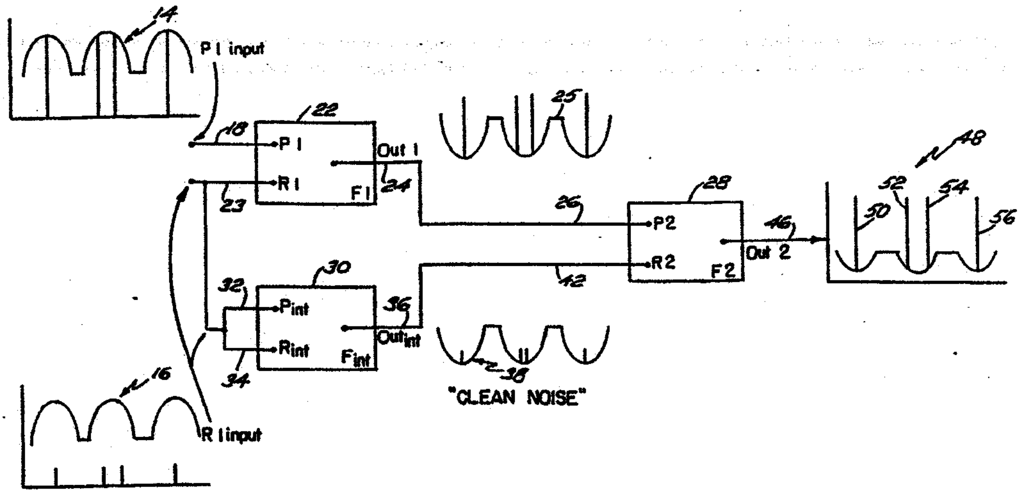
Heinone, et al., FIR-Median Hybrid Filters, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP-35, No. 6, Jun., 1987.

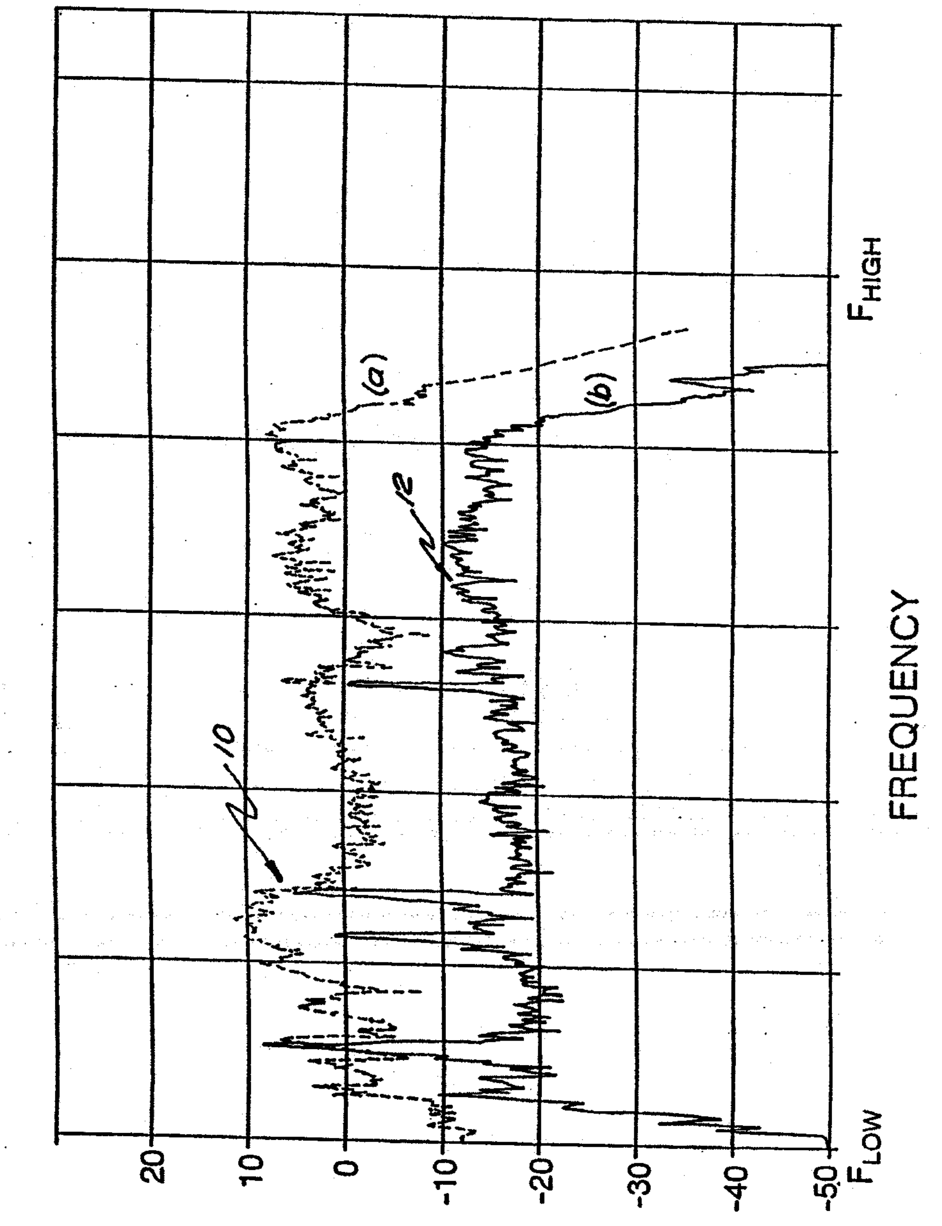
Primary Examiner—Forester W. Isen
Attorney, Agent, or Firm—Michael J. McGowan; Prithvi C. Lall

[57] **ABSTRACT**

An electronic noise-reducing system which includes a plurality of adaptive filters forming multiple stages of noise reduction and producing greatly increased signal-to-noise ratio. The input for the primary channel of the first adaptive filter, which forms the first noise-reducing stage, is the signal including multitones buried in noise. The reference channel ideally uses signal-free noise as input. The output of the first adaptive filter is used as the input to the primary channel of the second or final adaptive filter, whereas the reference channel thereof is fed with "clean noise". The clean noise can be obtained as the output of the intermediate adaptive filter by feeding simultaneously both the primary and reference channels of the intermediate filter with the noise-reduced waveform present at the output of the first noise-reducing filter.

8 Claims, 8 Drawing Sheets





F I G. I

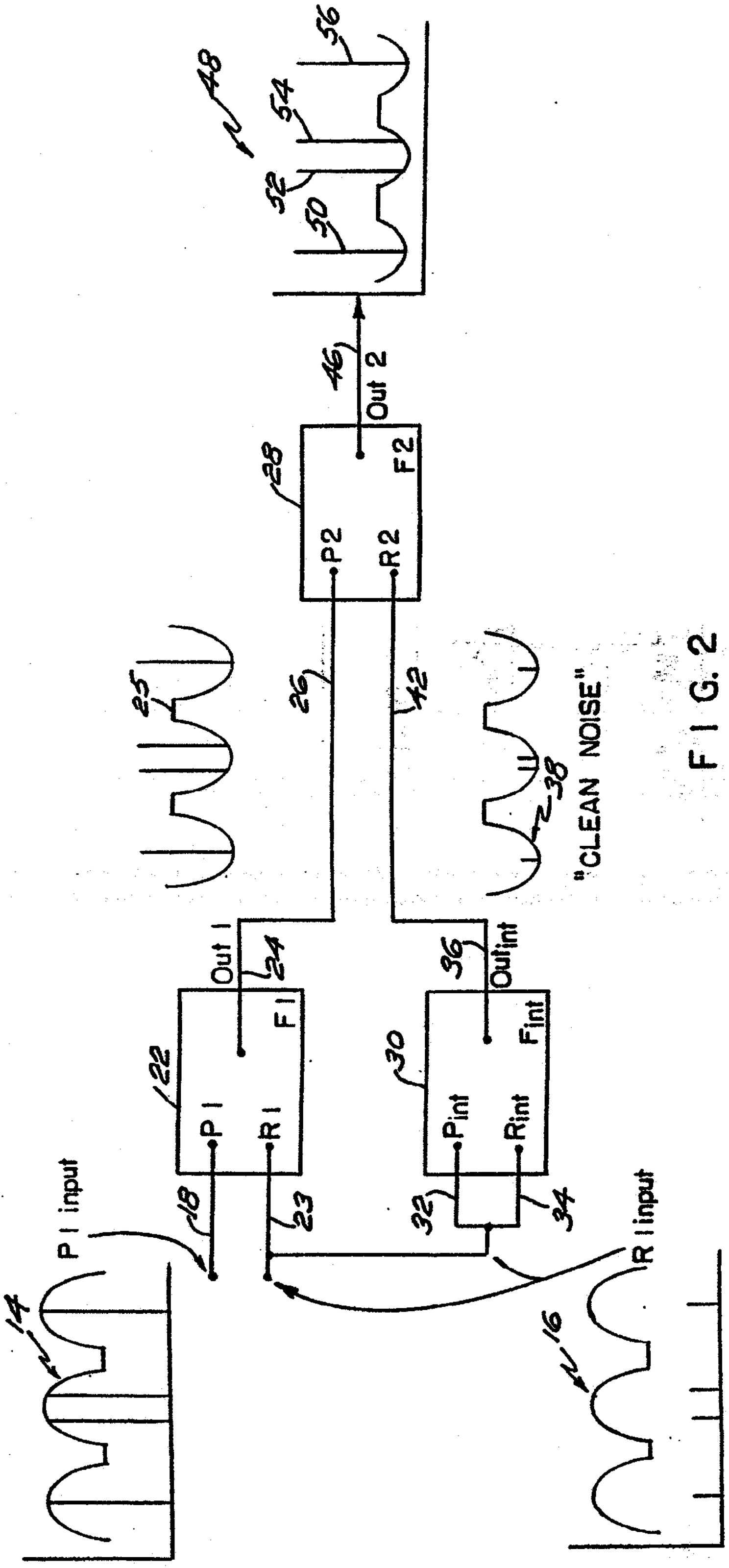
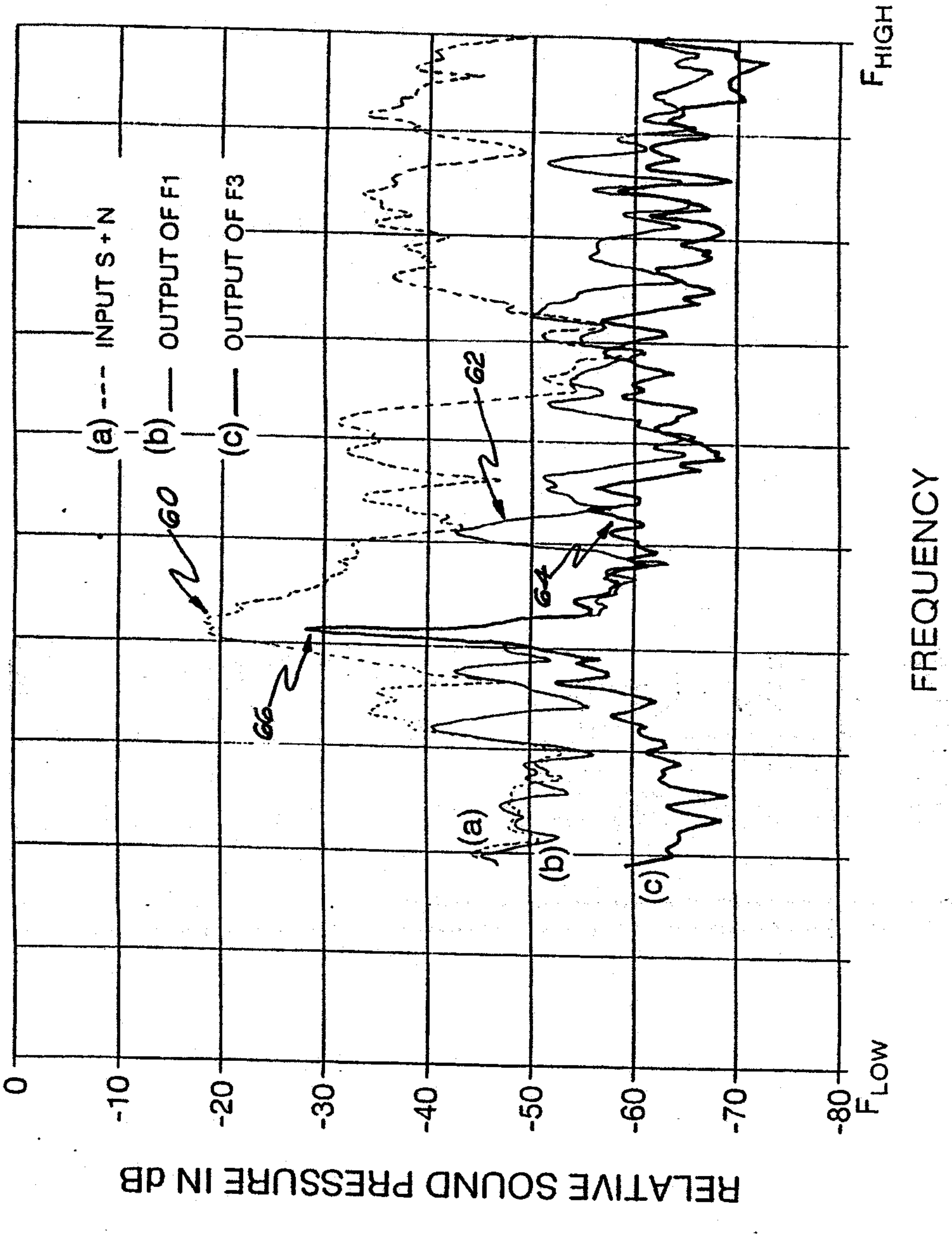
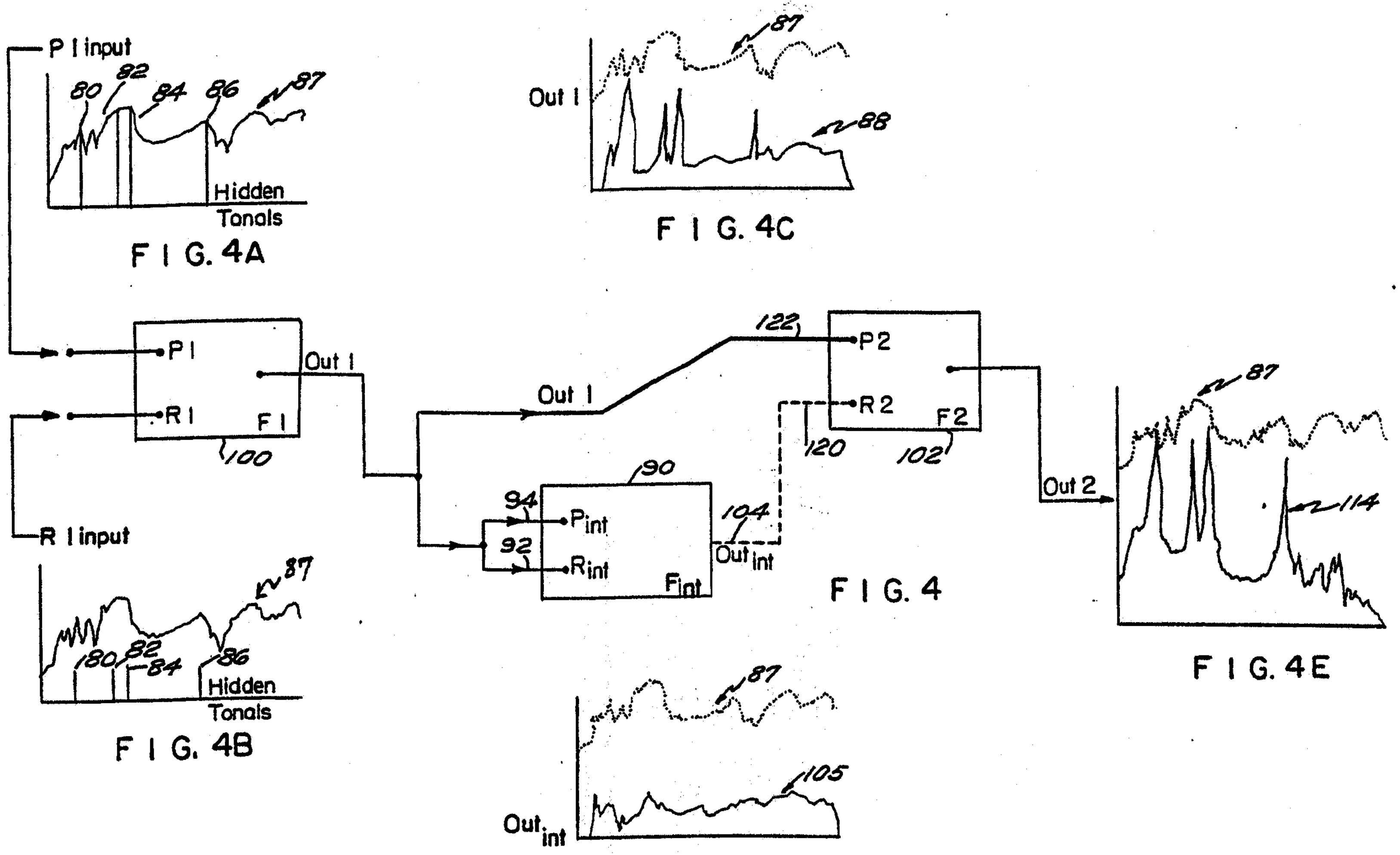

48
56
54
52
50
46
Out 2
28
P2
R2
F2
25
26
42
38
"CLEAN NOISE"
24
Out 1
22
P1
R1
F1
30
36
Out$_{int}$
P$_{int}$
R$_{int}$
F$_{int}$
18
23
32
34
P1 input
R1 input
14
16

FIG. 2


(a) --- INPUT S + N
(b) —— OUTPUT OF F1
(c) —— OUTPUT OF F3
60
62
64
66
(a)
(b)
(c)
0
-10
-20
-30
-40
-50
-60
-70
-80
RELATIVE SOUND PRESSURE IN dB
F_{LOW}
F_{HIGH}
FREQUENCY

F I G. 3



F I G. 4A

F I G. 4B

F I G. 4C

F I G. 4

F I G. 4D

F I G. 4E

United States Patent [19]

Miller

[11] **Patent Number: 4,965,834**

[45] **Date of Patent: Oct. 23, 1990**

[54] **MULTI-STAGE NOISE-REDUCING SYSTEM**

[75] Inventor: **Harry B. Miller**, Niantic, Conn.

[73] Assignee: **The United States of America as represented by the Secretary of the Navy**, Washington, D.C.

[21] Appl. No.: **328,651**

[22] Filed: **Mar. 20, 1989**

[51] **Int. Cl.^{5}** **H04B 15/00**

[52] **U.S. Cl.** **381/94**

[58] **Field of Search** 381/94, 71; 367/123

[56] **References Cited**

U.S. PATENT DOCUMENTS

4,589,137 5/1986 Miller 381/94

4,649,505 3/1987 Zinser, Jr. et al. 381/71

OTHER PUBLICATIONS

Heinone, et al., FIR–Median Hybrid Filters, IEEE Transactions on Acoustics, Speech, and Signal Processing, vol. ASSP–35, No. 6, Jun., 1987.

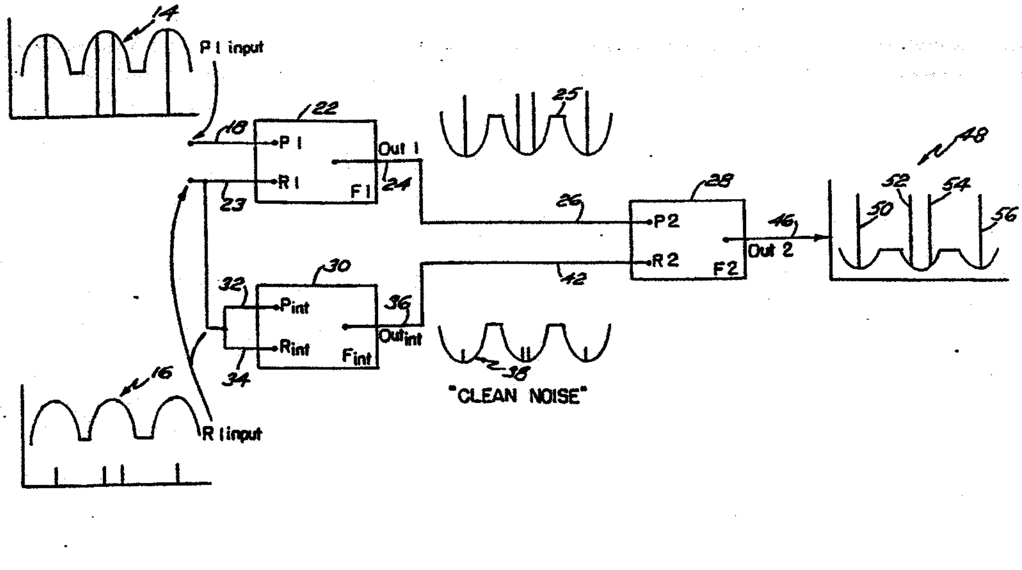
Primary Examiner—Forester W. Isen

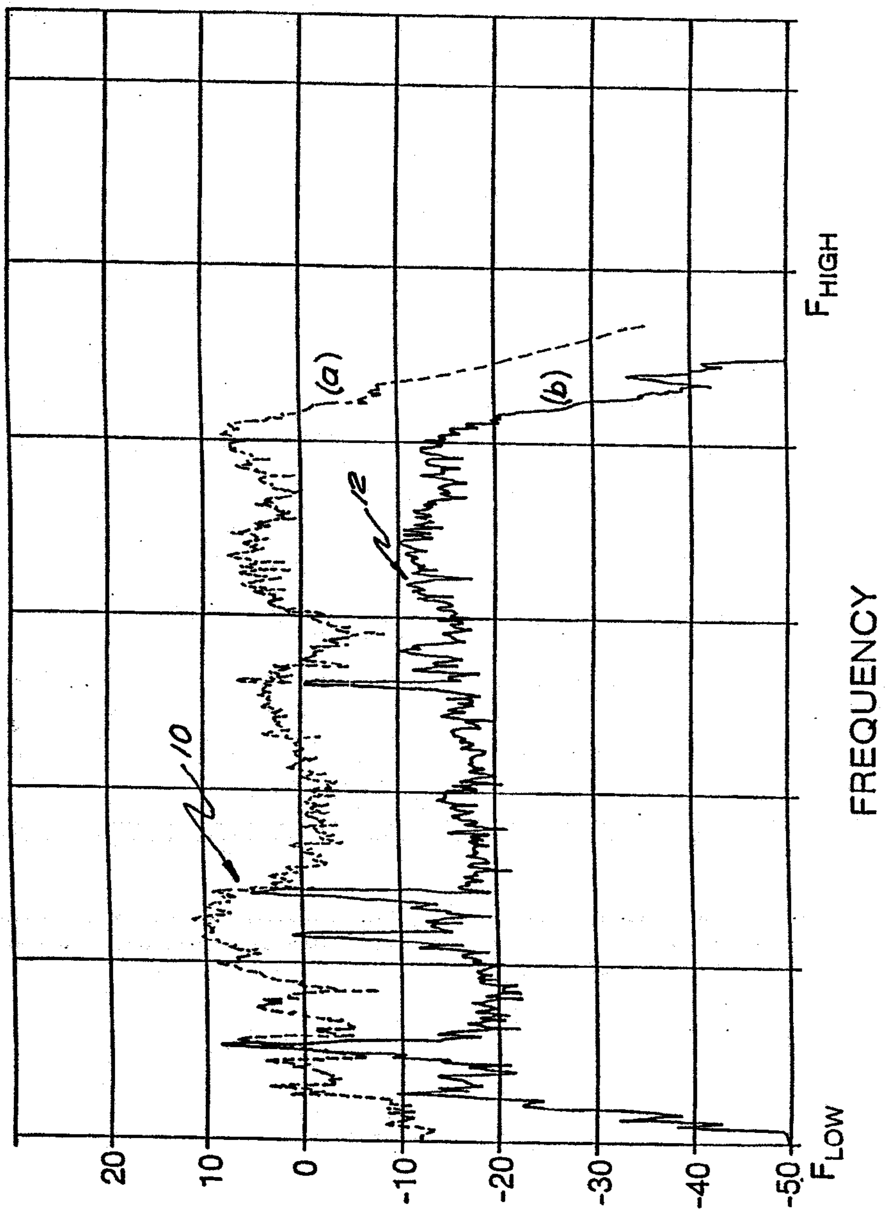
Attorney, Agent, or Firm—Michael J. McGowan; Prithvi C. Lall

[57] **ABSTRACT**

An electronic noise-reducing system which includes a plurality of adaptive filters forming multiple stages of noise reduction and producing greatly increased signal-to-noise ratio. The input for the primary channel of the first adaptive filter, which forms the first noise-reducing stage, is the signal including multitones buried in noise. The reference channel ideally uses signal-free noise as input. The output of the first adaptive filter is used as the input to the primary channel of the second or final adaptive filter, whereas the reference channel thereof is fed with "clean noise". The clean noise can be obtained as the output of the intermediate adaptive filter by feeding simultaneously both the primary and reference channels of the intermediate filter with the noise-reduced waveform present at the output of the first noise-reducing filter.

