PCT WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04M 9/08, H04B 3/20 — A1

(11) International Publication Number: **WO 99/26403**

(43) International Publication Date: 27 May 1999 (27.05.99)

(21) International Application Number: PCT/US98/24346

(22) International Filing Date: 13 November 1998 (13.11.98)

(30) Priority Data:
08/970,639 14 November 1997 (14.11.97) US

(71) Applicant *(for all designated States except US)*: TELLABS OPERATIONS, INC. [US/US]; 4951 Indiana Avenue, Lisle, IL 60532 (US).

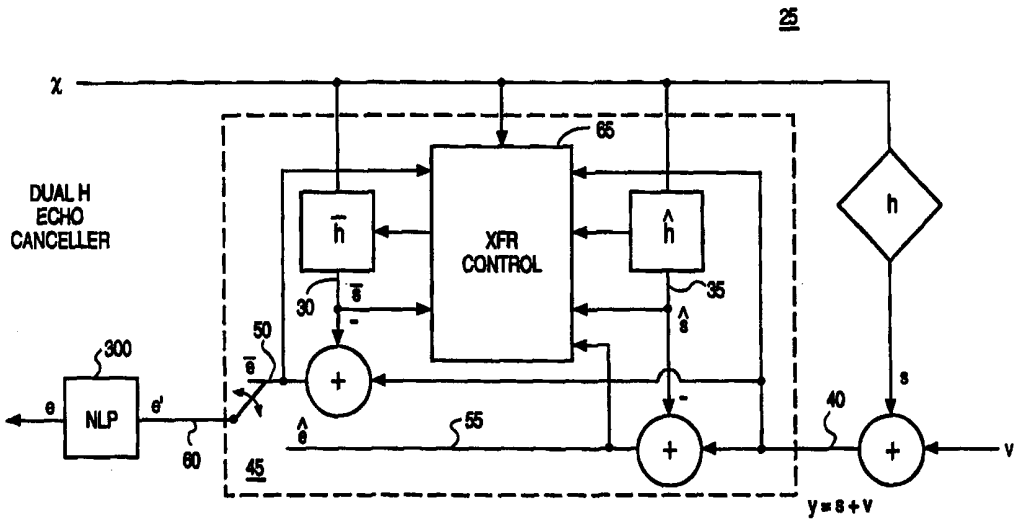
(72) Inventors; and
(75) Inventors/Applicants *(for US only)*: FARRELL, David, S. [US/US]; 4420 Basswood Drive, Lisle, IL 60532 (US). LIU, Tianfang [CN/US]; Apartment 305, 807 W. Taylor Street, DeKalb, IL 60115 (US).

(74) Agents: JARVIS, Lawrence, M. et al.; McAndrews, Held & Malloy, Ltd., Suite 3400, 500 West Madison Street, Chicago, IL 60661 (US).

(81) Designated States: AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CU, CZ, DE, DK, EE, ES, FI, GB, GE, GH, GM, HR, HU, ID, IL, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, UA, UG, US, UZ, VN, YU, ZW, ARIPO patent (GH, GM, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: ECHO CANCELLER HAVING IMPROVED NON-LINEAR PROCESSOR



(57) Abstract

A non-linear processor (300) for use in an echo canceller (25) is set forth. The non-linear processor (300) includes a center clipping digital filter receiving an echo compensated signal. The non-linear processor provides a center clipped output signal having non-linear thresholds at values of + T(sub NLP) and -T(sub NLP).The value of T(sub NLP) is dynamically dependent, at least, in part on echo return loss measurements. To limit the processor's susceptibility to corruption from double-talk conditions, the non-linear processor (300) inhibits the dynamic setting of the T(sub NLP) value when a double talk condition is present. Additionally, or in the alternative, the non-linear processor (300) locks the value of the echo return loss measurement after a predetermined number of consecutive echo return loss measurements have values falling within a predetermined range of one another. Such locking further reduces the susceptibility of the non-linear processor to corruption from double-talk conditions.