8 Claims, 8 Drawing Sheets





F I G. I

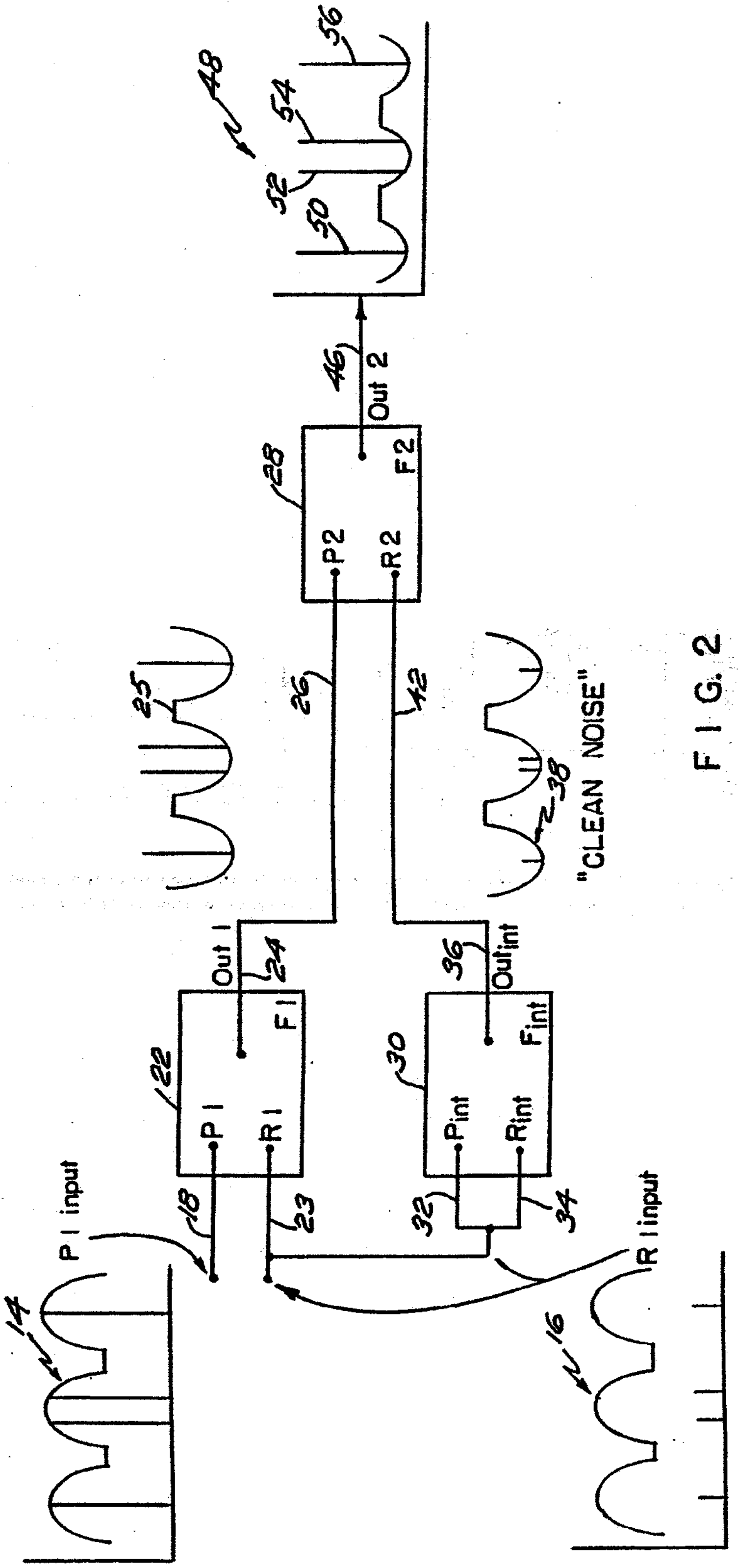
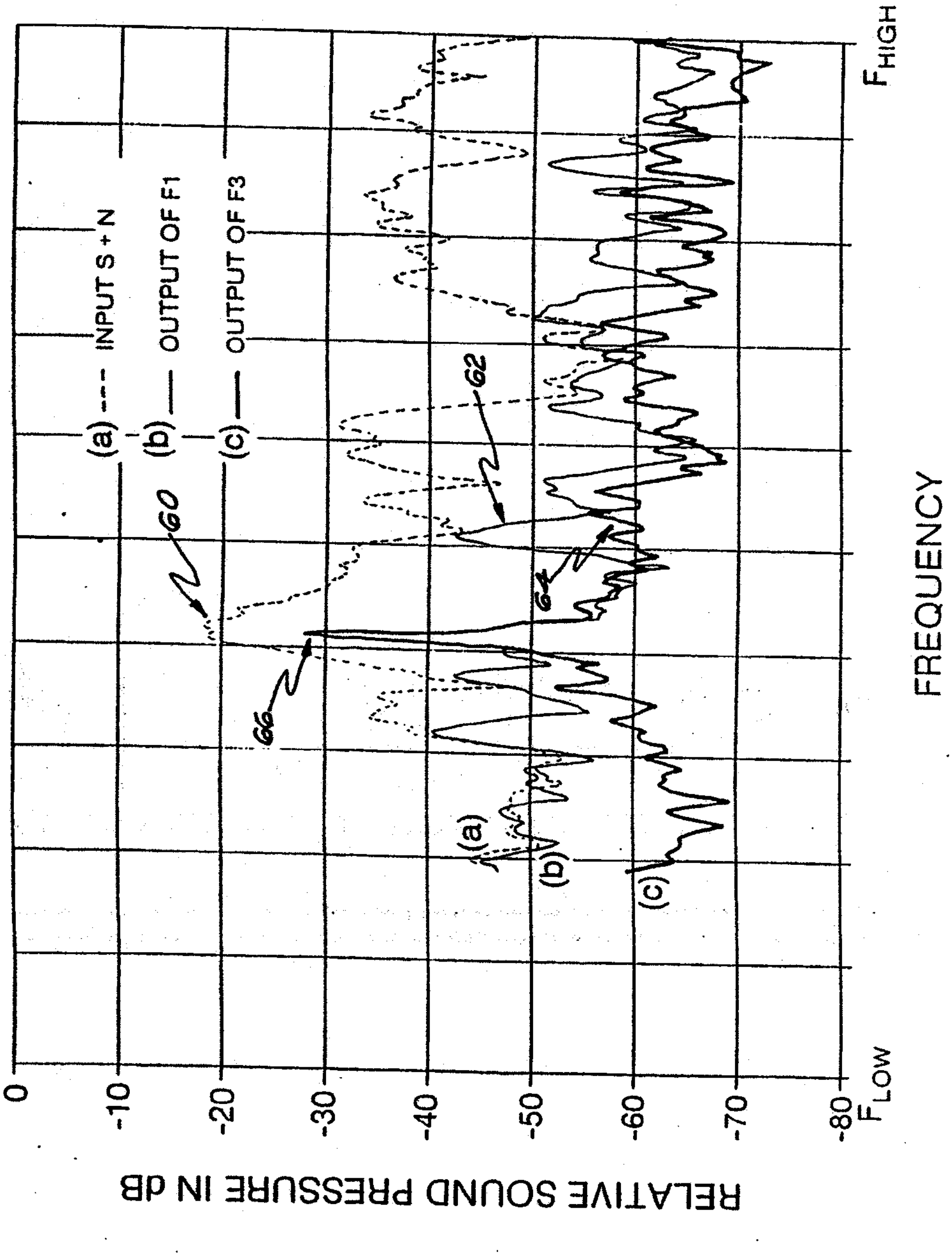

P1 input
14
18
P1
R1
22
F1
Out 1
24
23
25
26
28
P2
R2
F2
Out 2
46
48
50
52
54
56
42
38
"CLEAN NOISE"
30
P_{int}
R_{int}
F_{int}
Out_{int}
36
32
34
R1 input
16

FIG. 2


0
-10
-20
-30
-40
-50
-60
-70
-80
F_{LOW}
F_{HIGH}
RELATIVE SOUND PRESSURE IN dB
FREQUENCY
(a) --- INPUT S + N
(b) —— OUTPUT OF F1
(c) —— OUTPUT OF F3
60
62
64
66
(a)
(b)
(c)

F I G. 3

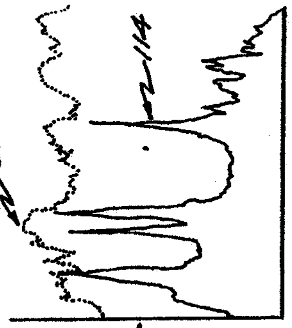


FIG. 4E

FIG. 4C

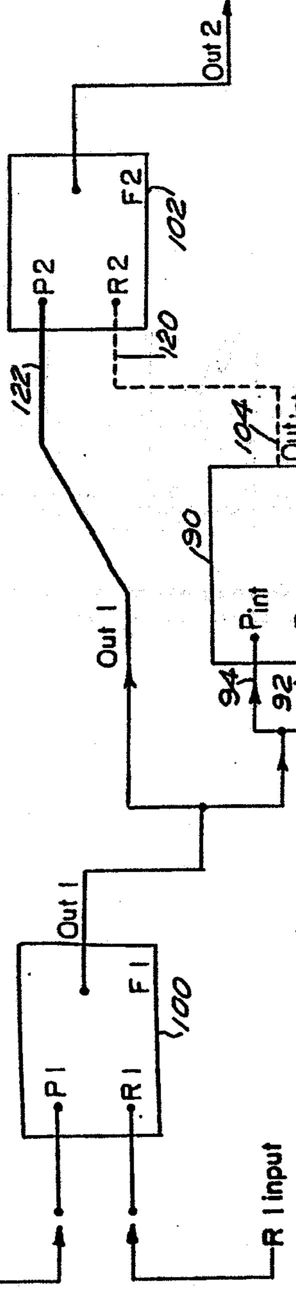


FIG. 4

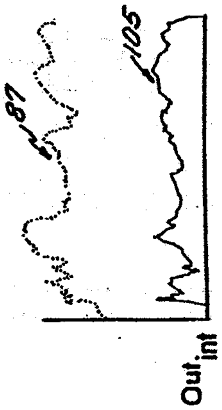


FIG. 4D

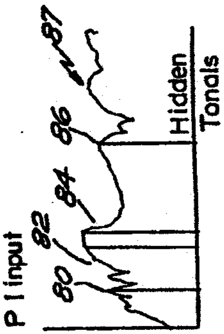


FIG. 4A

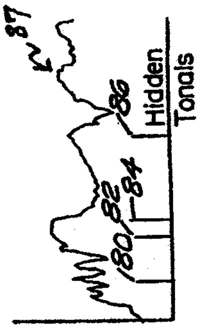
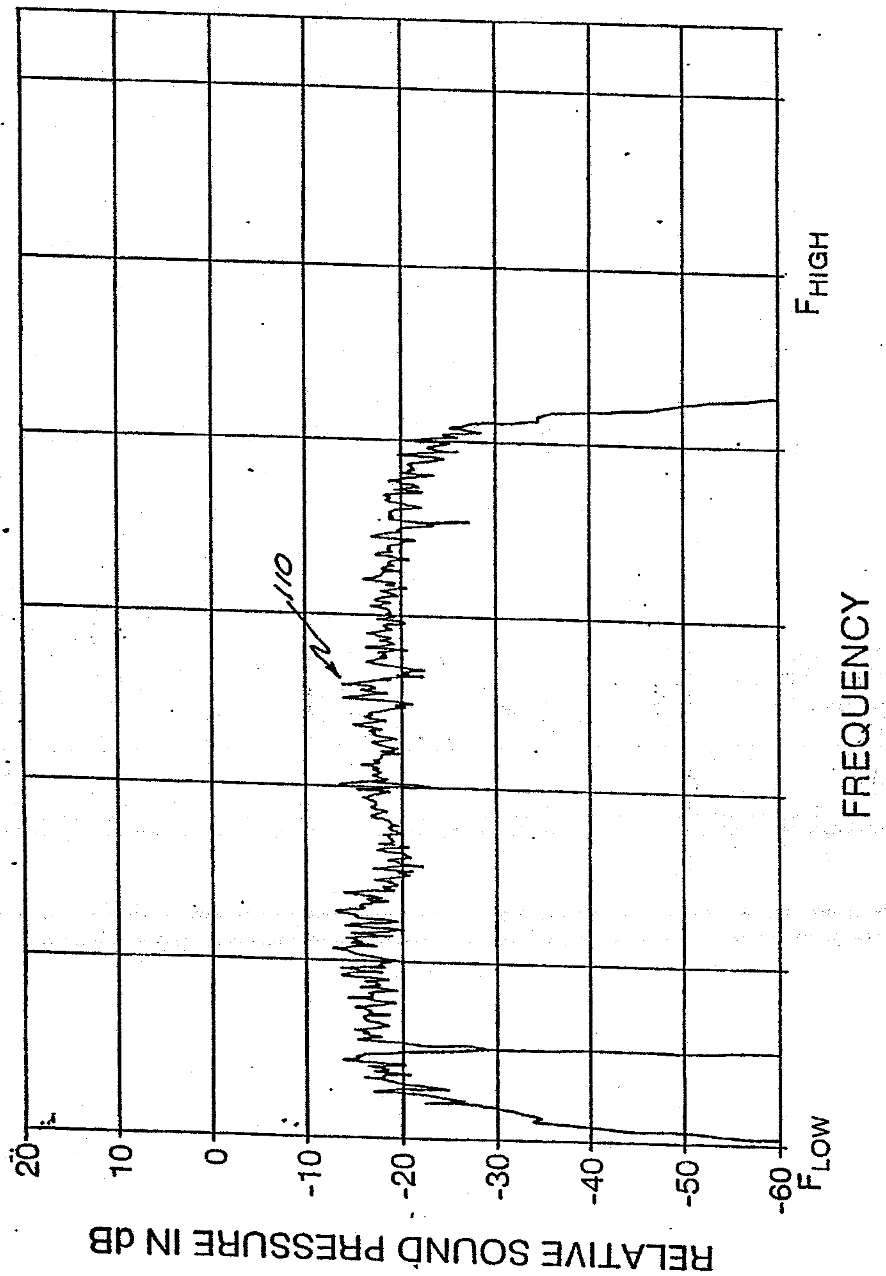
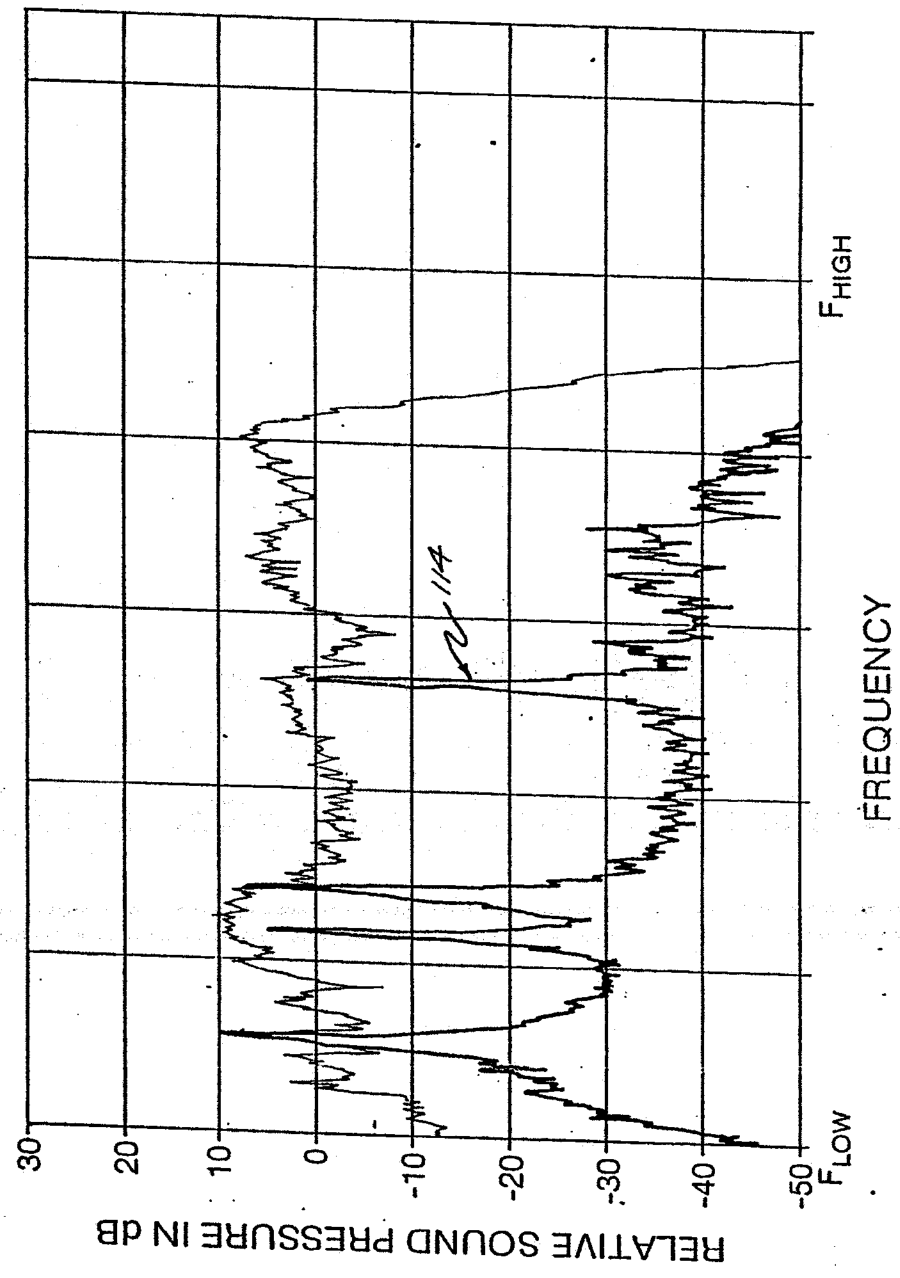


FIG. 4B


RELATIVE SOUND PRESSURE IN dB
20
10
0
-10
-20
-30
-40
-50
-60
F_{LOW}
F_{HIGH}
FREQUENCY
110

F I G. 5


30
20
10
0
-10
-20
-30
-40
-50
F_{LOW}
F_{HIGH}
RELATIVE SOUND PRESSURE IN dB
FREQUENCY
114

F I G. 6

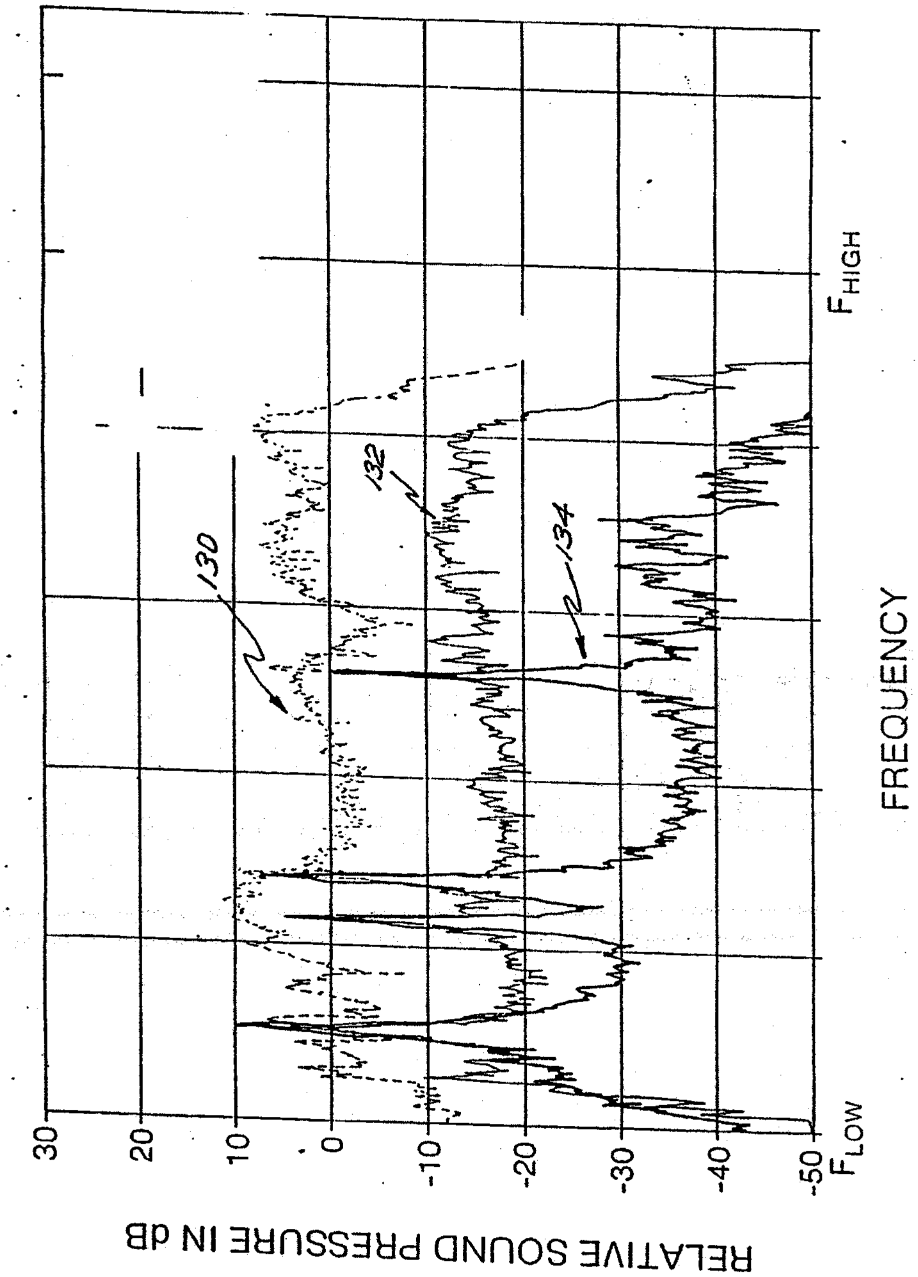

30
20
10
0
-10
-20
-30
-40
-50
F LOW
F HIGH
130
132
134
FREQUENCY
RELATIVE SOUND PRESSURE IN dB

FIG. 7

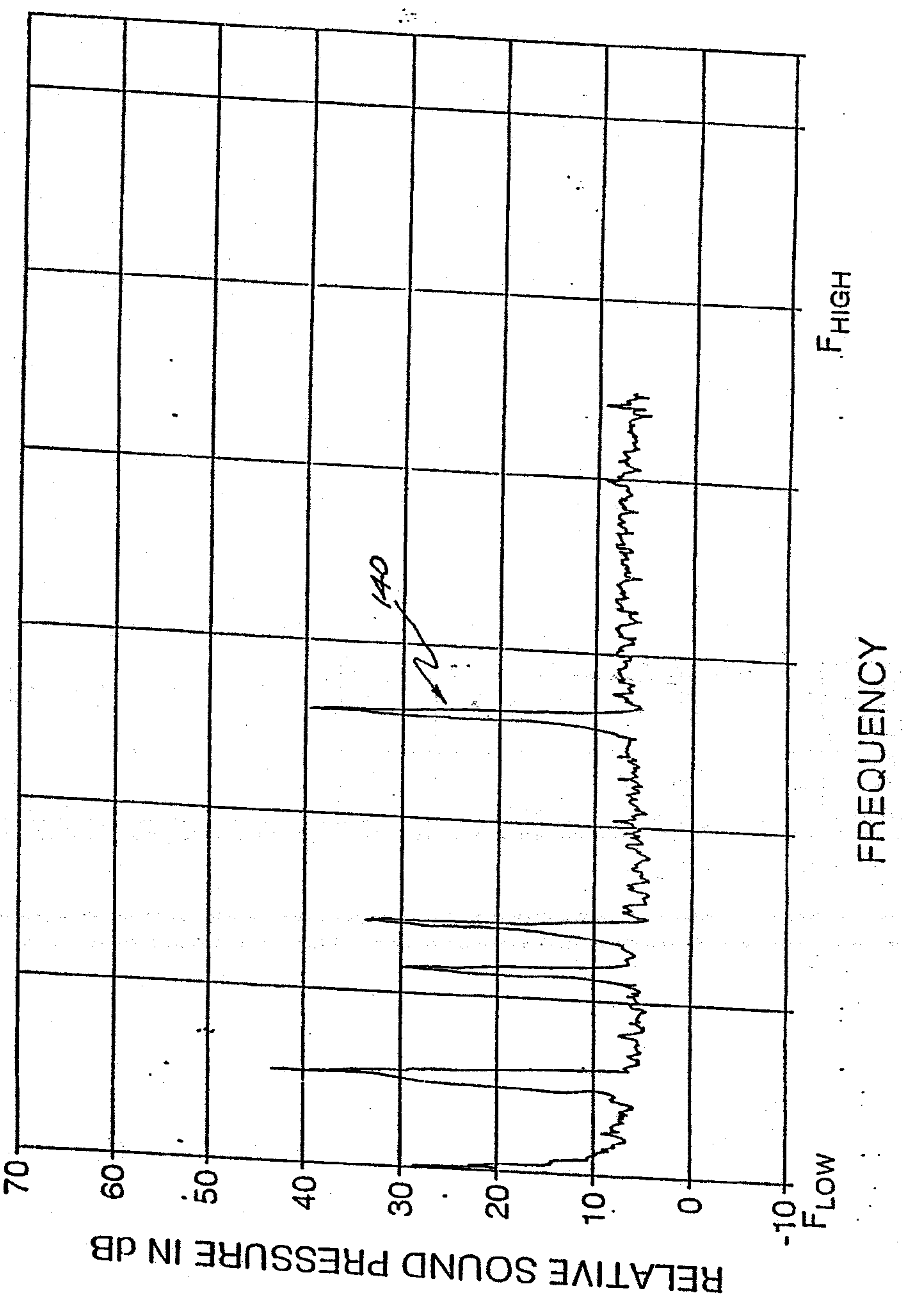

70
60
50
40
30
20
10
0
-10
RELATIVE SOUND PRESSURE IN dB
F_{LOW}
F_{HIGH}
FREQUENCY
140

FIG. 8

MULTI-STAGE NOISE-REDUCING SYSTEM

STATEMENT OF GOVERNMENT INTEREST

The invention described herein may be manufactured and used by or for the Government of the United States of America for governmental purposes without the payment of any royalties thereon or therefore.

BACKGROUND OF THE INVENTION

(1) Field of the Invention

Subject invention is related to signal processing in general and more particularly to a multi-stage system using adaptive filters for canceling noise without affecting the signal and thereby increasing signal-to-noise ratio, i.e., S/N.

(2) Description of the Prior Art

In my U.S. Pat. No. 4,589,137 and my pending patent application, Ser. No. 220,692, filed July 5, 1988 which are both incorporated here in entirety, one-stage noise-reducing systems were discussed wherein a single tone's S/N ratio was increased 17 dB by causing the noise in nearby frequency bands to attenuated by 17 dB. Extension of this work resulted in the "unmasking" of four tones (masked by broadband noise) at normalized frequencies of, roughly: 1, 2, 3 and 5 spread over a decade. The original masking noise was reduced over more than a decade of frequency by anywhere from 15 dB to 25 dB.

The adaptive filter using the Least Mean Squares (LMS) algorithm favors noise peak region's and tends not to favor noise dip-regions, so that the 15 dB attenuation occurred at dip regions while the 25 dB attenuation occurred at peak regions. A reasonable number would be 20 dB for the average attenuation of noise attained across a broadband. However, 20 dB attenuation is not sufficient for many applications, and so an attempt to cascade two or more stages of adaptive filters seemed worthwhile in order to try to attain 35 dB or 40 dB of attenuation.

SUMMARY OF THE INVENTION

We start with the output of a one-stage noise-canceling system. A reduction of about 20 dB in broadband noise over a band greater than a decade was accomplished routinely, using either a time-domain adaptive filter or a frequency-domain adaptive filter. But this noise floor, which we will arbitrarily call −20 dB, would not drop lower. In addition, if the original noise spectrum had a fairly sharp dip somewhere, this was ignored by the adaptive filter so that it became a residual peak, which we named a stalagmite. So the goal of the present invention was to start with the output O_1 of a first adaptive filter F_1 and to feed it in tandem into a second adaptive filter F_2, with the assistance of a third or "intermediate" adaptive filter F_{int}, and thereby lower the noise floor by perhaps an additional 13 dB, thus lowering the noise overall to −33 dB (−20 dB and −13 dB), all without greatly attenuating the N tones already unmasked in the output of the first filter.

An object of subject invention is to have a noise-canceling system which does not require a large volume of sound-absorbing material.

Another object of subject invention is to have a noise-canceling system which reduces the noise over a wide frequency bandwidth.

Still another object of subject invention is to have a noise-canceling system which uses multiple adaptive filters in order to obtain larger overall noise reduction.

Other objects, advantages and novel features of the invention may become apparent from the following detailed description of the invention when considered in conjunction with the accompanying drawings wherein:

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a graphical representation of the signal-plus-noise (S+N) input wave and the signal-plus-noise output wave of one adaptive filter, to be used as an input to the next adaptive filter over a given frequency range.

FIG. 2 is a representation of typical inputs to the primary and reference channels of the first adaptive filter and primary and reference channels of an intermediate adaptive filter and a second adaptive filter, together with the improved output from the second filter.

FIG. 3 is a graphical representation, using experimental data, of the input and output of the first adaptive filter as compared with the output of the second or final adaptive filter.

FIGS. 4 and 4A–4E show the block diagram and signal profiles for the decorrelation method of noise reduction using subject invention.

FIG. 5 shows the "clean noise" output of the intermediate adaptive filter F_{int} as a function of frequency.

FIG. 6 shows the input sound wave (signal-plus-noise) to the first filter, and the output sound wave from the final or second filter over the given frequency range.

FIG. 7 shows the input (signal-plus-noise) to the first adaptive filter, output of the first adaptive filter to be used as input to the second adaptive filter, and output of the second filter over the given frequency range; and

FIG. 8 is the normalized output of the final or second adaptive filter over the given frequency range.

DESCRIPTION OF THE PREFERRED EMBODIMENT

It should be noted that throughout our discussion each of the adaptive filters has a primary (P) channel and a reference (R) channel with subscripts designating the adaptive filter under discussion. The problem in a two-stage noise canceler is: what to feed into the reference channel R_2 of adaptive filter F_2. The input to the primary channel P_2 of adaptive filter F_2 should be the output of F_1. The desired input to reference channel R_2, which is always the output of an adaptive filter called intermediate adaptive filter F_{int}, would be a near-duplicate to the output of adaptive filter F_1 except with the N tones removed, so that the reference channel could truly be said to contain only signal-free noise. We will refer to this as "clean noise". FIG. 1 shows the amplitude of sound pressure over a frequency range between f_{low} and f_{high} including both signal and wide band noise, the input sound pressure wave (signal+noise) being represented by curve **10** and the output of the first adaptive filter which is also the input to the second filter being represented by curve **12**.

A few methods will be described, which were used to lower the noise floor and/or to remove the stalagmites.

a. Mirror-Image Method: It had been observed that any adaptive filter such as F_1 did not "go after" the whole band of noise, designated by **14** in FIG. 2, simultaneously, but rather that it worked on the peak regions first, and the dip regions later. This tends to create a mirror-image of **14**, the input **18**'s (P_1) noise, at the output O_1 or **24** of F_1 (disregarding the signal for the moment), as shown in FIG. 2. Note that the tones survive undiminished. The output at O_1, namely sound pressure wave **25**, then feeds into the input P_2 (**26**) of a second adaptive filter F_2 (**28**).

Simultaneously the "signal-free noise" or "noise plus residual signal" designated as **16** at reference input **23** (R_1) of adaptive filter F_1 (**22**) as shown in FIG. 2, feeds also into an adaptive filter called $F_{intermediate}$ or **30** (F_{int}). This noise **16** feeds into both input channels **32** (P_{int}) and **34** (R_{int}), as shown in FIG. 2, via a tee connection.

To define a term called "partial convergence" which we will use presently, we first define another term, "full convergence", as a term used to describe the action from an adaptive filter when it has canceled noise as much as possible. If full convergence is aborted, we call the process "partial convergence". When partial convergence is used in F_{int}, the output at **36** ($output_{int}$), namely **38**, tends to be a mirror-image of the input **16** at **34** (R_{int} input) as seen in FIG. 2 and thus is almost identical with sound pressure wave **25** at output **24** (O_1) of F_1 (except for the virtual absence of signal components). We call the sound pressure wave **38** "clean noise".

If now the sound pressure wave **38** at $output_{int}$ **36** is fed into reference input or channel **42** (R_2) of adaptive filter **28** (F_2), any stalagmites existing in sound pressure wave **25** at Output **24** (O_1) will cancel at appreciably at Output **46** (O_2) of adaptive filter **28** (F_2). The output at **46** has a wave form **48** where tones **50, 52, 54** and **56** are unattenuated and the stalagmites **58** are low. Additionally some of the residual noise will cancel further, across the whole band. This is also shown in FIG. 3, using experimental data, wherein curve **60** shows a noise-spectrum plus a hidden tone, i.e., input S+N. Curve **62** shows the results of a first cancellation (observe the stalagmites on either side of the tone **66** of FIG. 3). Curve **64** shows the results of a second cancellation, where the stalagmites are much reduced. Since the mirror-image method works best for an "unwhitened" noise spectrum, it is advisable to "pre-unwhiten" the noise spectrum, via a spectral shaper of both magnitude and phase, into a first set of mountains and valleys feeding into a first noise-reduction system; and simultaneously "pre-unwhiten" the noise spectrum into a second set of mountains and valleys staggered or offset from the first set.

b. The Noise-Decorrelation Method. In the mirror-image method the "clean noise" was generated in the intermediate adaptive filter by supplying its P_{int} and R_{int} inputs (via a tee connection) with the same raw input that was used in filter F_1's R_1 channel as shown in FIG. 2. In the noise-decorrelation method, as shown in FIG. 4, the "clean noise" curve **105** of FIG. 4D is generated in the intermediate filter F_1 or **90** by supplying its P_{int} and R_{int} inputs (via a tee connection) with the noise-reduced $Output_1$, namely curve **88**, of filter F_1 (**100**) as shown in FIG. 4. It should be noted that curve **87** represents the input signal-plus-noise as shown in FIGS. 4A-4E. The signal, e.g., the four tones which exist superposed on the noise, as seen in the solid curve **88** of FIG. 4C, entering **90** (F_{int}) must be removed in order to produce "clean noise" at the output of adaptive filter **90** (F_{int}). This can be done by one of the following methods.

1. A delay of say 20 msec can be inserted within the reference channel **92** (R_{int}). This shifts the signal and the noise (in the reference channel) by 20 msec, in the time domain, enough to decorrelate the noise from its counterpart in P_{int} (**94**), but with no effect on a repetitive signal, e.g., a tone, which keeps repeating its time-domain signature. Full convergence is allowed to take place in all three adaptive filters **90, 100** and **102**. Only the original tone peaks subtract because only they are still correlated. They decrease to small values. The spectrum at $Output_{int}$ (**104**) is then called "clean noise" and is shown as curve **105** of FIG. 4D and again as curve **110** of FIG. 5. The final adaptive filter **102** (F_2) receives the "clean noise" at input R_2 and receives the original output (O_1) at input P_2. A second cancellation then occurs within adaptive filter **102** (F_2). The result is curve **114**, as shown in FIG. 6.

2. Alternatively, a delay of say 20 msec can be inserted in the primary channel **94** (P_{int}) of the intermediate filter **90** (F_{int}) as seen in FIG. 4. Full convergence must be aborted, since otherwise the filter **90** (F_{int}) will slowly cancel everything that is residing within channel **94** (P_{int}). Partial convergence of F_{int} must be used, with a duration time of, for example, only 4 seconds and then the convergence being frozen. Filters **100** (F_1) and **102** (F_2), however, are meanwhile allowed to fully converge, and then run continuously. The original tone peaks in channel **92** (R_{int}) disappear by subtraction because they are still highly correlated with their counterparts in channel **94** (P_{int}). The spectrum **105** from $output_{int}$ (**104**) is again called "clean noise". The final adaptive filter **102** (F_2) receives the "clean noise" at input channel **120** (R_2), and receives the noise-reduced wave **88** from $Output_1$ at input channel **122** input (P_2). A second cancellation then occurs within **102** (F_2). The result is the same as shown in FIG. 6, where the noise floor has dropped by almost an additional 20 dB, to a level of −40 dB. Recapitulating the events, a signal-plus-noise input, the output of the first adaptive filter, and the combination of first stage and second stage cancellation are shown in FIG. 7 as curves **130, 132** and **134** respectively.

3. A different method of achieving "clean noise" is to send the noise-reduced spectrum of FIG. 1 through a thresholding device which clips the magnitude of each spectral peak down to that of the neighboring noise level. That portion of the spectrum which fails to be clipped is preserved, and used as the "clean noise" input to R_2 of a second adaptive filter **102** (F_2). This method is especially useful when the "surviving spectrum" is nearly flat, like white noise, as seen in FIG. 5.

In all these methods, the "clean noise" goes to R_2 of adaptive filter **102** (F_2), while the $Output_1$ of filter F_1 goes to P_2 of the second filter, and the resultant second cancellation at $Output_2$ is indicated as curve **114** of FIGS. 4(*e*) and 6.

In each of the three noise cancellation methods discussed, a third stage of cancellation can be cascaded by adding two additional adaptive filters after adaptive filter F_2, giving a total of five adaptive filters. And for N stages of cancellation, the number of adaptive filters required is 2N−1. However, a law of diminishing returns shows up. For, although the noise floor seems to drop an additional 6 or 7 dB with three stages of cancellation, a new digital noise arises from the signal processing itself, making the usefulness of multi-stage cancellation doubtful for three or more stages.

Another advantage of a three-filter method is displayed in FIG. 8 wherein curve **140** shows the relative frequency response of four tones after a single noise-cancellation. Basically the autospectrum at the output of first adaptive filter **100** (F_1) is mathematically divided by the "clean noise" autospectrum at the output of adaptive filter (F_{int}); but since we are using logarithmic units, namely dB, we subtract (not divide) the two autospectra. Notice the straightened-out baseline. The three-filter method also allows other 2-channel comparisons to be made such as cross-correlation and coherence.

Thus a multiple stage noise-cancelling system according to the teachings of subject invention comprises a first adaptive filter and a plurality of pairs of adaptive filters, each stage requiring one pair of adaptive filters (wherein each adaptive filter includes one primary channel and one reference channel). Thus each stage of noise cancellation is cascaded by using an additional pair of adaptive filters wherein "clean noise" becomes the input to the reference channel of the second filter. The noise cancellation of the two successive stages is logarithmically additive.

Many modifications and variations of the presently disclosed invention are possible in light of the above teachings. As an example, the number of stages in the noise cancellation system can be varied without deviating from the teachings of subject invention. The number of signal tones buried in the noise may vary. Furthermore, the frequency range over which the signal tones are distributed may also vary. It is therefore understood that within the scope of the appended claims, the invention may be practiced otherwise than as specifically described.

What is claimed is:

1. An electronic noise-reducing system, wherein the noise comes from a nearfield noise source, utilizing a plurality of adaptive filters, including a final filter, fed by at least two sensors, namely a reference sensor comprising at least two electroacoustic elements, and an omnidirectional primary sensor, wherein at least one electroacoustic element of the reference sensor is used both in the reference sensor and simultaneously in the primary sensor; and wherein only the output of the final filter is used as the output of the system.

2. The electronic noise-reducing system of claim 1 wherein said plurality of adaptive filters used is equal to 2N−1 where N is an integer having values of 2, 3, 4, . . . respectively representing second, third, fourth, fifth, . . . stages of noise-reduction and each stage thereof using a primary channel and a reference channel.

3. The electronic noise-reducing system of claim 2 having said plurality of adaptive filters to be three (with N=2) and thus including two noise-reduction stages wherein the first adaptive filter delivers a noise-reduced wave at the output thereof, the intermediate adaptive filter delivers "clean noise" as output thereof, and the final adaptive filter delivers an improved noise-reduced output with the tones having an increased signal-to-noise ratio.

4. The electronic noise-reducing system as in claim 2 wherein each of said second, third, fourth, fifth, . . . stages of noise reduction uses "clean noise" as input to the primary and reference channels of the intermediate adaptive filters.

5. The electronic noise-reducing system of claim 4 wherein said plurality of adaptive filters are least mean square (LMS) adaptive filters.

6. The electronic noise-reducing system of claim 4 wherein the electroacoustic element of said reference sensor used simultaneously as the primary sensor is the element closest to the nearfield noise source.

7. The electronic noise-reducing system of claim 6, wherein said reference sensor comprising at least two electroacoustic elements is a line microphone having its axis positioned at an angle to a plane in which said nearfield noise source lies.

8. The electronic noise-reducing system of claim 7 wherein the directional reference sensor comprising at least two electroacoustic elements is a line microphone with axis thereof perpendicular to the plane of said nearfield noise source.

* * * * *

US005226016A

United States Patent [19]

Christman

[11] **Patent Number: 5,226,016**

[45] **Date of Patent: Jul. 6, 1993**

[54] **ADAPTIVELY FORMED SIGNAL-FREE REFERENCE SYSTEM**

[75] Inventor: **Russel A. Christman**, Old Lyme, Conn.

[73] Assignee: **The United States of America as represented by the Secretary of the Navy**, Washington, D.C.

[21] Appl. No.: **872,263**

[22] Filed: **Apr. 16, 1992**

[51] **Int. Cl.**^{5} **H04R 27/00**

[52] **U.S. Cl.** **367/135**; 367/901; 381/94; 381/71

[58] **Field of Search** 367/901, 124, 136, 135; 381/94, 71; 364/574

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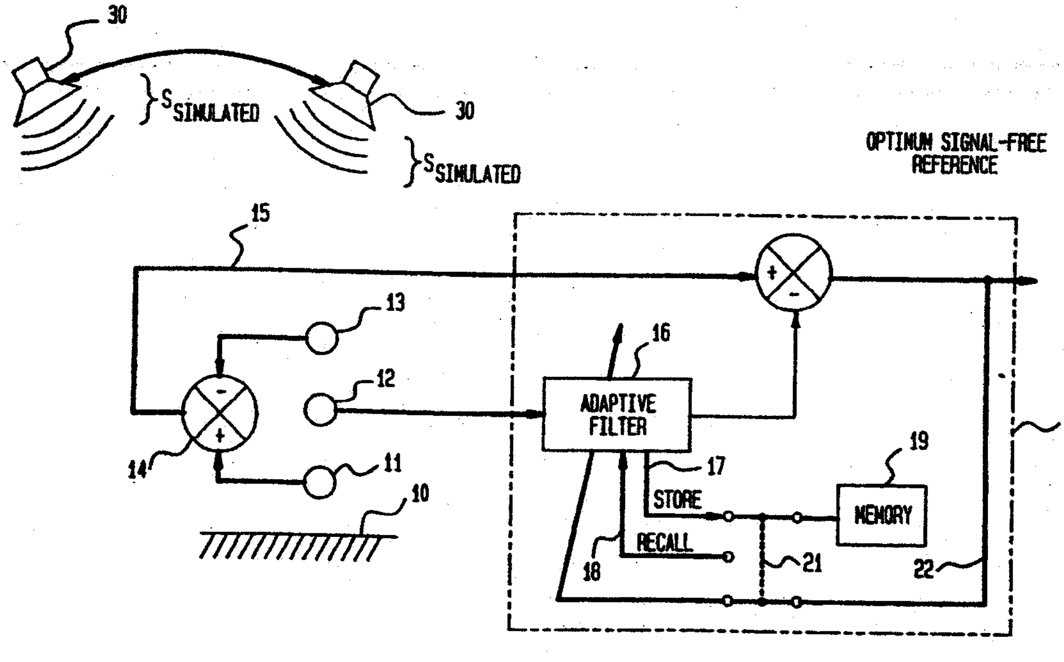
Primary Examiner—Daniel T. Pihulic
Attorney, Agent, or Firm—Michael J. McGowan; Prithvi C. Lall; Michael F. Oglo

[57] **ABSTRACT**

A method and apparatus are provided to adaptively form an optimum signal-free reference used to cancel near-field noise in an adaptive plate-noise cancellation system. First, second and third pressure sensors are positioned to detect any near-field plate-radiated noise. The first sensor is positioned closest to the plate, the third sensor is positioned furthest from the plate, and the second sensor is positioned between the first and third sensors. An acoustic far-field projector generates a plurality of broadband signals from a plurality of incidence angles. Each broadband signal is projected at an amplitude indicative of a noise-off condition. Outputs from the first and third sensors are combined to form a dipole responsive to each broadband signal). A feedback system is operatively associated with the dipole and the second sensor. The feedback system includes an adaptive filter that converges to generate an optimum signal-free reference for each of the plurality of the broadband signals and stores filter coefficients indicative of the optimum signal-free reference for each of the broadband signals. Switching means are provided to selectively switch the adaptive filter out of the feedback system such that the stored filter coefficients may be used.