FOR THE PURPOSES OF INFORMATION ONLY

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AL	Albania	**ES**	Spain	**LS**	Lesotho	**SI**	Slovenia
AM	Armenia	**FI**	Finland	**LT**	Lithuania	**SK**	Slovakia
AT	Austria	**FR**	France	**LU**	Luxembourg	**SN**	Senegal
AU	Australia	**GA**	Gabon	**LV**	Latvia	**SZ**	Swaziland
AZ	Azerbaijan	**GB**	United Kingdom	**MC**	Monaco	**TD**	Chad
BA	Bosnia and Herzegovina	**GE**	Georgia	**MD**	Republic of Moldova	**TG**	Togo
BB	Barbados	**GH**	Ghana	**MG**	Madagascar	**TJ**	Tajikistan
BE	Belgium	**GN**	Guinea	**MK**	The former Yugoslav Republic of Macedonia	**TM**	Turkmenistan
BF	Burkina Faso	**GR**	Greece			**TR**	Turkey
BG	Bulgaria	**HU**	Hungary	**ML**	Mali	**TT**	Trinidad and Tobago
BJ	Benin	**IE**	Ireland	**MN**	Mongolia	**UA**	Ukraine
BR	Brazil	**IL**	Israel	**MR**	Mauritania	**UG**	Uganda
BY	Belarus	**IS**	Iceland	**MW**	Malawi	**US**	United States of America
CA	Canada	**IT**	Italy	**MX**	Mexico	**UZ**	Uzbekistan
CF	Central African Republic	**JP**	Japan	**NE**	Niger	**VN**	Viet Nam
CG	Congo	**KE**	Kenya	**NL**	Netherlands	**YU**	Yugoslavia
CH	Switzerland	**KG**	Kyrgyzstan	**NO**	Norway	**ZW**	Zimbabwe
CI	Côte d'Ivoire	**KP**	Democratic People's Republic of Korea	**NZ**	New Zealand		
CM	Cameroon			**PL**	Poland		
CN	China	**KR**	Republic of Korea	**PT**	Portugal		
CU	Cuba	**KZ**	Kazakstan	**RO**	Romania		
CZ	Czech Republic	**LC**	Saint Lucia	**RU**	Russian Federation		
DE	Germany	**LI**	Liechtenstein	**SD**	Sudan		
DK	Denmark	**LK**	Sri Lanka	**SE**	Sweden		
EE	Estonia	**LR**	Liberia	**SG**	Singapore		

TITLE OF THE INVENTION

ECHO CANCELLER HAVING IMPROVED NON-LINEAR PROCESSOR

CROSS-REFERENCE TO RELATED APPLICATIONS

The following applications, filed on even date, herewith, are incorporated by reference: USSN _____, (11724US01), "Echo Canceller Employing Dual-H Architecture Having Improved Coefficient Transfer"; USSN _____, (11998US01), "Echo Canceller Employing Dual-H Architecture Having Improved Double-Talk Detection"; USSN _____, (11999US01), "Echo Canceller Employing Dual-H Architecture Having Improved Non-Linear Echo Path Detection"; USSN _____, (Attorney Docket No. 12000US01), "Echo Canceller Employing Dual-H Architecture Having Variable Adaptive Gain Settings"; USSN _____, (Attorney Docket No. 12002US01), "Echo Canceller Employing Dual-H Architecture Having Split Adaptive Gain Settings."