6 Claims, 2 Drawing Sheets



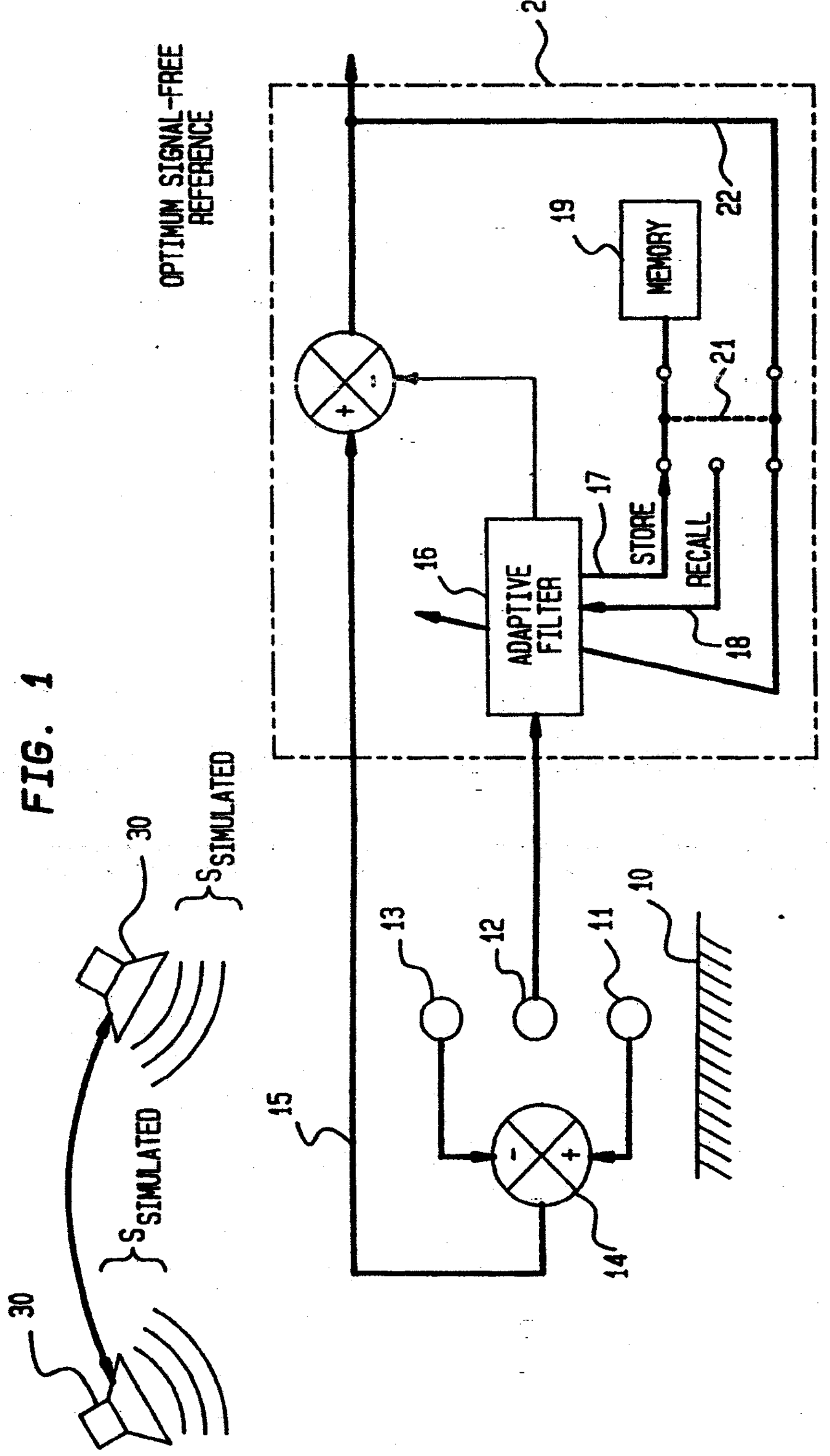
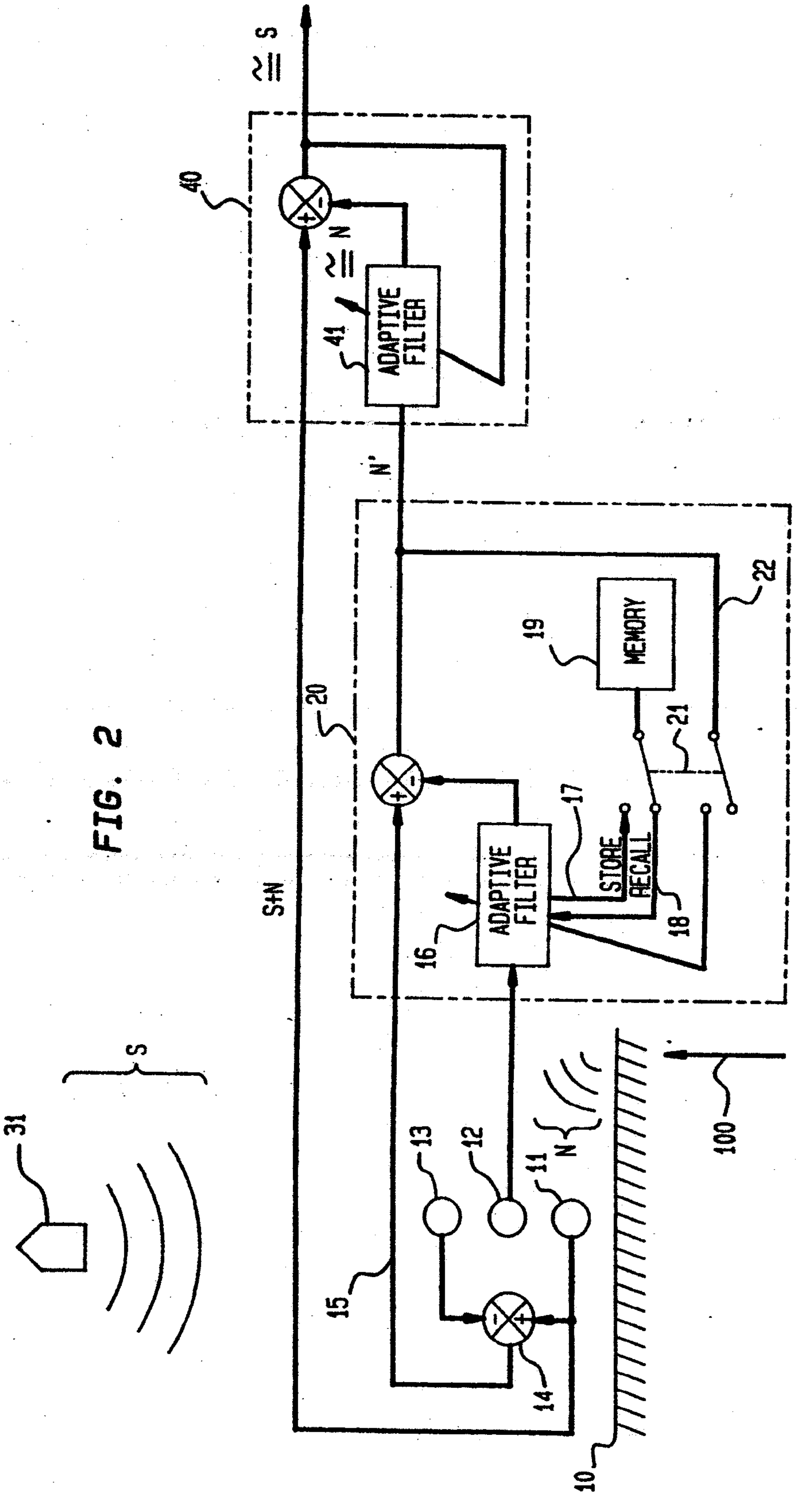

OPTIMUM SIGNAL-FREE REFERENCE
20
22
19
MEMORY
21
17
STORE
RECALL
18
16
ADAPTIVE FILTER
+
-
13
12
11
10
15
14
30
$S_{SIMULATED}$
$S_{SIMULATED}$
30

FIG. 1

FIG. 2


≅S
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ADAPTIVELY FORMED SIGNAL-FREE REFERENCE SYSTEM

STATEMENT OF GOVERNMENT INTEREST

The invention described herein may be manufactured and used by or for the Government of the United States of America for Governmental purposes without the payment of any royalties thereon or therefor.

BACKGROUND OF THE INVENTION

(1) Field of the Invention

The present invention relates generally to adaptive signal processing, and more particularly to an adaptive signal processing system for forming an optimum signal-free reference for ultimate use in an adaptive hull-radiated noise cancellation system.

(2) Description of the Prior Art

Adaptive signal processing methods have been developed for the cancellation of submarine sonar hull-radiated noise in the near-field region of the submarine hull. The procedure typically utilizes three hydrophone pressure sensors aligned normal to the hull. The hydrophones are positioned within the near-field of any hull noise that may be generated. This is a region where the noise decays exponentially with respect to distance from the hull. It is well defined by the material properties of the hull structure and frequency range of interest as is well known in the art.

The pressure measured by the hydrophones contains both target signals and unwanted hull-radiated noise components. The adaptive signal processing procedure is based on the circuitry developed by Widrow for noise cancellation. See "Adaptive Noise Canceling: Principles and Applications," by Widrow et al., Proceedings IEEE, Volume 63, No. 12, pp. 1692–1716, December 1975. Such circuitry requires a sensor which measures both signal and noise and is referred to as the primary sensor. More importantly, a secondary input, referred to as the reference, requires a sensor that measures noise only and must therefore be "signal-free". This reference input is filtered adaptively by using the Least Mean Square (LMS) algorithm which attempts to produce an output that is a replica of the noise on the primary input. The subtraction of the filtered reference replica from the primary input then provides the cancellation of noise. A "signal-free" reference is thus an essential requirement for an effective adaptive noise cancellation system. If any portion of the signal is present on the reference channel, the signal as well as noise may be canceled adaptively. This would reduce the effectiveness of the adaptive noise cancellation system as well as any other systems that are required for post-processing of signals.

One method of generating the "signal-free" reference is disclosed by Miller in U.S. Pat. No. 4,589,137, issued May 13, 1986. Miller teaches the use of a three element line of hydrophones, or tripole, as the reference channel input source. The signal-free reference is arrived at by trial-and-error adjustment of a plurality of preamplifiers and phase shifters. Adjustment is required for each of twenty or more narrow frequency bins in order to achieve a wide band signal-free reference. This process is time consuming, is prone to operator error and typically lacks sufficient frequency resolution since the time required to generate each (frequency dependent) signal-free reference limits the number of frequencies processed.

SUMMARY OF THE INVENTION

Accordingly, it is an object of the present invention to provide a method and apparatus that generates a signal-free reference for ultimate use in an adaptive noise cancellation system.

Another object of the present invention is to provide method and apparatus that generates optimum signal-free references over a wide frequency band of interest.

Still another object of the present invention is to provide a method and apparatus that generates optimum signal-free references over a wide frequency band of interest quickly and with minimal operator involvement.

Yet another object of the present invention is to provide a method and apparatus that generates an optimum signal-free reference used to cancel near-field, plate-radiated noise in an adaptive noise cancellation system.

Other objects and advantages of the present invention will become more obvious hereinafter in the specification and drawings.

In accordance with the present invention, a method and apparatus are provided for adaptively forming an optimum signal-free reference used to cancel near-field noise in an adaptive plate-noise cancellation system. First, second and third pressure sensors are aligned in a straight line normal to a plate and are in the near-field of any generated plate-noise. The first sensor is positioned closest to the plate, the third sensor is positioned furthest from the plate, and the second sensor is positioned between the first and third sensors. An acoustic farfield projector generates a plurality of broadband signals from a plurality of incidence angles. Each broadband signal is projected at an amplitude indicative of a noise-off condition. Outputs from the first and third sensors are combined to form a dipole responsive to each broadband signal. The second sensor is adaptively filtered then subtracted from the dipole to form the reference output. A feedback system is connected between this output and the adaptive filter. Using outputs from both second sensor and feedback system, the adaptive filter converges to generate an optimum signal-free reference for each of the plurality of the broadband signals and stores filter coefficients indicative of the optimum signal-free reference for each of the broadband signals. Switching means are provided to selectively switch the adaptive filter out of the feedback system such that the stored filter coefficients may be recalled as needed.

BRIEF DESCRIPTION OF THE DRAWING(S)

FIG. 1 is a block diagram of the apparatus used to form the signal-free reference according to the present invention; and

FIG. 2 is a block diagram of an adaptive hull-noise cancellation system using the signal-free reference apparatus of FIG. 1.

DESCRIPTION OF THE PREFERRED EMBODIMENT(s)

Referring now to the drawings, and more particularly to FIG. 1, a block diagram is shown of the apparatus used to form a signal-free reference according to the present invention. The method and apparatus of the present invention will be described simultaneously as they pertain to a hull-radiated noise cancellation system for a ship or submarine. However, as will be readily apparent to one skilled in the art, the method and apparatus of the present invention are adaptable to any noise cancellation system in which wall or plate-radiated noise is a concern.

In FIG. 1, three hydrophones 11, 12 and 13 are mounted above a section of the ship's hull plating 10. Hydrophones 11, 12 and 13 are aligned with one another and are normal to hull plating 10. Since it is near-field hull-radiated noise that is of concern, all these hydrophones must lie within the near-field noise region of hull plating 10. (While hull-radiated noise is comprised of near-field and far-field components, they are uniquely different and as a result are not correlated.) The near-field positioning and normal alignment of the three hydrophones provide a high degree of noise correlation among the hydrophones. This is an essential requirement for effective adaptive noise cancellation. The distance from hull plating 10 at which the near-field dominates can be determined based on hull plating material properties and size as is well known in the art.

The spacing between each hydrophone is not rigidly constrained. Theoretically, any spacing greater than zero will work as long as the hydrophones remain within the near-field. However, practically speaking, very small spacing may result in low level uncorrelated hydrophone-to-hydrophone electronic noise which can dominate and degrade performance of the system. Accordingly, typical spacing ranges from 2.5 to 7.5 centimeters. Further, it is not a requirement of the present invention that hydrophones 11, 12 and 13 be evenly spaced as is necessary in prior art methods.

An acoustic far-field projector 30 is movably positioned as shown to project broadband target-like signals through the water towards hull plating 10 from a variety of incidence angles. The acoustic waves that arrive at hydrophones 11, 12 and 13 look exactly like that of a target, but are controlled in amplitude. Specifically, the amplitude is adjusted to be well above expected near-field, hull-noise levels resulting in an effective "noise off" condition. Amplitude adjustment of projector 30 is thus a factor of the amplitude of hull-noise during calibration. As a minimum, the calibration signal amplitude should be 10 times greater than the hull-noise amplitude.

In response to the signals from projector 30, outputs from hydrophones 11 and 13 are combined at a summer 14 thereby forming a dipole output 15. In response to the same signals, output from hydrophone 12 is provided as a reference input to a filter 16. Filter 16 is further part of an adaptive feedback system indicated by that portion of the block diagram within dotted line box 20.

Adaptive feedback system 20 is a conventional noise canceling circuit known in the art. When a double pole, double throw switch 21 is closed as shown, filter 16 acts as an adaptive filter that is responsive to an error signal on feedback line 22. Thus, when switch 21 is closed and hydrophones 11, 12 and 13 are subjected to the target-like signals $S_{simulated}$ from projector 30, filter 16 converges to a value that results in a minimum error voltage on line 22. This minimized error voltage is characteristic of any such cancellation circuit employing the Least Mean Square algorithm taught by Widrow. The minimized error voltage serves as the optimum signal-free reference since the signals from projector 30 simulate a "noise off" condition. This procedure is repeated for a variety of positions (i.e. incidence angles) of projector 30 until a desired granularity is achieved. The

coefficients used by filter 16 are stored in a memory 19 via path 17 (switch 21 closed) for the optimum signal-free references at all incidence angles. In other words, filter 16 has been trained to effectively cancel any incoming signal from projector 30.

Typically, the apparatus of the present invention used to adaptively form a signal-free reference is part of an adaptive hull-noise cancellation system shown in FIG. 2. Common elements share common reference numerals where appropriate. Once the desired granularity is achieved, switch 21 may be opened to essentially eliminate feedback line 22 from the adaptive feedback system 20. In operation, a far-field target 31 radiates acoustic waves S towards hydrophones 11, 12 and 13. Stored filter coefficients are recalled from memory 19 via path 18 (switch 21 opened) for optimum signal cancellation. At the same time, hull plating 10 is typically subjected to an excitation force, shown by arrow 100, due primarily to ship movement and/or engine noise. Excitation force 100 thus generates hull-radiated noise N in the near-field of hull plating 10. Hydrophone 11 nearest to hull plating 10 is the primary hydrophone for sensing both signal S and noise N and producing an output (S+N) indicative thereof as shown.

The weighted sum of hydrophones 11, 12 and 13, or optimum signal-free reference N', is somewhat modified with respect to the noise N. This is due to the amplitude and phase differences between hydrophone 11 and the weighted sum of all three hydrophones. Thus, the signal-free reference N' must be applied to an adaptive filter 41 which is part of another adaptive feedback loop 40. As is known in the art, adaptive filter 41 and feedback loop 40 work to minimize the error/adaptive output. Since the optimum signal-free reference N' is an approximation of the noise N, adaptive filter 41 converges to a multiplying factor of approximately N/N' to yield a filter output that is approximately equal to N. This approximation of N is then subtracted from the (S+N) output of hydrophone 11 to yield an output that is approximately equal to S.

The advantages of the present invention are numerous. By training a filter to output an optimum signal-free reference, an adaptive noise canceling system is able to extract the signal from a noisy environment. This extraction occurs without the loss of any signal since there is no excess signal on the signal-free reference. This allows a follow-on adaptive noise cancellation process to maintain the signal structure.

It will be understood that many additional changes in the details, materials, steps and arrangement of parts, which have been herein described and illustrated in order to explain the nature of the invention, may be made by those skilled in the art within the principle and scope of the invention as expressed in the appended claims.

What is claimed is:

1. A method of adaptively forming an optimum signal-free reference used to cancel near-field noise in an adaptive platenoise cancellation system, comprising the steps of:

aligning, in the near-field of any generated plate-noise, first, second and third pressure sensors in a straight line normal to a plate, wherein the first sensor is positioned closest to the plate, the third sensor is positioned furthest from the plate, and the second sensor is positioned between the first and third sensors;

subjecting the three sensors to a broadband signal from an acoustic far-field projector, the broadband signal having an amplitude effective to provide a noise-off condition;

combining outputs from the first and third sensors to form a dipole responsive to the broadband signal;

weighting, at an adaptive filter, an output from the second sensor responsive to the broadband signal;

generating an error signal as the difference between the output from the dipole and the weighted output from the second sensor; and

applying the error signal to the adaptive filter, wherein convergence of the adaptive filter is indicative of an optimum signal-free reference.

2. A method according to claim 1 further comprising the step of storing coefficients of the adaptive filter indicative of the optimum signal-free reference.

3. A method according to claim 1 wherein the broadband signal is successively propagated towards the three sensors from a plurality of incidence angles.

4. In an adaptive plate-noise cancellation system having first, second and third hydrophone pressure sensors aligned in a straight line normal to a plate, the first sensor being positioned closest to the plate, the third sensor being positioned furthest from the plate, and the second sensor being positioned between the first and third sensors, wherein all three sensors lie in the near-field of any generated plate- noise, a method of adaptively forming an optimum signal-free reference used to cancel near-field noise in the cancellation system comprising the steps of:

successively subjecting the three sensors to a plurality of broadband signals projected from a plurality of incidence angles, each broadband signal having an amplitude indicative of a noise-off condition;

combining outputs from the first and third sensors to form a dipole responsive to each of the broadband signals;

providing, in operative association with the dipole and the second sensor, a feedback system having an adaptive filter that converges to generate an optimum signal-free reference for each of the broadband signals; and

storing, for each of the broadband signals, filter coefficients of the adaptive filter indicative of the optimum signal-free reference.

5. An apparatus for adaptively forming an optimum signal-free reference used to cancel near-field noise in an adaptive platenoise cancellation system comprising:

first, second and third pressure sensors aligned in a straight line normal to a plate and in the near-field of any generated plate-noise, wherein the first sensor is positioned closest to the plate, the third sensor is positioned furthest from the plate, and the second sensor is positioned between the first and third sensors;

an acoustic far-field projector for generating a plurality of broadband signals projected from a plurality of incidence angles, each broadband signal being successively projected at an amplitude indicative of a noise-off condition;

means for combining outputs from the first and third sensors to form a dipole responsive to each broadband signal;

a feedback system, operationally associated with dipole and the second sensor, having an adaptive filter that converges to generate an optimum signal-free reference for each of the plurality of the broadband signals and stores filter coefficients indicative of the optimum signal-free reference for each of the broadband signals; and

means for selectively switching said filter out of said feedback system, wherein said adaptive filter uses the stored filter coefficients.

6. An apparatus as in claim 5 wherein the plate is a submarine hull and said first, second and third sensors are hydrophones.

* * * * *

#5

PATENT
43086.00.7007

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant(s) : Joseph Marash et al.
U.S. Patent No. : 6,049,607
Issued : April 11, 2000
Serial No. : 09/157,035
For : INTERFERENCE CANCELING METHOD AND APPARATUS
Filed : September 18, 1998
Examiner : SAINT SURIN, JACQUES M
Art Unit : 2747
Confirmation No. : 1884

1633 Broadway, 47th Floor,
New York, NY 10019

FILED VIA EFS-WEB
ON January 27, 2014

REQUEST FOR CERTIFICATE OF CORRECTION

Certificate of Correction Branch
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

Dear Sir:

It is requested that a Certificate of Correction be issued in the above-entitled patent in accordance with the accompanying form PTO 1050. Please make the following changes:

ON THE FACE OF THE PATENT:

Left column, field (73), Assignee, replace the text "**Lamar Signal Processing, Yokneam, Israel**" with "**Andrea Electronics Corporation, Melville, New York**".

REMARKS

The requested changes do not constitute new matter and this application does not require re-examination. A completed Form PTO 1050 is enclosed.

Since the error to be corrected is due to Applicants' error, a charge of $100.00 is believed to be due. The Commissioner is authorized to charge any additional fees for this paper or credit any overpayment to Deposit Account No. No. 22-0259.

Respectfully submitted,
VEDDER PRICE P.C.
Attorney for Applicants

By: Thomas J. Kowalski

Thomas J. Kowalski
Reg. No. 32,147
Deborah L. Lu, Ph.D.
Reg. No. 50,940
Tel. No. (212) 407-7700
Fax No. (212) 407-7799

PTO/SB/44 (09-07)
Approved for use through 08/31/2013. OMB 0651-0033
U.S. Patent and Trademark Office; U.S. DEPARTMENT OF COMMERCE

(Also Form PTO-1050)

UNITED STATES PATENT AND TRADEMARK OFFICE
CERTIFICATE OF CORRECTION

Page 1 of 1

PATENT NO. : 6,049,607
APPLICATION NO. : 09/157,035
ISSUE DATE : April 11, 2000
INVENTOR(S) : Joseph Marash, et al.

It is certified that an error appears or errors appear in the above-identified patent and that said Letters Patent is hereby corrected as shown below:

ON THE FACE OF THE PATENT

(73) Assignee: ~~Lamar Signal Processing, Yokneam, Israel~~ Andrea Electronics Corporation, Melville, New York

MAILING ADDRESS OF SENDER (Please do not use customer number below):
Vedder Price P.C.
1633 Broadway, 47th Floor
New York, New York 10019

NEWYORK/#335390.1

UNITED STATES PATENT AND TRADEMARK OFFICE

Commissioner for Patents
United States Patent and Trademark Office
P.O. Box 1450
Alexandria, VA 22313-1450
www.uspto.gov

3-19-2014

Patent No: 6,049,607
Serial Number: 09/157,035
Inventor(s): Joseph Marash, et. al.
Issued: April 11, 2000

Request for Certificate of Correction

Consideration has been given your request for the issuance of a certificate of correction for the above-identified patent under the provisions of Rule(s) 1.322or 1.323.

Assignees' names and addresses (assignment data) printed in a patent, are based *solely* on information supplied in the appropriate space for identifying the assignment data, i.e., item 3 of the Issue Fee Transmittal Form PTOL-85B. Granting of a request under 37 CFR 3.81(b) is required to correct applicant's error providing ***incorrect or erroneous*** assignment data, *before* issuance of a Certificate of Correction, under 37 CFR 1.323 *(see Manual of Patent Examining Procedures (M.P.E.P) Chp.1400, sect. 1481).* This procedure is required *at any time after the issue fee is paid*, including after issuance of the patent.

In view of the foregoing, your request is hereby denied.

A request to correct the Assignee under 37 CFR 3.81(b) should include:

A. **the processing fee set forth in 37 CFR 1.17(i) (currently $130);**
B. a statement that the failure to include the correct assignee name on the PTOL-85B was inadvertent; and
C. a copy of the Notice of Recordation of Assignment Document, reflecting the reel and frame number where the assignment(s) is recorded and/or reflecting proof of *the date* the assignment was submitted for recordation.

In the Request, Applicant(s) may request that the file be forwarded to Certificates of Correction Branch, for issuance of a Certificate of Correction, if the Request is granted.

Any request under 37 CFR 3.81(b) should be directed to the following address or facsimile number:

By mail:

Mail Stop PETITIONS
Commissioner for Patents
Post Office Box 1450
Alexandria, VA 22313-1450

By hand: Customer Service Window
Mail Stop Petitions
Randolph Building
401 Dulany Street
Alexandria, VA 22314

By fax: (703) 872-9306
ATTN: Office of Petitions

If a fee (currently $100) was previously submitted for consideration of a Request for Certificate of Correction, under CFR 1.323, to correct assignment data, no additional fee is required.

Eva James
Certificate of Correction Branch
571-272-3422

Vedder Price P.C.
1633 Broadway, 47th floor
New York, NY

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FEB 15 2000
PATENT & TRADEMARK OFFICE

PATENT
670025-7007

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Applicant(s) : Joseph Marash et al.
Serial No. : 09/157,035
For : INTERFERENCE CANCELING METHOD AND APPARATUS
Filed : September 18, 1998
Examiner : SAINT-SURIN, J.
Art Unit : 2747
Batch No. : G01

745 Fifth Avenue, New York, NY 10151

I hereby certify that this correspondence is being deposited with the United States Postal Service as first class mail in an envelope addressed to: Assistant Commissioner for Patents, Washington, DC 20231, on February 8, 2000.

THOMAS J. KOWALSKI, Reg. No. 32,147
Name of Applicant, Assignee or Registered Representative

Signature

February 8, 2000
Date of Signature

COMMUNICATION

Assistant Commissioner for Patents
Washington, D.C. 20231

Dear Sir:

In response to the NOTICE OF ALLOWABILITY, mailed December 21, 1999, enclosed are formal drawings for this application.

Respectfully submitted,
FROMMER LAWRENCE & HAUG LLP

By:
THOMAS J. KOWALSKI
Reg. No. 32,147
(212) 588-0800

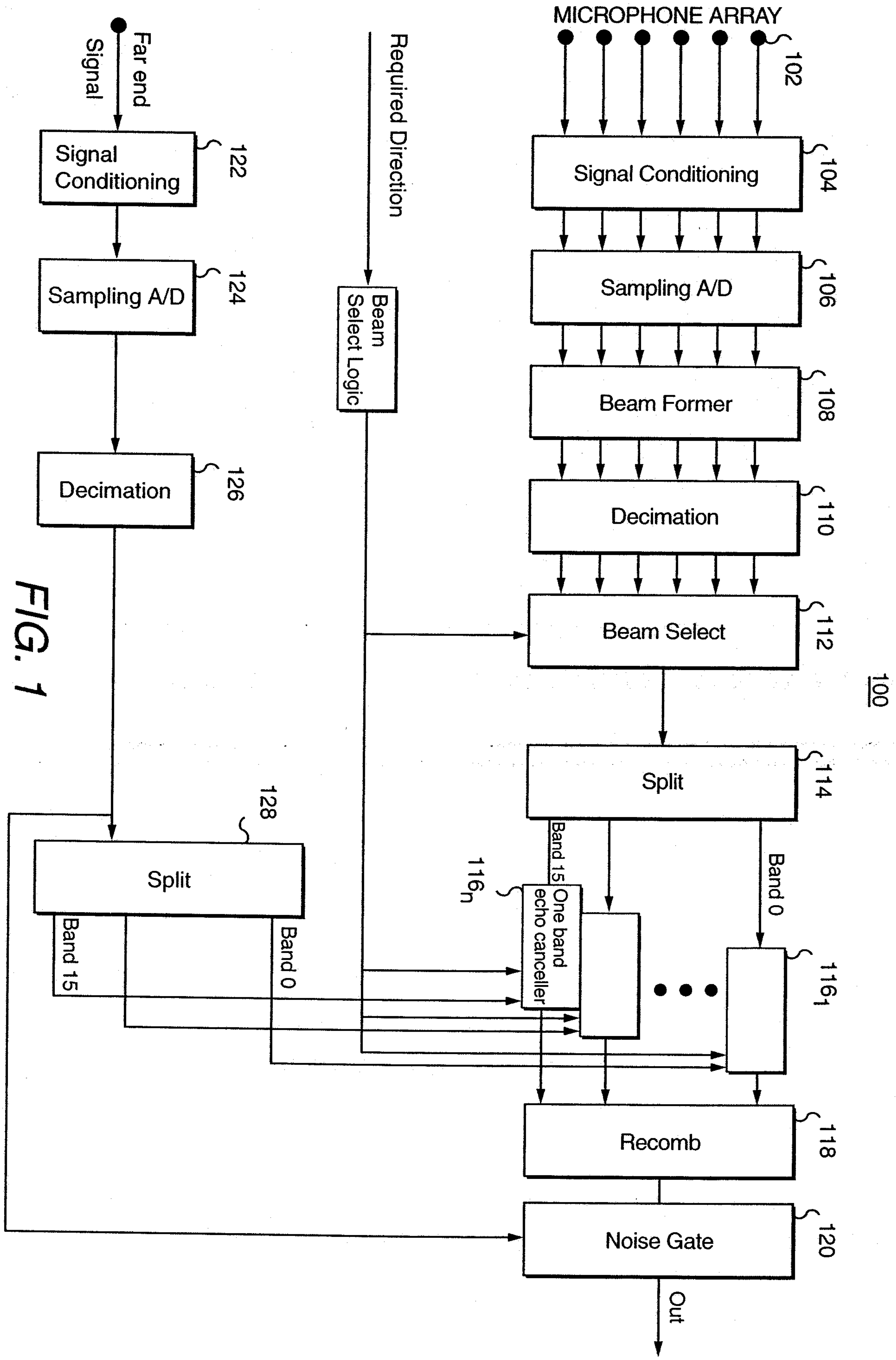

MICROPHONE ARRAY
102
Signal Conditioning
104
Sampling A/D
106
Beam Former
108
Decimation
110
Beam Select
112
100
Split
114
Band 15
Band 0
116n
One band echo canceller
116₁
Recomb
118
Noise Gate
120
Out
Required Direction
Beam Select Logic
Far end Signal
Signal Conditioning
122
Sampling A/D
124
Decimation
126
Split
128
Band 15
Band 0

FIG. 1

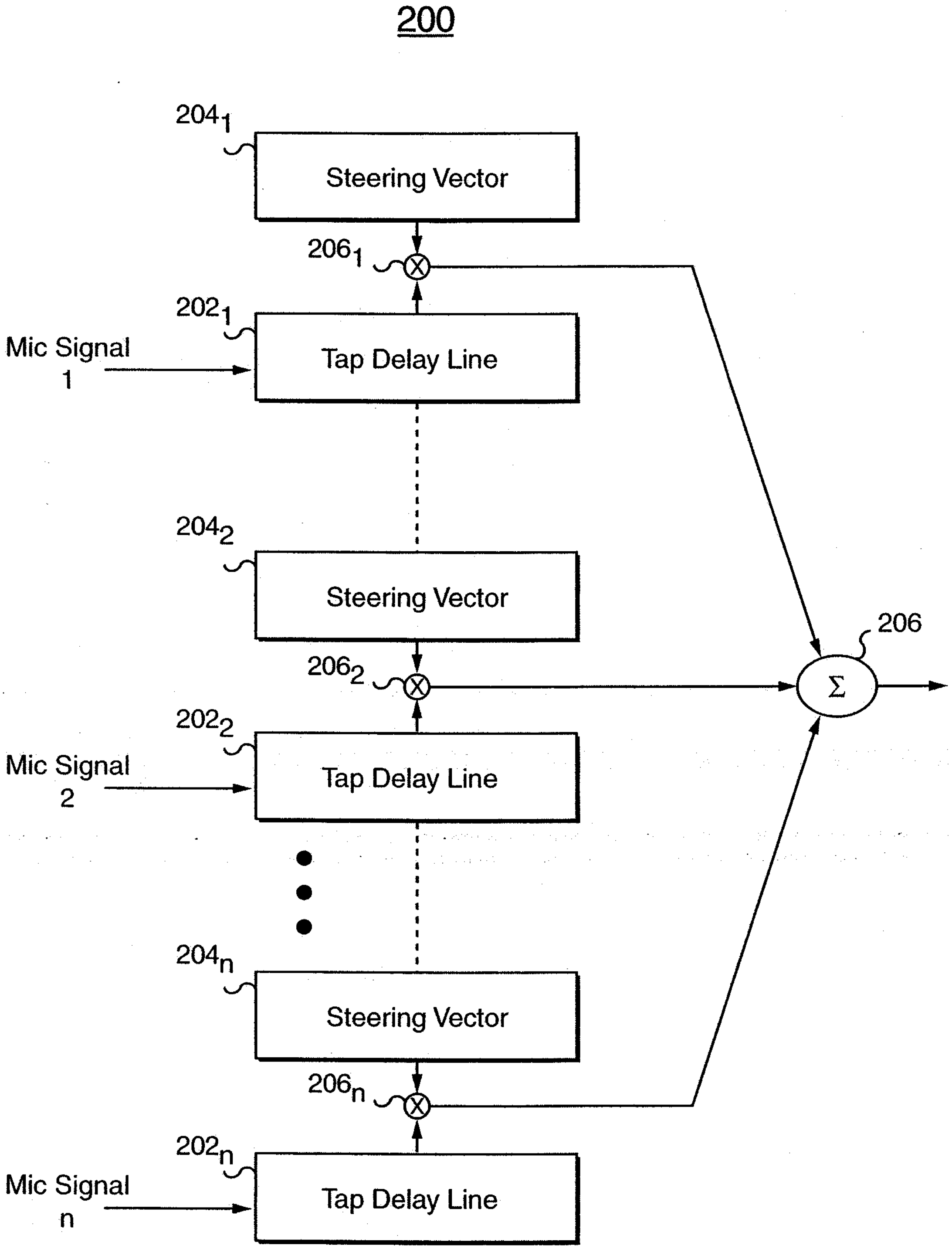
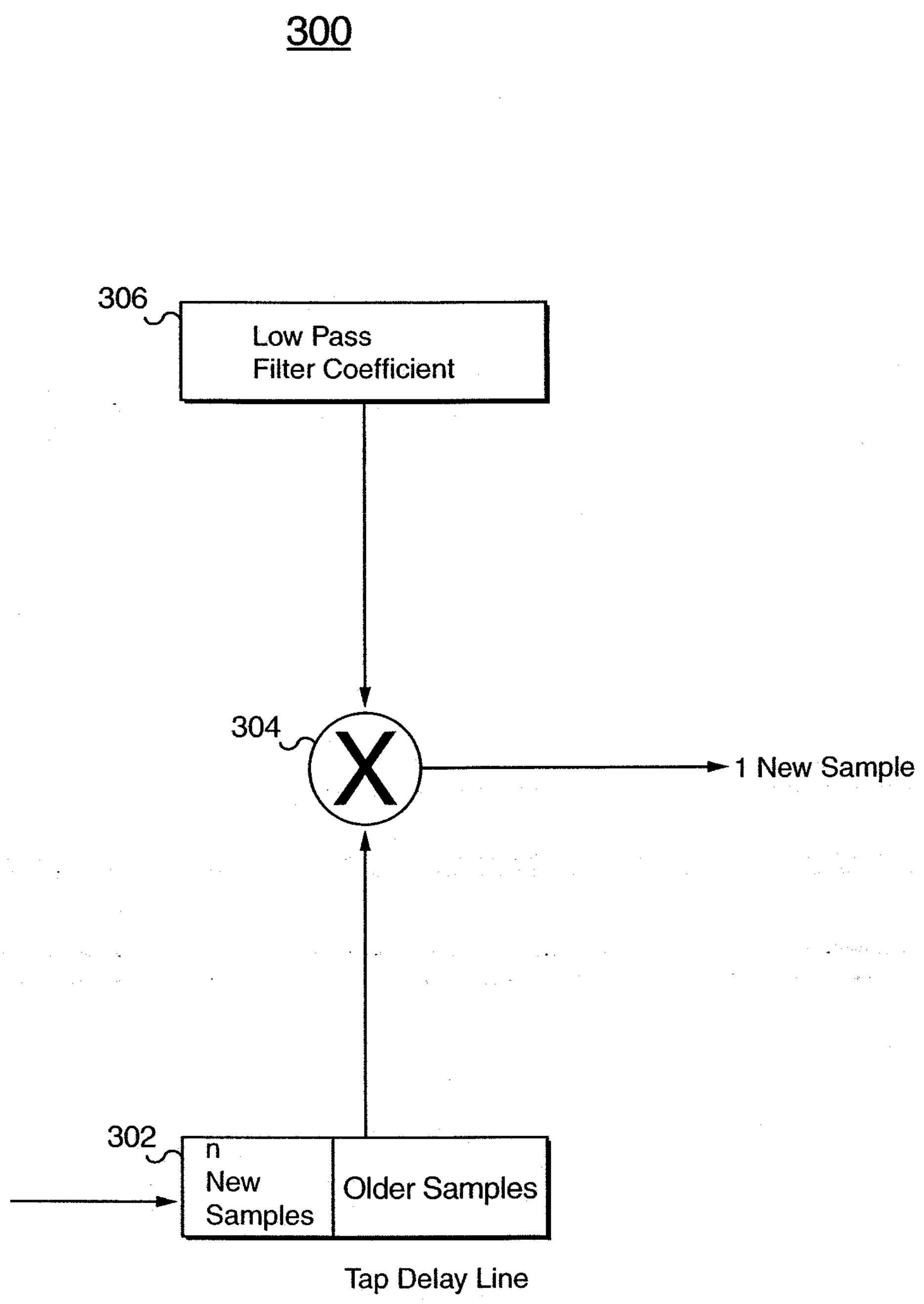

200
204₁
Steering Vector
206₁
202₁
Mic Signal 1
Tap Delay Line
204₂
Steering Vector
206₂
206
Σ
202₂
Mic Signal 2
Tap Delay Line
204ₙ
Steering Vector
206ₙ
202ₙ
Mic Signal n
Tap Delay Line

FIG. 2



Decimation by n

FIG. 3

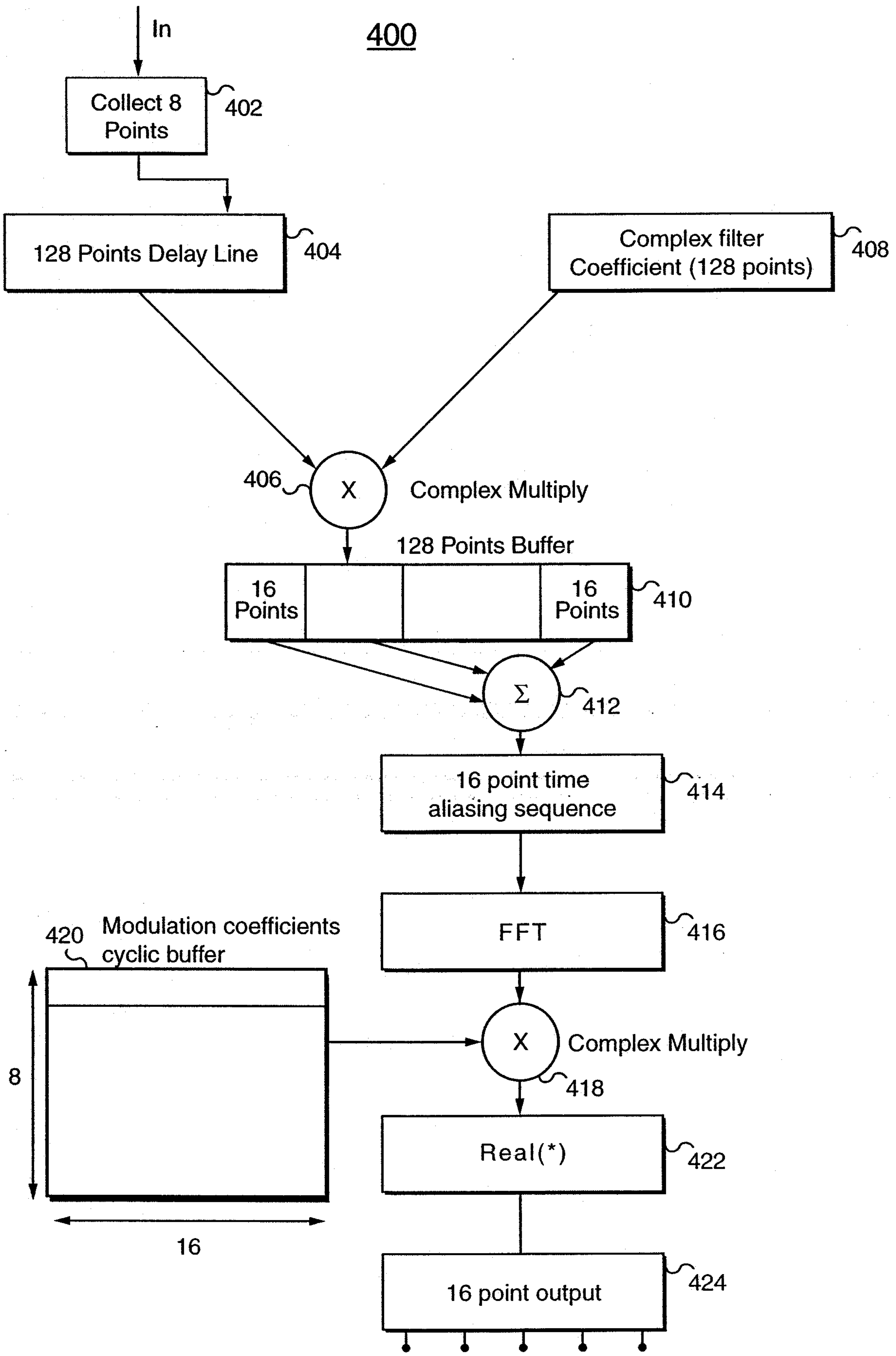

In
400
Collect 8 Points
402
128 Points Delay Line
404
Complex filter Coefficient (128 points)
408
406
X
Complex Multiply
128 Points Buffer
16 Points
16 Points
410
Σ
412
16 point time aliasing sequence
414
FFT
416
420
Modulation coefficients cyclic buffer
8
16
X
Complex Multiply
418
Real(*)
422
16 point output
424

FIG. 4

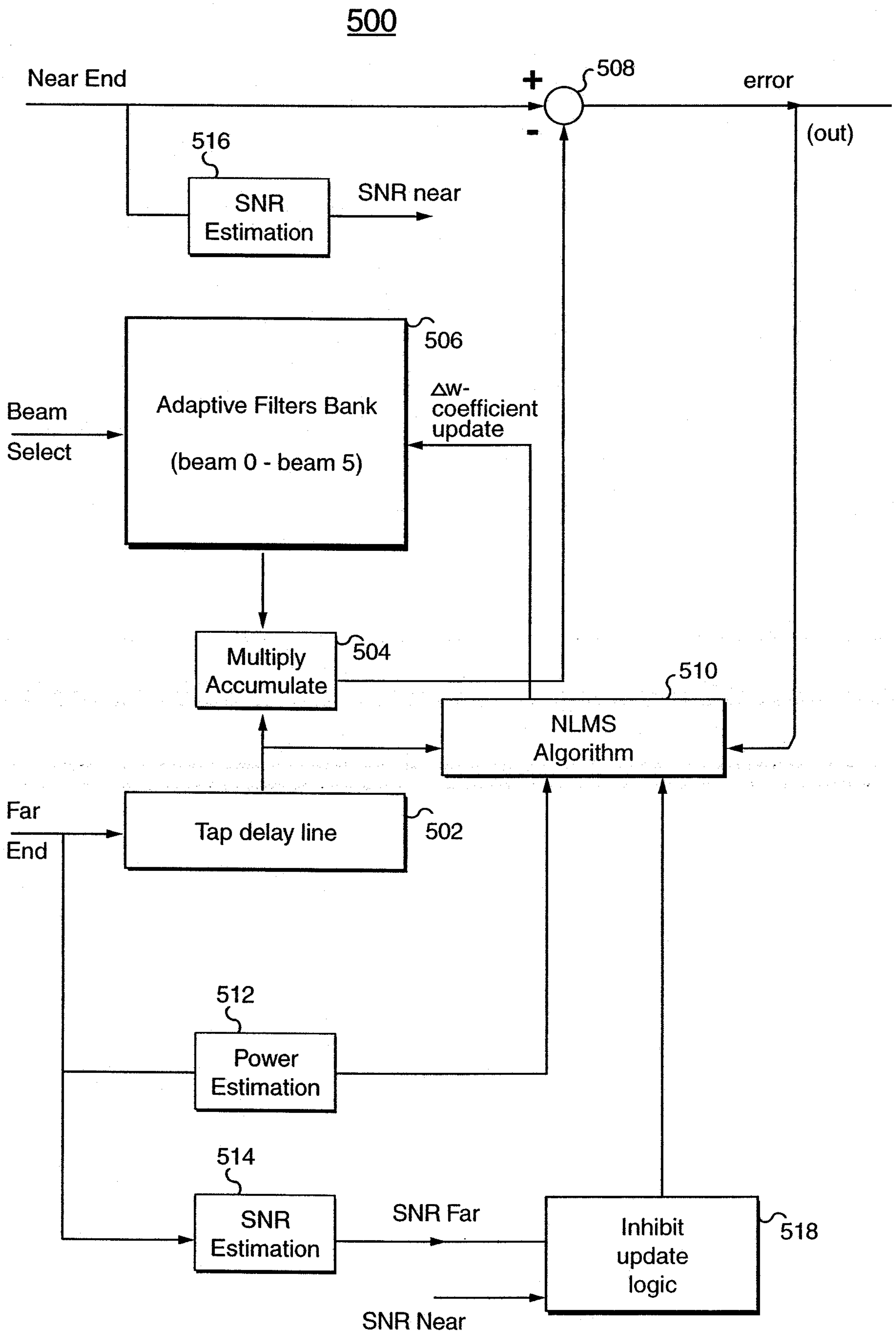
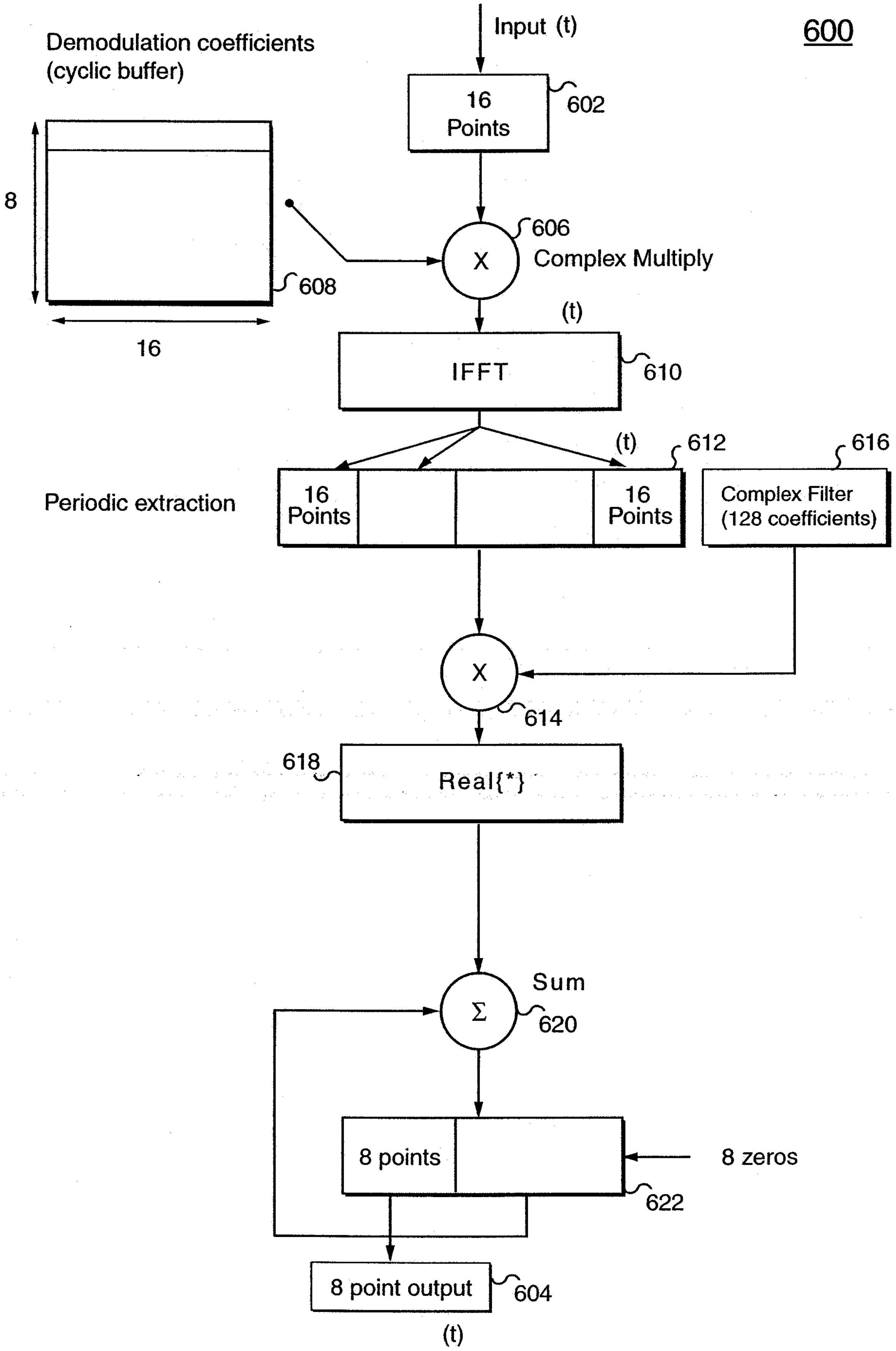

500
Near End
+
-
508
error
(out)
516
SNR Estimation
SNR near
506
Beam Select
Adaptive Filters Bank
(beam 0 - beam 5)
Δw-coefficient update
Multiply Accumulate
504
510
NLMS Algorithm
Far End
Tap delay line
502
512
Power Estimation
514
SNR Estimation
SNR Far
Inhibit update logic
518
SNR Near

FIG. 5



Input (t)
600
Demodulation coefficients
(cyclic buffer)
16
Points
602
8
606
X
Complex Multiply
608
16
(t)
IFFT
610
(t)
612
616
Periodic extraction
16
Points
16
Points
Complex Filter
(128 coefficients)
X
614
618
Real{*}
Sum
Σ
620
8 points
8 zeros
622
8 point output
604
(t)

FIG. 6

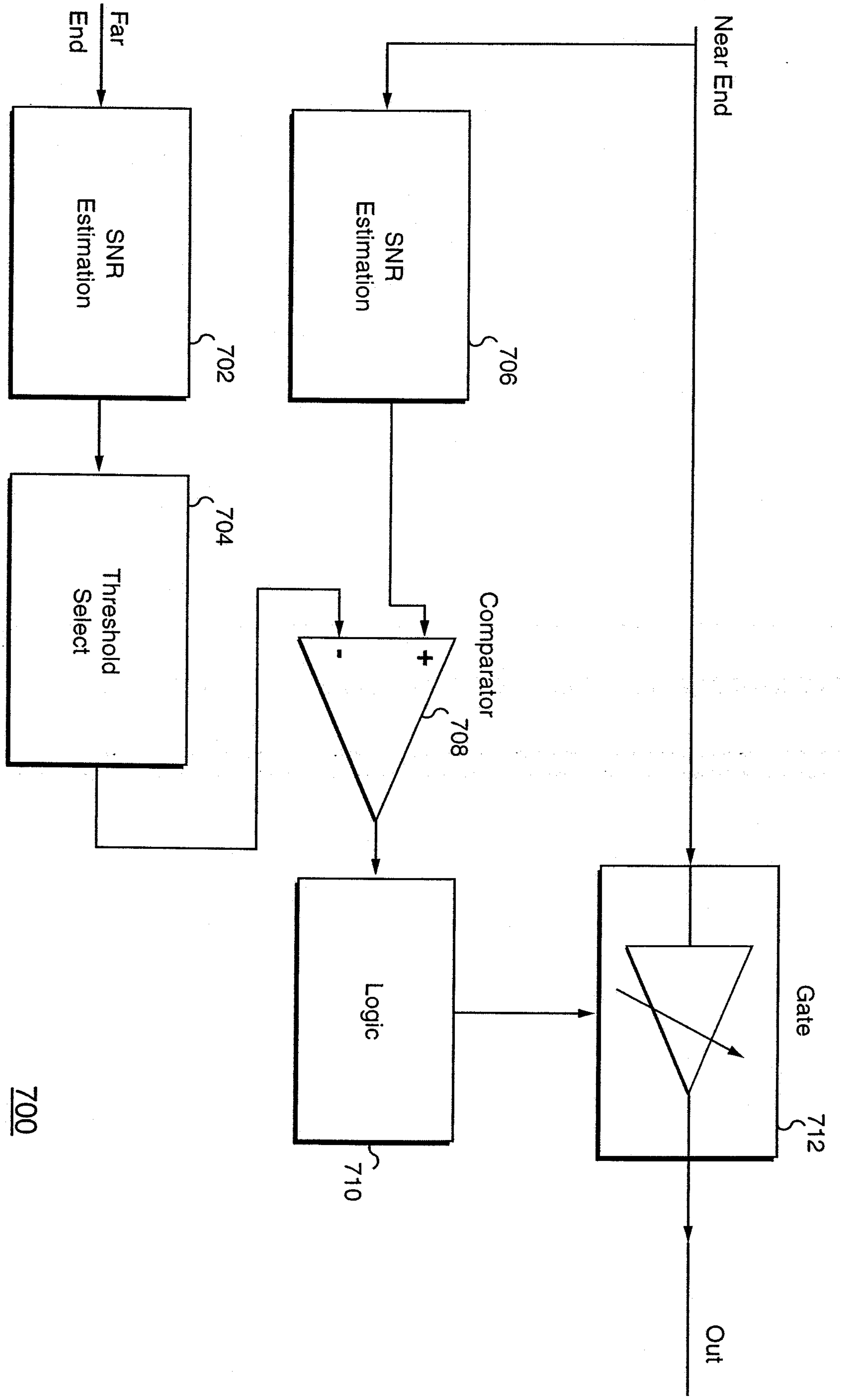

Near End
Far End
SNR Estimation
706
SNR Estimation
702
Threshold Select
704
Comparator
+
-
708
Logic
710
Gate
712
Out
700

FIG. 7

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OF PATENTS AND TRADEMARKS
Washington, D.C. 20231

CHANGE OF ADDRESS/POWER OF ATTORNEY

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THE CORRESPONDENCE ADDRESS HAS BEEN CHANGED TO CUSTOMER # 20999

THE PRACTITIONERS OF RECORD HAVE BEEN CHANGED TO CUSTOMER # 20999

ON 03/28/01 THE ADDRESS OF RECORD FOR CUSTOMER NUMBER 20999 IS:

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745 FIFTH AVENUE
NEW YORK NY 10151

AND THE PRACTITIONERS OF RECORD FOR CUSTOMER NUMBER 20999 ARE:

21002 25456 25506 27413 28029 29309 30800 31086 31223 32147
33943 34930 35582 37274 37514 37870 37937 38511 38580 39440
40352 41205 41531 43228 44071

PTO INSTRUCTIONS: PLEASE TAKE THE FOLLOWING ACTION WHEN THE CORRESPONDENCE ADDRESS HAS BEEN CHANGED TO CUSTOMER NUMBER: RECORD, ON THE NEXT AVAILABLE CONTENTS LINE OF THE FILE JACKET, 'ADDRESS CHANGE TO CUSTOMER NUMBER'. LINE THROUGH THE OLD ADDRESS ON THE FILE JACKET LABEL AND ENTER ONLY THE 'CUSTOMER NUMBER' AS THE NEW ADDRESS. FILE THIS LETTER IN THE FILE JACKET. WHEN ABOVE CHANGES ARE ONLY TO FEE ADDRESS AND/OR PRACTITIONERS OF RECORD, FILE LETTER IN THE FILE JACKET.
THIS FILE IS ASSIGNED TO GAU 2747.

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TALBOT-1/97

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CHANGE OF ADDRESS/POWER OF ATTORNEY

FILE LOCATION 9200 SERIAL NUMBER 09157035 PATENT NUMBER 6049607

THE CORRESPONDENCE ADDRESS HAS BEEN CHANGED TO CUSTOMER # 20999

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21002 25456 25506 27413 28029 29309 30800 31086 31223 32147
33943 34930 35582 37274 37514 37870 37937 38511 38580 39440
40352 41205 41531 43228 44071

PTO INSTRUCTIONS: PLEASE TAKE THE FOLLOWING ACTION WHEN THE CORRESPONDENCE ADDRESS HAS BEEN CHANGED TO CUSTOMER NUMBER: RECORD, ON THE NEXT AVAILABLE CONTENTS LINE OF THE FILE JACKET, 'ADDRESS CHANGE TO CUSTOMER NUMBER'. LINE THROUGH THE OLD ADDRESS ON THE FILE JACKET LABEL AND ENTER ONLY THE 'CUSTOMER NUMBER' AS THE NEW ADDRESS. FILE THIS LETTER IN THE FILE JACKET. WHEN ABOVE CHANGES ARE ONLY TO FEE ADDRESS AND/OR PRACTITIONERS OF RECORD, FILE LETTER IN THE FILE JACKET.
THIS FILE IS ASSIGNED TO GAU 2747.

PTO-FMD
TALBOT-1/97

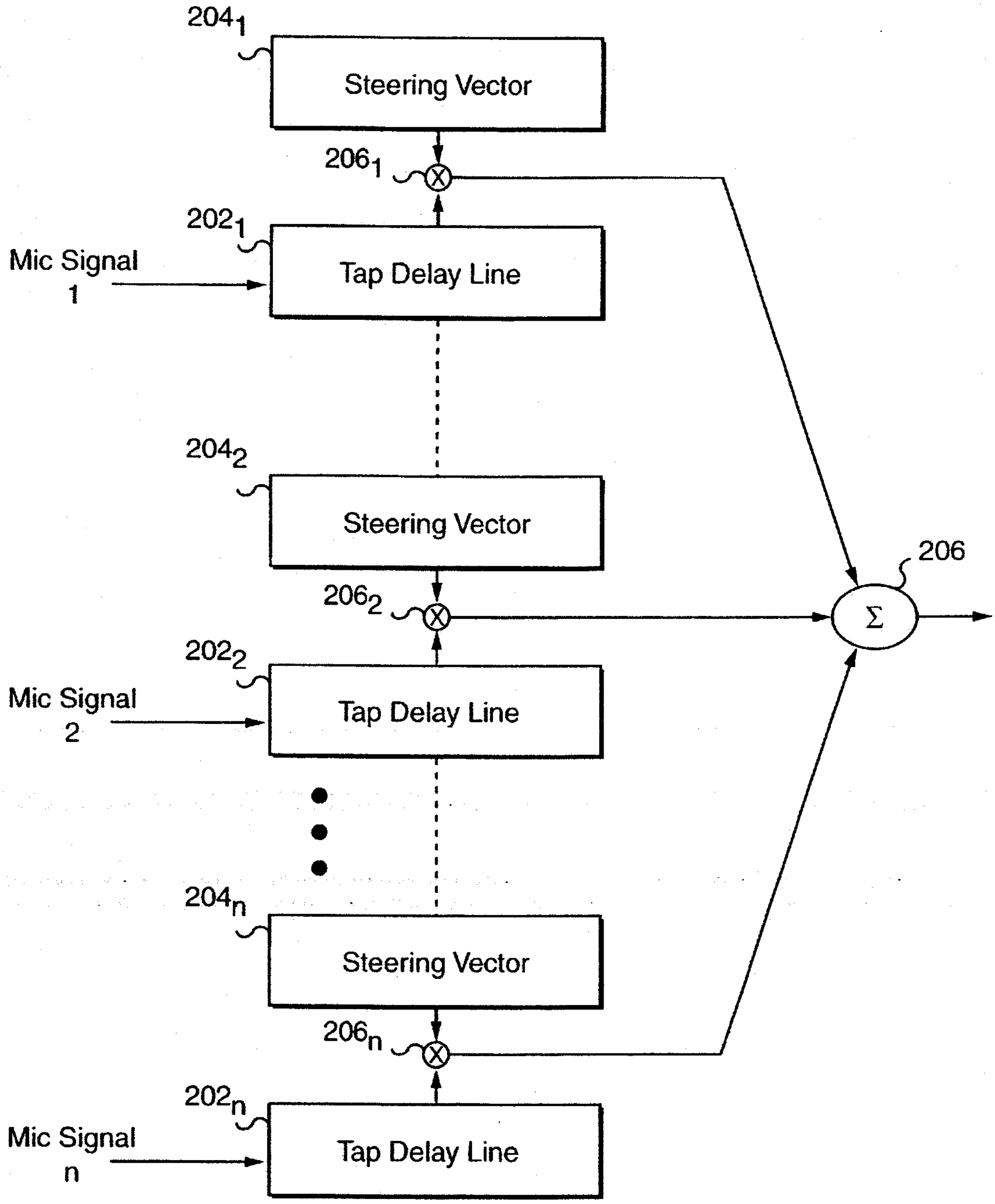
204_1
Steering Vector
206_1
202_1
Mic Signal 1
Tap Delay Line
204_2
Steering Vector
206_2
202_2
Mic Signal 2
Tap Delay Line
206
Σ
204_n
Steering Vector
206_n
202_n
Mic Signal n
Tap Delay Line

FIG. 2

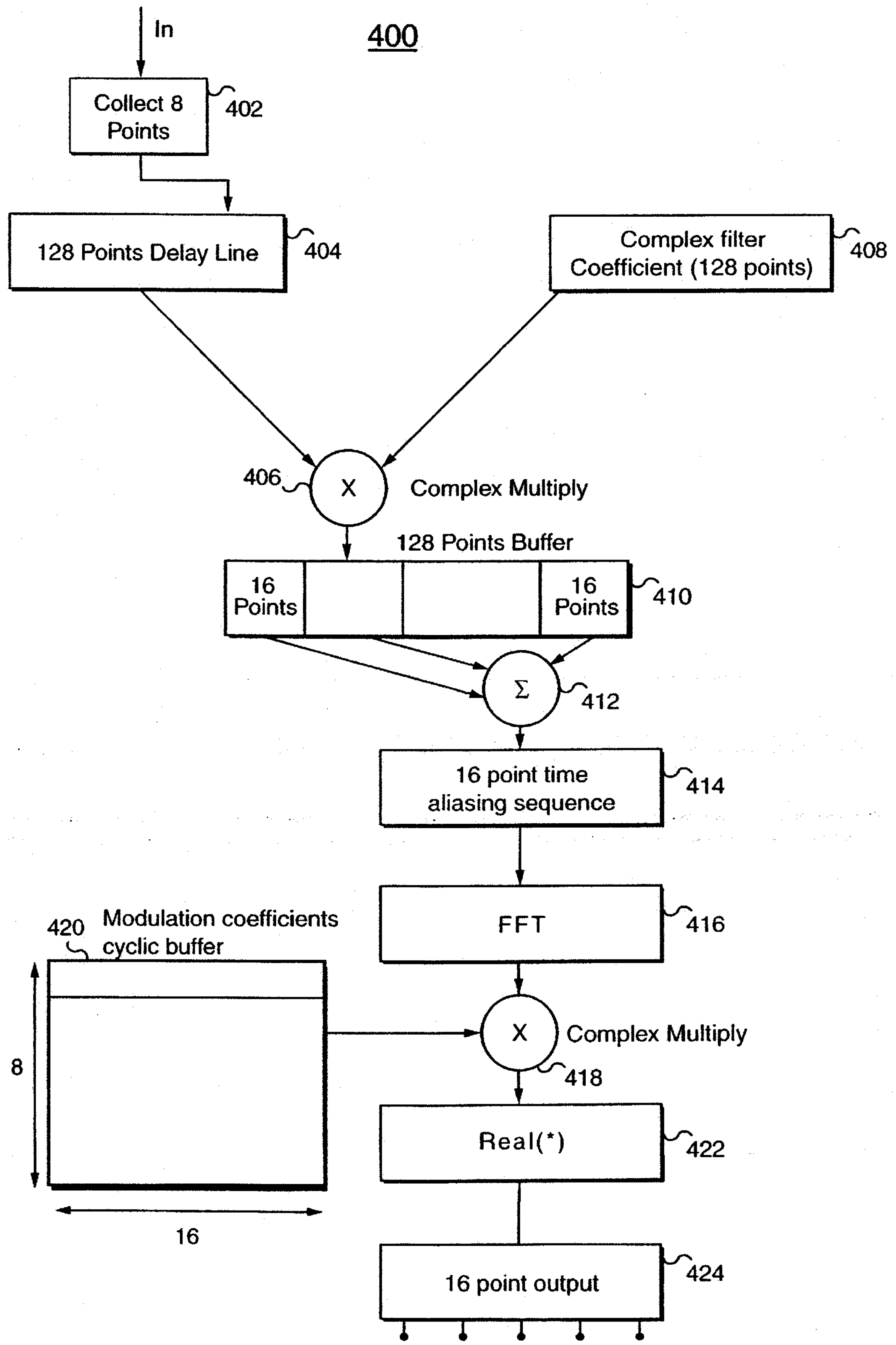
In
400
Collect 8 Points
402
128 Points Delay Line
404
Complex filter Coefficient (128 points)
408
406
X
Complex Multiply
128 Points Buffer
16 Points
16 Points
410
Σ
412
16 point time aliasing sequence
414
FFT
416
420
Modulation coefficients cyclic buffer
8
16
X
Complex Multiply
418
Real(*)
422
16 point output
424

FIG. 4

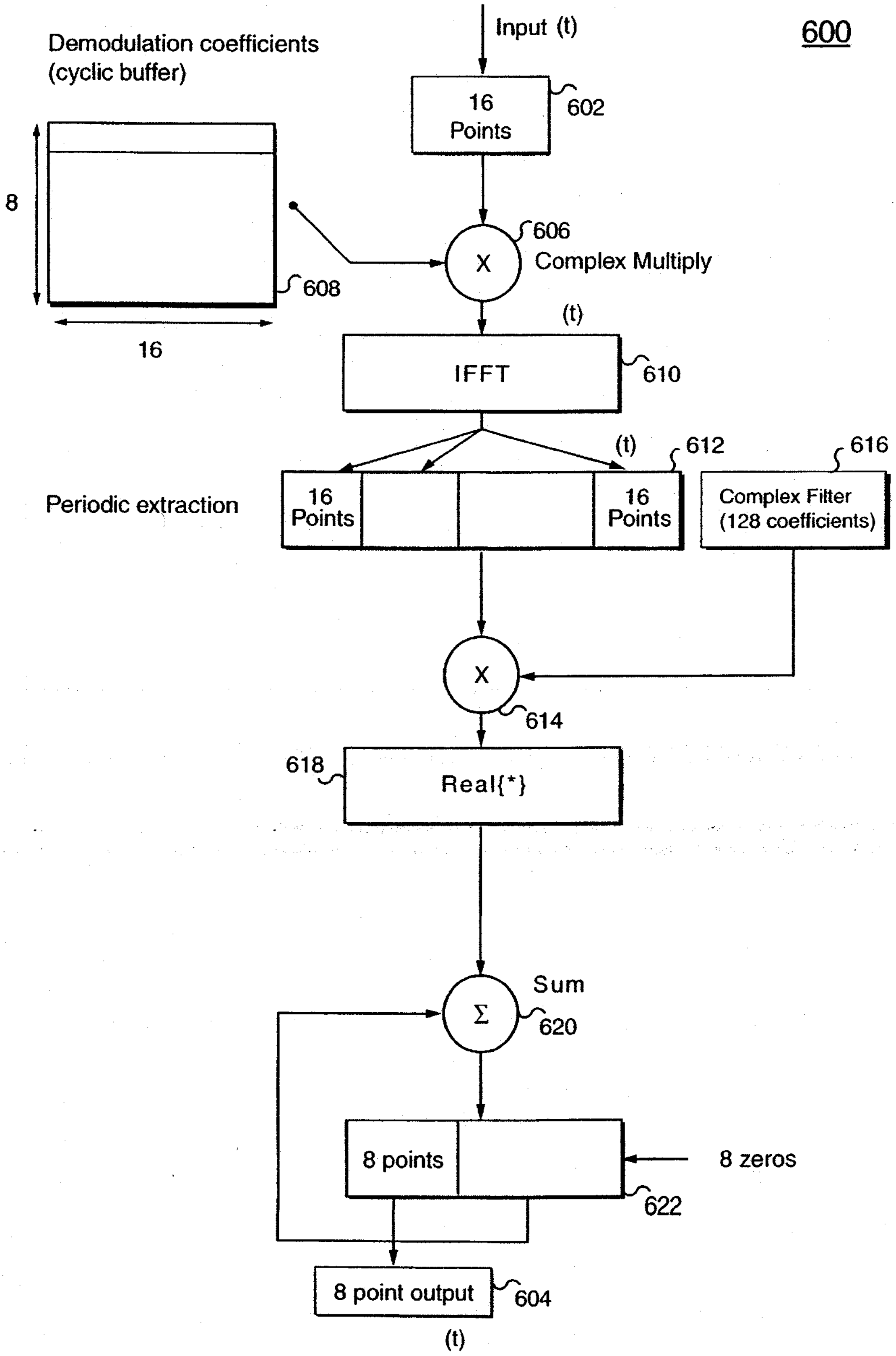
600
Input (t)
16 Points
602
Demodulation coefficients (cyclic buffer)
8
16
608
606
X
Complex Multiply
(t)
IFFT
610
(t)
612
616
Periodic extraction
16 Points
16 Points
Complex Filter (128 coefficients)
X
614
618
Real{*}
Sum
Σ
620
8 points
8 zeros
622
8 point output
604
(t)

FIG. 6

INTERFERENCE CANCELING METHOD AND APPARATUS

RELATED APPLICATIONS

Reference is made to co-pending U.S. application Ser. Nos. 08/672,899 (allowed), 09/130,923, 08/840,159, 09/059,503 and 09/055,709, each of which is hereby incorporated herein by reference; and each and every document cited in those applications, as well as each and every document cited herein, is hereby incorporated herein by reference.

FIELD OF THE INVENTION

The present invention relates to an interference canceling method and apparatus and, for instance, to an echo canceling method and apparatus which provides echo-canceling in full duplex communication, especially teleconferencing communications.

BACKGROUND OF THE INVENTION

Tele-conferencing plays an extremely important role in communications today. The teleconference, particularly the telephone conference call, has become routine in business, in part because teleconferencing provides a convenient and inexpensive forum by which distant business interests communicate. Internet conferencing, which provides a personal forum by which the speakers can see one another, is enormously popular on the home front, in part because it brings together distant family and friends without the need for expensive travel.

In a teleconferencing system, the sounds present in a room, hereinafter referred to as the "near-end room" such as those of a near-end speaker are received by a microphone, transmitted to a "far end system" and broadcast by a far-end loudspeaker. Similarly, the far-end speaker is received by the far-end microphones and transmitted to the near-end system, and broadcast by the near-end loudspeaker. The near-end microphone receives the broadcasted sounds along with their reverberations and transmits them back to the far-end, together with the desired signals generated by, for example, speakers at the near-end, thereby resulting in a disturbing echo heard by the speaker at the far-end. The far-end speaker will hear himself after the sound has traveled to the near-end system and back, thereby resulting in a delayed echo which will annoy and confuse the far-end speaker. The problem is compounded in video and internet conferencing systems where the delay is more extremely pronounced.

The simplest way to overcome the problem of echo is by blocking the near-end microphone while the far-end signal is broadcast by the near-end loudspeaker. Sometimes referred to as "ducking", the technique of blocking the microphone is effectively a half-duplex communication. Problematically, if the microphone is blocked for a prolonged period to avoid transmission of the reverberations, the half-duplex communication becomes a significant drawback because the far-end speaker will lose too much of the near-end speaker. In the video or Internet conferencing system, where the delay created by the communication lines is extreme, ducking becomes quite annoying.

A more complex method to avoid echo is to employ an echo canceling system which measures the signals send from the far-end and broadcast it the near-end loudspeaker, estimates the resulting signal present at the near-end microphone (including the reverberations) and subtracts those signals representing the echo from the near-end microphone signals. The echo-free signals are then transmitted back to the far-end system.

In order to reduce the echo from the near-end microphone signal, it is required to obtain the transfer function that expresses the relationship between the near-end loudspeaker signal and the reverberations as they actually appear at the near-end microphone. This transfer function depends on the relative position of the near-end loudspeaker to the near-end microphone, the room structure, position of the system and even the presence of people in the room. Since it is impossible to predict these parameters a priori, it is preferred that the echo-canceling system updates the transfer function continuously in real time.

The adaptation process by which the echo-canceling system is updated in real time may be an LMS (least means square) adaptive filter (Widrow, et al., Proc. IEEE, vol. 63, pp. 1692–1716, Proc. IEEE, vol. 55, No. 12, December 1967) with the far-end signal used as the reference signal. The LMS filter estimates the interference elements (echoes) present in the interfered channel by multiplying the reference channel by a filter and subtracting the estimated elements from the interfered signal. The resulting output is used for updating the filter coefficients. The adaptation process will converge when the resulting output energy is at a minimum, leaving an echo-free signal.

Important to the adaptation process is the selection of the size of the adaptation step of the filter coefficients. In the standard LMS algorithm the step size is controlled by a predetermined adaptation coefficient, the level of the reference channel and the output level. In other words, the adaptation process will have bigger steps for strong signals and smaller steps for weaker signals.

A better behaved system is one in which its adaptation steps are independent of the reference channel levels. This is accomplished by normalizing the adaptation coefficient by the reference channel energy, this method is called the Normalized Least Mean Square (NLMS) as, for example, described in see for example "*A Family of Normalized LMS Algorithms*", Scott C. Douglas, IEEE Signal Processing Letters, Vol. 1, No. 3, March 1994. It should be noted that the energy estimator, if not designed properly, may fail to track when large and fast changes in the level of the reference channel occur. Thus, the normalized coefficient may be too big during the transition period, and the filter coefficient may diverge.

Another problem is that the adaptive process feeds the output back to determine the new filter coefficients. When the interfering elements in the signal are less pronounced than the non-interfering signal, there is not much to reduce and the filter may diverge or converge to a wrong value which results in signal distortions.

When properly converged, the adaptive filter actually estimates the transfer function between the far-end loudspeaker signal and the echo elements in the main channel. However, changes in the room will effect a change in the transfer function and the adaptive process will adapt itself to the new conditions. Sudden or quick changes, in particular, will take the adaptive filter time to adjust for and an echo will be present until the filter adapts itself to the new conditions.

In order to improve the audio quality, sometimes a number of microphones are used instead of a single one. This system either selects a different microphone each time someone is speaking in the room or creates a directional beam using a linear combination of microphones. By multiplexing the microphones or steering the directional audio receive and convert acoustic sound in a room into an analog signal which is amplified by the signal conditioning block **104** and converted into digital form by the A/D converter **106**. While FIG. **1** appears to depict the microphone elements **102** as an array, it will be appreciated by those skilled in the art that other configurations are readily applicable to the present invention. The microphone elements, for example, may be arranged in a circular array, a linear, or any other type of array. The A/D converter **106** may be an array of Delta Sigma converters set to, for example, a sampling frequency of 64 KHz per channel but, of course, may be substituted with other types of converters and sampling frequencies which are suitable as those skilled in the art will readily understand.

The sampled signals of each microphone are stored in a tap delay line (not shown) and multiplied by a steering matrix in the beam forming unit **108** to form a number of directional beams. As an example, 6 beams are formed which are aimed in directions evenly spread over 360 degrees (60 degrees apart). Of course, the present invention is not limited to any specific number of beams as one skilled in the art will readily understand. The beam signals are then low pass filtered to, for example, 8 KHz and decimated by decimating unit **110** to reduce the sampling rate and hence the computational load on the system. In this manner, the sampling rate is reduced to 16 KHz for each channel. It shall be appreciated that the decimation process may be performed prior to the beamforming process to further reduce the processing burden.

The system receives an indication as to the direction of the speaker either through a direction finding system or through a manual steering process. In the exemplary embodiment, the beam select logic unit **112** selects the beam with the closest direction to that actual and performs echo cancellation processing on the selected beam.

A particular aspect of the present invention is that the selected beam is split into a number of frequency bands, preferably 16 evenly spaced bands, by the beam splitter **114** such that echo cancellation processing is performed on each frequency band separately. Without this arrangement, an echo which typically lasts for more than 100 msec would require an adaptive filter, assuming that the filter samples the 100 msec of signal at a rate of 16 KHz, to have 1600 coefficients. Such a long adaptive filter is not likely to converge in the time that the echo is present. Moreover, an adaptive filter of 1600 coefficients presents an enormous processing burden which is unrealistic to handle. By splitting the bands into, for example, 16 channels the present invention reduces the sampling rate for each adaptive filter to, in this case, 2 KHz per channel. It will be appreciated that, not only is this system much more manageable, the adaptive filters can be optimized for each frequency separately by, for example, selecting longer filters for lower frequencies where the echo is typically located and shorter filters for higher frequencies where the echo is less. In this case, the filter lengths range, for example, from 16 to 128 coefficients. With this arrangement, the adaptive filters can converge much more easily with these lengths, the treatment of each band is independent from the others thereby preventing the problem of a broadband filter concentrating on a band limited interference while ignoring less pronounced ones and the processing burden is reduced.

Meanwhile, the far end signal (referred to as the reference channel) is conditioned, sampled, decimated and split in the manner discussed above by respective signal conditioning block **122**, A/D converters **124**, decimating unit **126** and splitter **128**. Each band of the selected beam is processed for echo reduction using echo canceling unit **116**$_{1-m}$. While Normalized LMS filters are preferred, those skilled in the art will readily understand that other type of adaptive filters are applicable to the present invention. The resulting echo-free signals of the different frequency bands are recombined into one broadband output by a recombine output unit **118**.

The output of the recombined process is fed into a noise gate processor **120**. The purpose of the noise gate is to prevent steady background noise in the room (such as fan noise) from being transmitted to the far end system and eliminate residual echoes. The system of the present invention measures the level of the steady noise and blocks up the signals that are below a certain threshold above this noise level. When residual echoes are present they may penetrate the process and be transmitted to the far end system. In order to prevent that, the blocking threshold is actively adjusted to the level of the signal present at the reference channel (far end). When a high level energy is detected at the far end signal, the threshold will be boosted up and gradually reduced when this signal disappears. This will prevent residual echoes from being transmitted while leaving only speech signals from the near end.

FIG. 2 illustrates the beamforming unit **200** (FIG. 1, **108**) of the present invention. Signals originated at a certain relative direction to the microphone array arrive at different phases to each microphone. Summing them up will create a reduced signal depending on the phase shift between the microphones. The reduction goes down to zero when the phases of the microphones are the same, thus creating a preferred direction while reducing all other directions. In the beamforming process, the microphone signals are phase shifted to create a zero phase difference for signals originated at a predetermined direction. The phase shift is achieved by multiplying the microphone signal stored in the tap delay lines **202**$_{1-n}$ by a FIR filter coefficient or steering vector output from steering vector units **204**$_{1-n}$.

In one embodiment, a different weight is applied for each microphone to create a shading effect and reduce the side lobe level. The weighting factors are implemented as part of the FIR filter coefficients. The filters for each direction and each microphone are pre-designed and stored as a steering vector matrix **204**$_{1-n}$. The microphone signals are stored in a tapped delay line **2021-n** with the length of the FIR filter. For each direction, each microphone delay line is multiplied by multipliers **206**$_{1-n}$ by its FIR and summed with the other microphones after they have been multiplied. The process repeats for each direction resulting in a beam output for each direction.

FIG. 3 illustrates the decimation unit **300** (FIG. 1, **110**, **126**) of the present invention. Decimation, which is intended to reduce the sampling frequency, can be done only once the high frequency elements are removed to maintain the Nyquist criteria. For example, if the sampling frequency is to be reduced to 16 KHz, it is necessary to make sure that the signal does not contain elements above 8 KHz because sampling will result in aliasing. In order to remove the troublesome high frequencies, the signals are first filtered by a low pass filter that cuts off the higher frequencies. In more detail, the beam samples are stored in a tapped delay line **302** and multiplied via a multiplier **304** by a low pass filter coefficient produced by the low pass filter **306**.

FIG. 4 illustrates the beam splitting unit **400** (FIG. 1, **114**, **128**) of the present invention. Although various beam splitting techniques may be employed, it is preferred that the generalized DFT filter bank using single side band modulation be employed as described, for example, in "*Multirate*

FIG. **6** illustrates the recombining unit **600** (FIG. **1**, **118**) of the present invention which is symmetrical, i.e., opposite, to the band splitting technique described above. The goal here is to recombine the 16 limited frequency bands of the echo free signal into one broad band output. The process goes through an IFFT process but both the input and output are time domain signals. The recombining unit of the exemplary embodiment processes 16 input points **602** each representing 1 time domain sample per frequency band resulting in 8 output points **604** of the broadband signal. Of course, those skilled in the art will readily understand that other quantities of sampling input points are applicable to the present invention.

In more detail, the new 16 input points **602** are multiplied by a multiplier **606** with a 16 points demodulation filter coefficient which is stored in a demodulation coefficients cyclic buffer **608** containing, for example, 8 groups of 16 coefficients wherein a new group is selected each cycle. The result is processed through a 16 points IFFT **610**, or any equivalent transform, and the result of this Inverse Fast Fourier Transform is extracted to 128 complex points by duplicating the 16 points data 8 times. The 128 points result vector which is stored in a buffer **612** is multiplied via the multiplier **614** by a 128 point complex coefficient generated by a predesigned complex filter **616** and stored in real buffer **618**. The real portion of the result is summed by summer **620** into a 128 points cyclic history buffer **622** in which the oldest 8 points are taken as the result **604** and replaced with zeros in the buffer **622** for the next iteration of the recombination process.

FIG. **7** illustrates the noise gate system **700** (FIG. **1**, **120**) of the present invention. The far end signal-to-noise ratio SNR is calculated by SNR estimation unit **702** which estimates the signal energy of the current block (40 msec in the exemplary embodiment) and divides the signal energy by the lowest estimated block energy in the current period (2 sec in the exemplary embodiment). The threshold is selected by the threshold select depending on the far end signal-to-noise ratio SNR. When the far end SNR is low, a low threshold is selected. Once the SNR of the far end goes up, the threshold is updated immediately upwards by the threshold selection unit **704**. When the far end SNR goes down, the threshold is gradually reduced to a minimum with a decay time in the exemplary embodiment around 100 msec.

The near end signal-to-noise ratio SNR is measured by the SNR estimation unit **706** in the same manner. Then, the near end SNR signal is compared by the comparator **708** to the selected threshold. According to the logic provided by the logic circuit **710**, if the difference is positive, meaning that the near end signal is present, the gate **712** is open, preferably immediately or quickly (e.g., so as to not miss a syllable, for instance in less than about 10 msec or less such as instantly or nearly instantly). On the other hand, if the result of the comparison is negative, meaning that the near end signal is not above the allowed threshold, the gate is closed and the level of sound is significantly reduced such that the reduced signal is transmitted to the far end system. The reduction of the sound or the closure of the gate is preferably gradual such as over about 100 msec or longer, e.g., over about 0.5 sec or 1.0 sec, so as to prevent a pumping sound or noise transmission when a user is speaking fast and to have the gate truly close when there is a real pause or silence.

It will be appreciated from the foregoing description that the present invention provides an echo-canceling system which overcomes the problem of background noise in the conferencing system, reduces the residual echo to a minimum, allows full duplex communication and provides a steerable directional audio beam.

Although preferred embodiments of the present invention and modifications thereof have been described in detail herein, it is to be understood that this invention is not limited to those precise embodiments and modifications, and that other modifications and variations may be effected by one skilled in the art without departing from the spirit and scope of the invention as defined by the appended claims.

We claim:

1. An interference canceling apparatus for canceling, from a target signal generated from a target source, an interference signal generated by an interference source, said apparatus comprising:

a main input for inputting said target signal;

a reference input for inputting said interference signal;

a beam splitter for beam-splitting said target signal into a plurality of band-limited target signals and beam-splitting said interference signal into band-limited interference signals, wherein the amount and frequency of band-limited target signals equal the amount and frequency of band-limited interference signals, whereby for each band-limited target signal there is a corresponding band-limited interference signal;

an adaptive filter for adaptively filtering, each band-limited interference signal from each corresponding band-limited target signal.

2. The apparatus according to claim **1**, wherein said target signal represents speech generated at a near end of a teleconference, said reference signal represents said target signal broadcast from a far end of said teleconference and said interference signal represents an echo generated by said broadcast of said reference signal of said far end.

3. The apparatus according to claim **2**, wherein said adaptive filter is an adaptive filter array with each adaptive filter in said array filtering a different frequency band.

4. The apparatus according to claim **2**, wherein said adaptive filter estimates a transfer function of said reference signal broadcast of said far end.

5. The apparatus according to claim **4**, further comprising an inhibitor for permitting said adaptive filter to change coefficients when a signal-to-noise ratio of said reference signal exceeds a predetermined threshold over a signal-to-noise ratio of said main signal.

6. The apparatus according to claim **5**, wherein said inhibitor determines said predetermined threshold periodically.

7. The apparatus according to claim **2**, wherein said beam splitter is a DFT filter bank using single side band modulation.

8. The apparatus according to claim **2**, further comprising a beam selector for selecting at least one of a plurality of beams for adaptive filtering by said adaptive filter representing a direction from which said main signal is received.

9. The apparatus according to claim **8**, wherein said adaptive filter updates coefficients representing said transform function and comprehensively stores said coefficients for each beam selected by said beam selector.

10. The apparatus according to claim **8**, wherein said beam selector selects said plurality of said beams for simultaneous adaptive filtering by said adaptive filter.

11. The apparatus according to claim **10**, wherein said beam selector selects a beam having a fixed direction and a beam which rotates in direction.

12. The apparatus according to claim **2**, further comprising a noise gate for gating said main signal adaptively filtered by said adaptive filter by opening said noise gate

36. The method according to claim **26**, further comprising the step of gating said main signal adaptively filtered in said step of adaptive filtering by opening a noise gate when a signal-to-noise ratio at the near end is above a predetermined threshold and closing said noise gate when said signal-to-noise ratio at the near end is below the predetermined threshold.

37. The method according to claim **36**, further comprising the step of determining said predetermined threshold by selecting a low threshold when a signal-to-noise ratio of said reference signal at the far end is low, updating said predetermined threshold upwards when said signal-to-noise ratio of said reference signal at the far end goes up and gradually reducing said predetermined threshold when said signal-to-noise ratio of the reference signal from the far end goes down.

* * * * *

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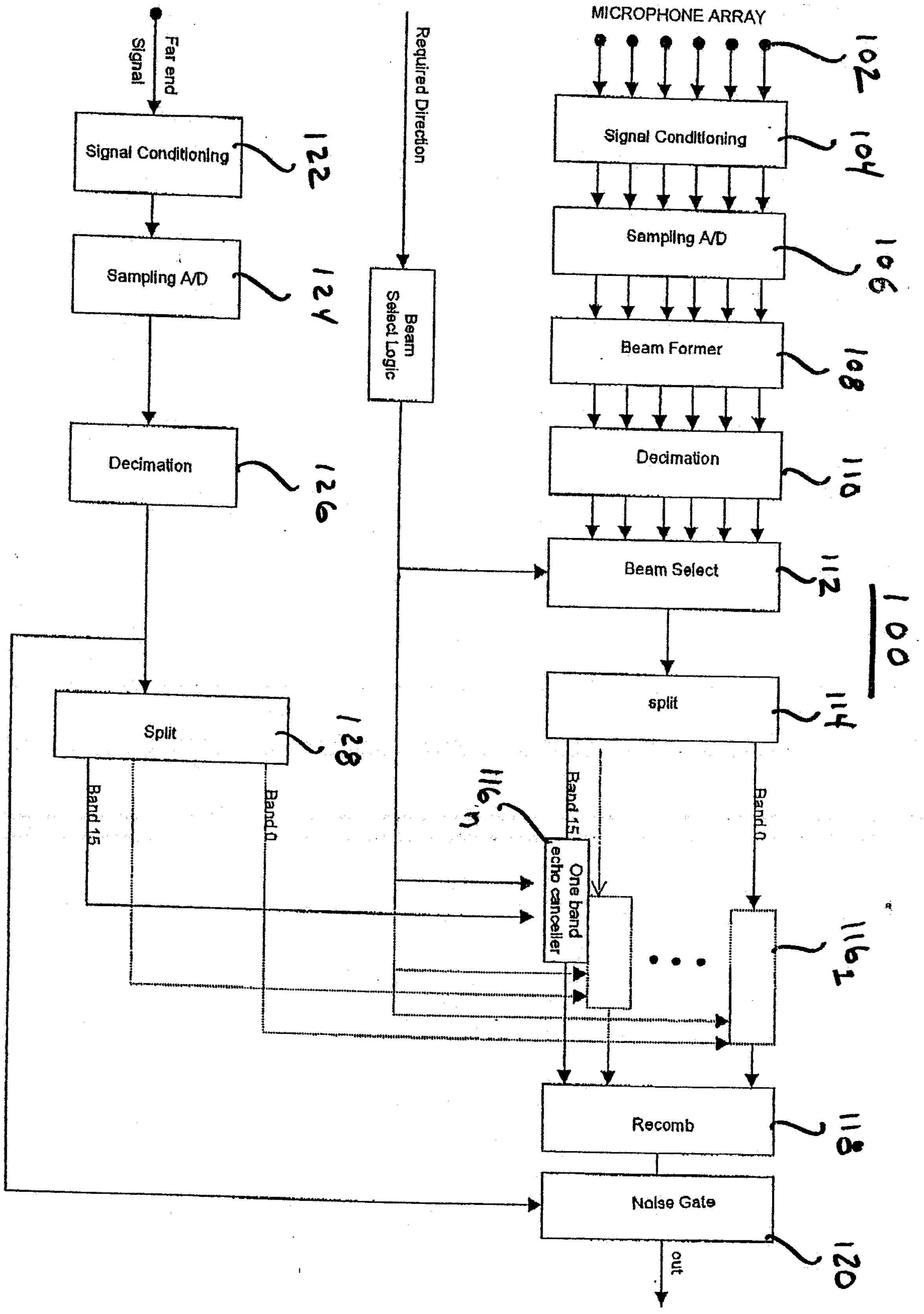

Fig. 1

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M fig

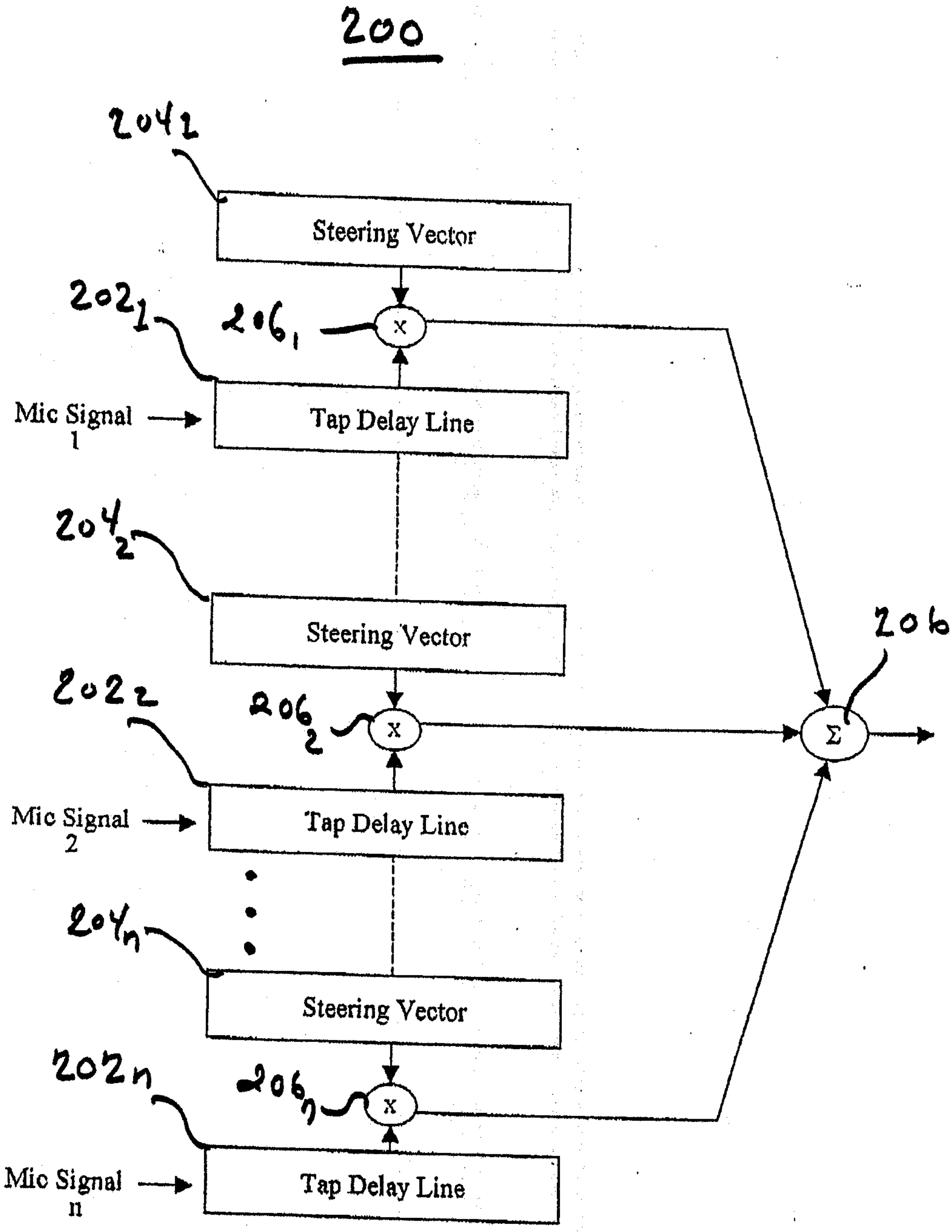

200
204₁
Steering Vector
202₁
206₁
x
Mic Signal 1
Tap Delay Line
204₂
Steering Vector
202₂
206₂
x
Mic Signal 2
Tap Delay Line
204ₙ
Steering Vector
202ₙ
206ₙ
x
Mic Signal n
Tap Delay Line
206
Σ

Fig. 2

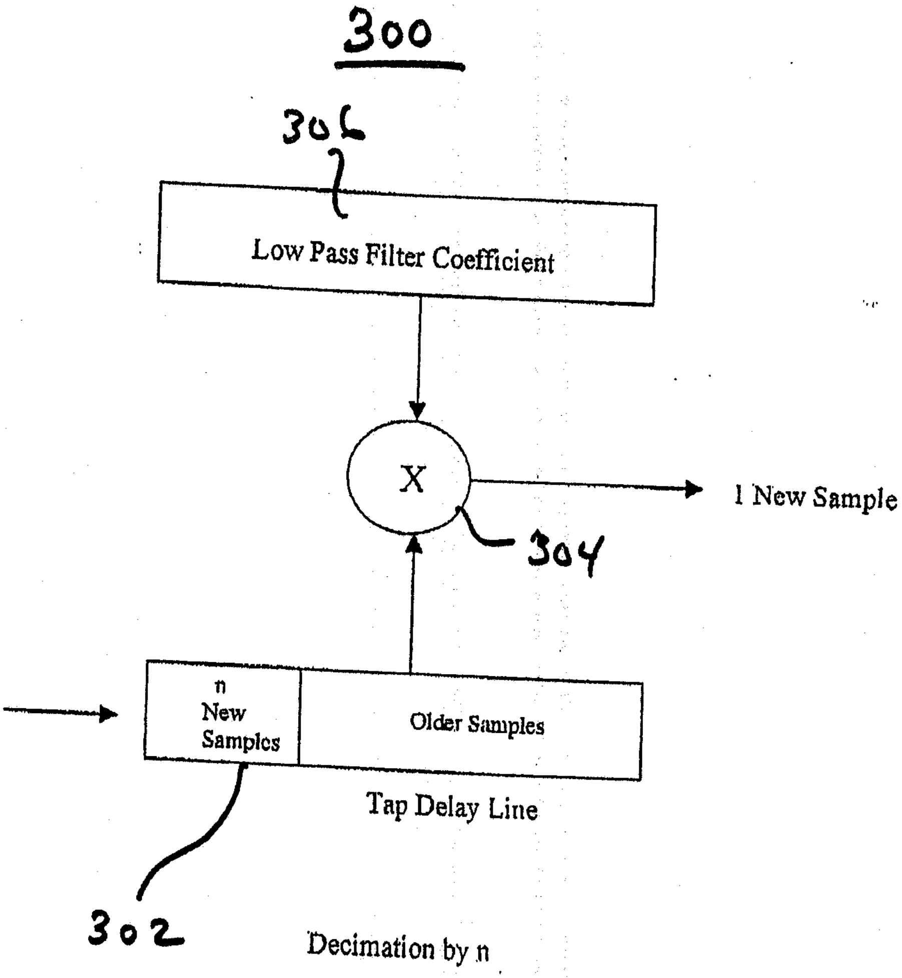
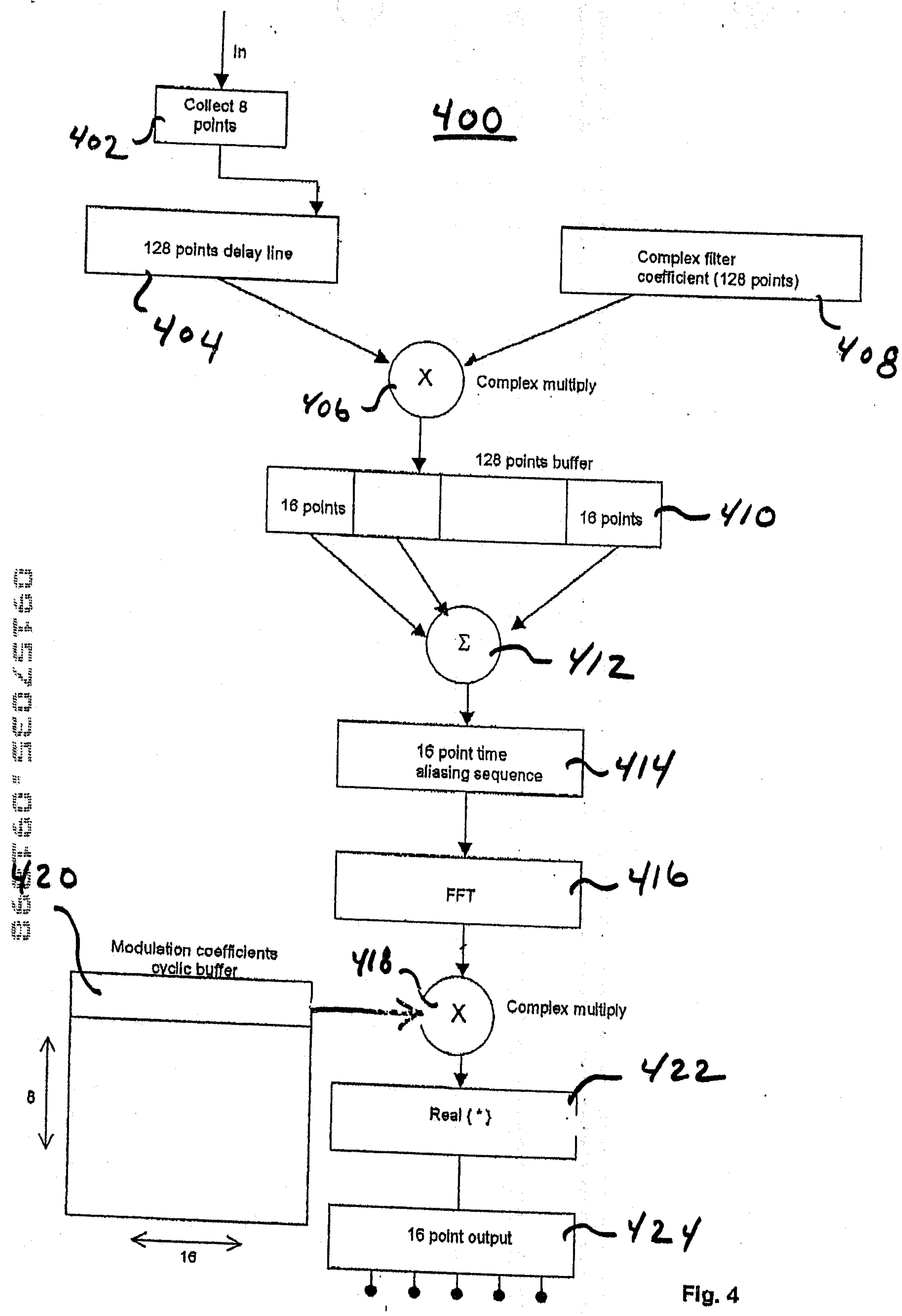

300
306
Low Pass Filter Coefficient
X
1 New Sample
304
n
New
Samples
Older Samples
Tap Delay Line
302
Decimation by n

Fig. 3



In
Collect 8 points
400
402
128 points delay line
404
Complex filter coefficient (128 points)
408
X
Complex multiply
406
128 points buffer
16 points
16 points
410
Σ
412
16 point time aliasing sequence
414
FFT
416
420
Modulation coefficients cyclic buffer
418
X
Complex multiply
8
Real (*)
422
16 point output
424
16
09595025 . 090402

Fig. 4

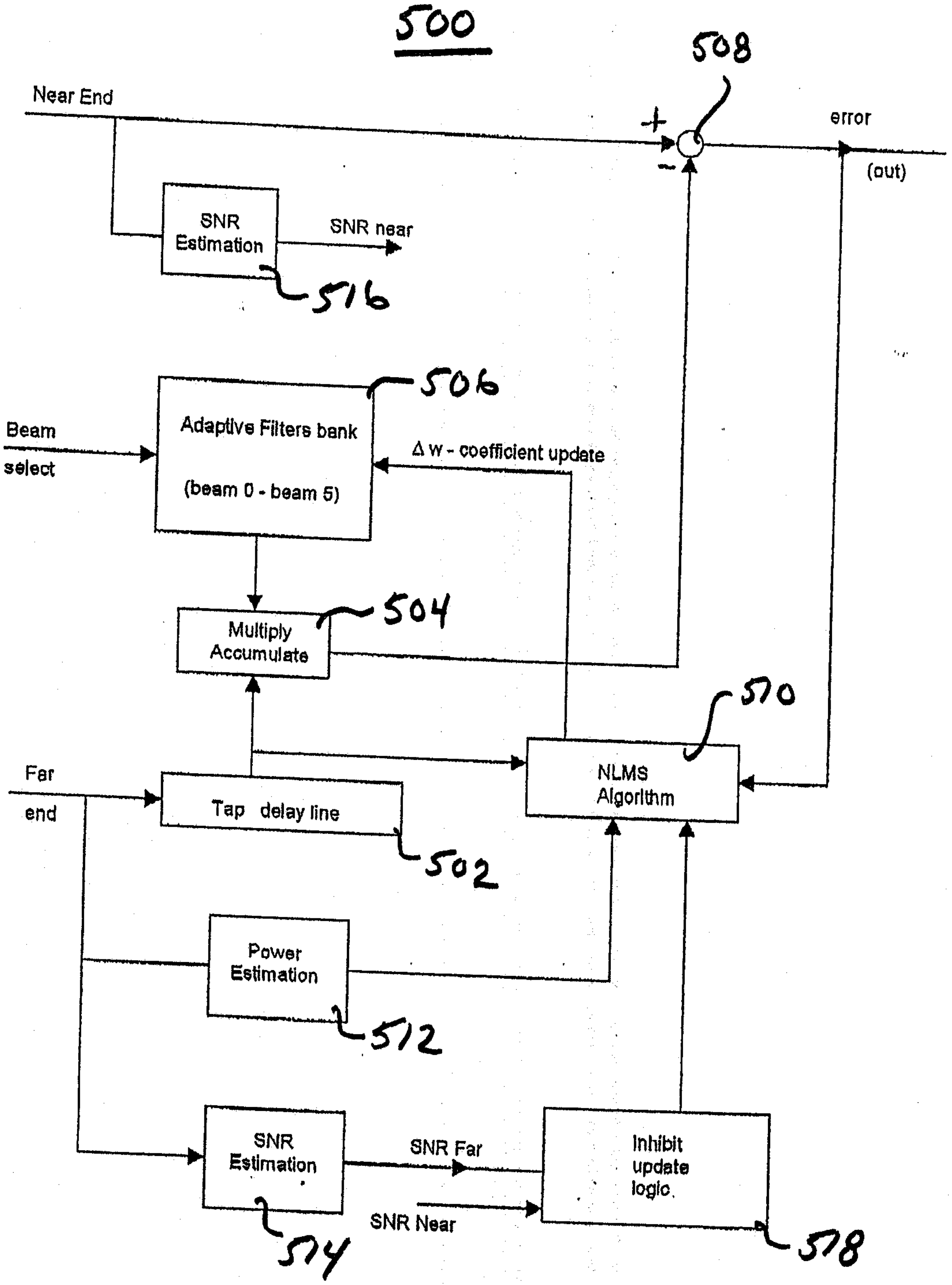

500
508
Near End
+
-
error
(out)
SNR Estimation
SNR near
516
506
Beam select
Adaptive Filters bank
(beam 0 - beam 5)
Δ w - coefficient update
504
Multiply Accumulate
510
NLMS Algorithm
Far end
Tap delay line
502
Power Estimation
512
SNR Estimation
SNR Far
SNR Near
Inhibit update logic.
514
518

Fig. 5

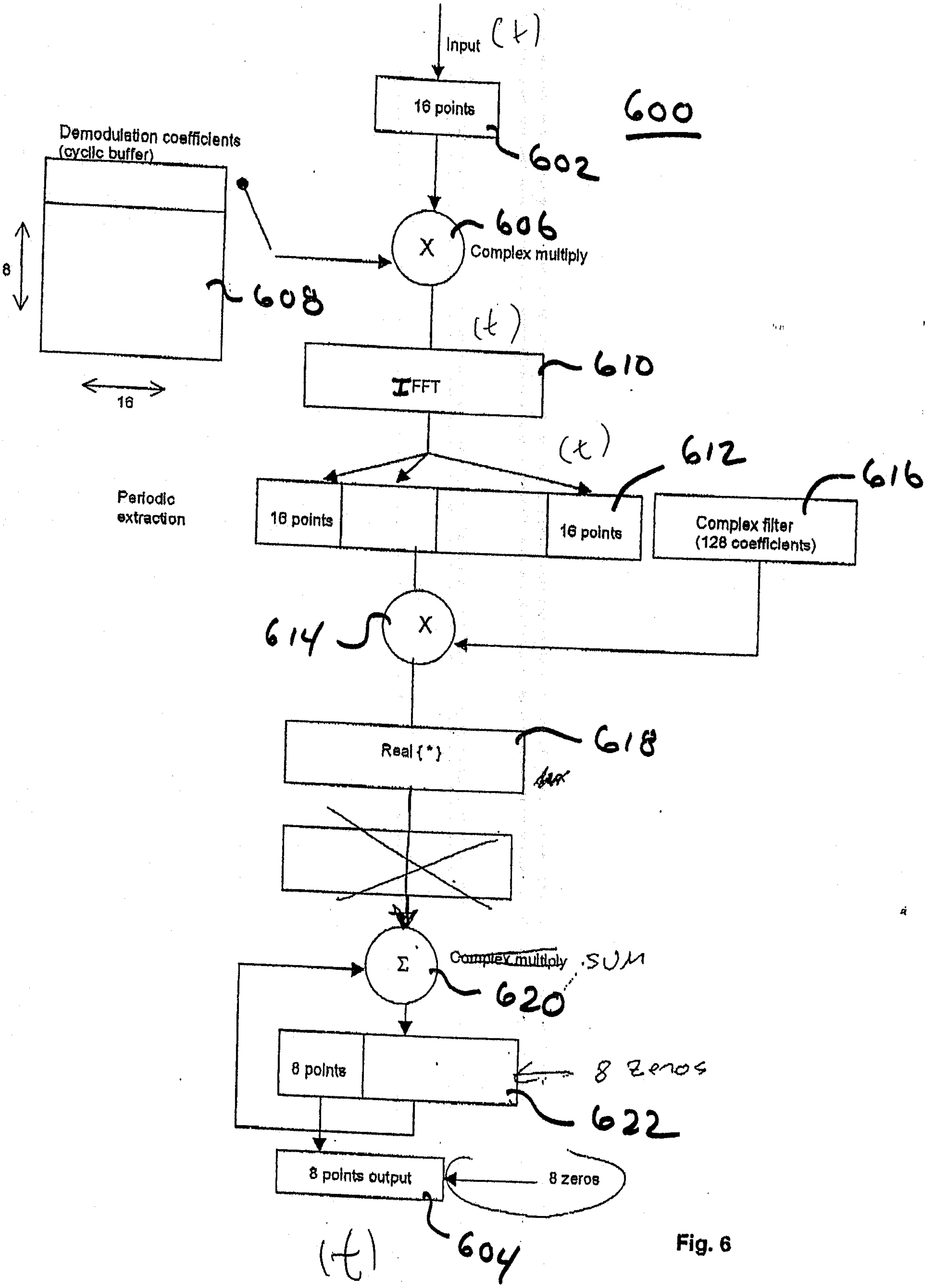


Fig. 6

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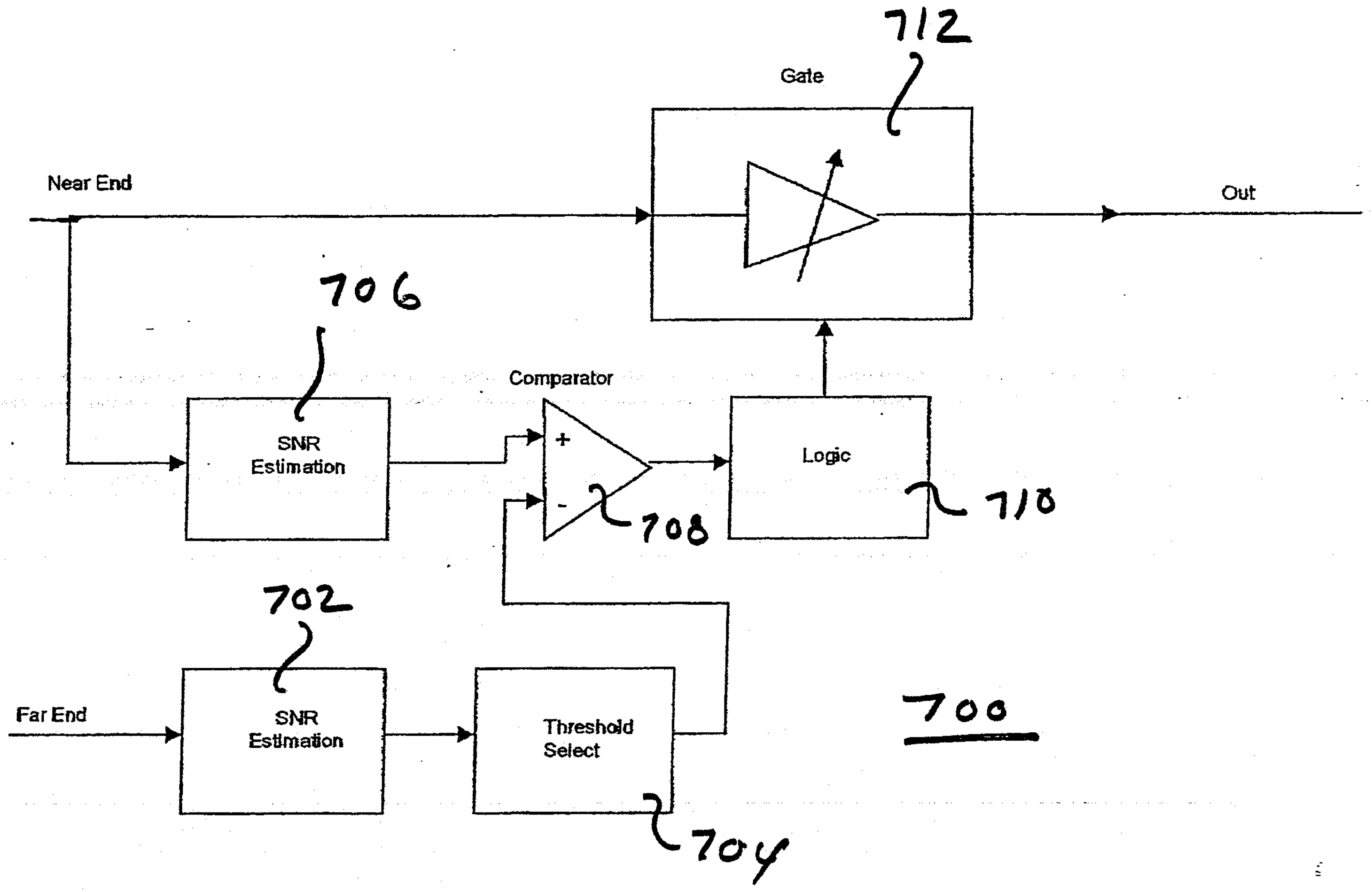
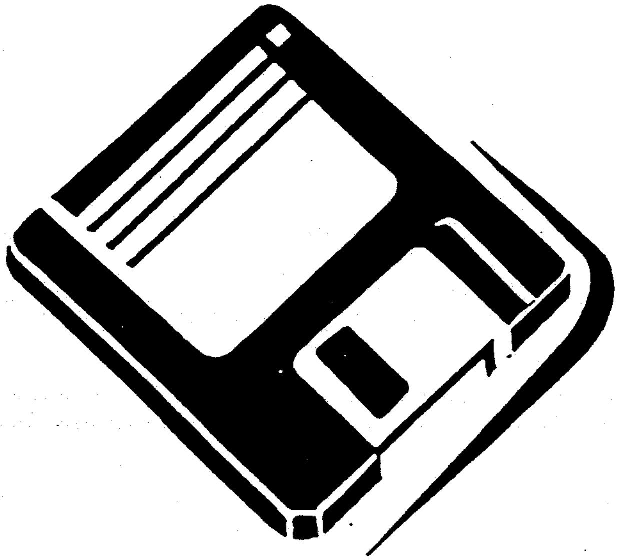

712
Gate
Near End
Out
706
Comparator
+
-
SNR
Estimation
Logic
708
710
702
Far End
SNR
Estimation
Threshold
Select
700
704

Fig. 7

PLEASE
DO NOT
SCAN



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PAPERS

PATENT APPLICATION FEE DETERMINATION RECORD

Effective October 1, 1997

Application or Docket Number
9/157635

CLAIMS AS FILED - PART I

SMALL ENTITY TYPE ☐ OR OTHER THAN SMALL ENTITY

FOR	NUMBER FILED (Column 1)	NUMBER EXTRA (Column 2)	RATE	FEE		RATE	FEE
BASIC FEE				395.00	OR		790.00
TOTAL CLAIMS	37 minus 20 =	* 17	x$11=		OR	x$22=	374
INDEPENDENT CLAIMS	3 minus 3 =	*	x41=		OR	x82=	
MULTIPLE DEPENDENT CLAIM PRESENT			+135=		OR	+270=	
			TOTAL		OR	TOTAL	1164

* If the difference in column 1 is less than zero, enter "0" in column 2

CLAIMS AS AMENDED - PART II

SMALL ENTITY OR OTHER THAN SMALL ENTITY

AMENDMENT A

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x41=		OR	x82=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+135=		OR	+270=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT B

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x41=		OR	x82=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+135=		OR	+270=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

AMENDMENT C

	CLAIMS REMAINING AFTER AMENDMENT (Column 1)		HIGHEST NUMBER PREVIOUSLY PAID FOR (Column 2)	PRESENT EXTRA (Column 3)	RATE	ADDITIONAL FEE		RATE	ADDITIONAL FEE
Total	*	Minus	**	=	x$11=		OR	x$22=	
Independent	*	Minus	***	=	x41=		OR	x82=	
FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM					+135=		OR	+270=	
					TOTAL ADDIT. FEE		OR	TOTAL ADDIT. FEE	

* If the entry in column 1 is less than the entry in column 2, write "0" in column 3.
** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20."
*** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3."
The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.

FORM PTO-875 (Rev. 8/97)

*U.S. Government Printing Office: 1997 - 430-571/69194

Patent and Trademark Office, U.S. DEPARTMENT OF COMMERCE

ISSUE SLIP STAPLE AREA (for additional cross references)

POSITION	INITIALS	ID NO.	DATE
FEE DETERMINATION	PS		9/24
O.I.P.E. CLASSIFIER		8	9-25-98
FORMALITY REVIEW		68151	10-1-98

INDEX OF CLAIMS

✓ .... Rejected
= .... Allowed
— (Through numeral) Canceled
÷ .... Restricted
N .... Non-elected
I .... Interference
A .... Appeal
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Claim		Date									
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381	92		
	94.1		
	91.2		
	94.7		
	144		
367	116		
	117, 118		
	119-127		
708	322		

INTERFERENCE SEARCHED

Class	Sub.	Date	Exmr.
Searched as above		12/3/99	Jey

SEARCH NOTES
(INCLUDING SEARCH STRATEGY)

	Date	Exmr.
Forester I Ser	12/3/99	√Jey
West Search	12/3/99	√Jey
areas suggested 367/116 t– 127		

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PATENT APPLICATION

09157035

jc558 U.S. PTO
09/157035
09/18/98

INITIALS ______

SEP 25 98 26

CONTENTS

	Date received (Incl. C. of M.) or Date Mailed
1. *Application* ______ *papers.*	
② Informal Ltr	10-13-98
3. Small Entity req letter	4/13/99
④ Ex's Amdt/A	12-21-99
5. Req. for C of C	3-19-14
6. Denied	3-19-14
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