STATEMENT REGARDING FEDERALLY SPONSORED RESEARCH OR DEVELOPMENT

Not Applicable

BACKGROUND OF THE INVENTION

Long distance telephone facilities usually comprise four-wire transmission circuits between switching offices in different local exchange areas, and two-wire circuits within each area connecting individual subscribers with the switching office. A call between subscribers in different exchange areas is carried over a two-wire circuit in each of the areas and a four-wire circuit between the areas, with conversion of speech energy between the two and four-wire circuits being effected by hybrid circuits. Ideally, the hybrid circuit input ports perfectly match the impedances of the two and four-wire circuits, and its balanced network impedance perfectly matches the impedance of the two-wire circuit. In this manner, the signals transmitted from one exchange area to the other will not be reflected or returned to the one area as echo. Unfortunately, due to impedance differences which inherently exist between two and four-wire circuits, and because impedances must be matched at each frequency in the voice band, it is virtually impossible for a given hybrid circuit to perfectly match the impedances of any particular two and four-wire transmission circuit. Echo is, therefore, characteristically part of a long distance telephone system.

Although undesirable, echo is tolerable in a telephone system so long as the time delay in the echo path is relatively short, for example, shorter than about 40 milliseconds. However, longer echo delays can be distracting or utterly confusing to a far end speaker, and to reduce the same to a tolerable level an echo canceller may be used toward each of the path to cancel echo which otherwise would return to the far end speaker. As is known, echo cancellers monitor the signals on the

receive channel of a four-wire circuit and generate estimates of the actual echos expected to return over the transmit channel. The echo estimates are then applied to a subtractor circuit in the transmit channel to remove or at least reduce the actual echo.

In simplest form, generation of an echo estimate comprises obtaining individual samples of the signal on the receive channel, convolving the samples with the impulse response of the system and then subtracting, at the appropriate time, the resulting products or echo estimates from the actual echo on the transmit channel. In actual practice generation of an echo estimate is not nearly so straightforward.

Transmission circuits, except those which are purely resistive, exhibit an impulse response that has amplitude and phase dispersive characteristics that are frequency dependent, since phase shift and amplitude attenuation vary with frequency. To this end, a suitable known technique for generating an echo estimate contemplates manipulating representations of a plurality of samples of signals which cause the echo and samples of impulse responses of the system through a convolution process to obtain an echo estimate which reasonably represents the actual echo expected on the echo path. One such system is illustrated in FIG. 1.

In the system illustrated in FIG. 1, a far end signal x from a remote telephone system is received locally at line 10. As a result of the previously noted imperfections in the local system, a portion of the signal x is echoed back to the remote site at line 15 along with the signal v from the local telephone system. The echo response is illustrated here as a signal s corresponding to the following equation:

$$s = x * h$$

where h is the impulse response of the echo characteristics. As such, the signal sent from the near end to the far end, absent echo cancellation, is the signal y, which is the sum of the telephone signal v and the echo signal s. This signal is illustrated as y at line 15 of FIG. 1.

To reduce and/or eliminate the echo signal component s from the signal y, the system of FIG. 1 uses an echo canceller having an impulse response filter $\bar{h}$ that is the estimate of the impulse echo response h. As such, a further signal $\bar{s}$ representing an estimate of the echo signal s is generated by the echo canceller in accordance with the following equation:

$$\bar{s} = \bar{x} * h$$

The echo canceller subtracts the echo estimate signal $\bar{s}$ from the signal y to generate a signal e at line 20 that is returned to the far end telephone system. The signal e thus corresponds to the following equation:

$$e = s + v - \bar{s} \approx v$$

As such, the signal returned to the far end station is dominated by the signal v of the near end telephone system. As the echo impulse response $\bar{h}$ more closely correlates to the actual echo response h, then $\bar{s}$ more closely approximates s and thus the magnitude of the echo signal component s on the signal e is more substantially reduced.

The echo impulse response model $\bar{h}$ may be replaced by an adaptive digital filter having an impulse response $\hat{h}$. Generally, the tap coefficients for such an

adaptive response filter are found using a technique known as Normalized Least Mean Squares adaptation.

Although such an adaptive echo canceller architecture provides the echo canceller with the ability to readily adapt to changes in the echo path response h, it is highly susceptible to generating sub-optimal echo cancellation responses in the presence of "double talk" (a condition that occurs when both the speaker at the far end and the speaker at the near end are speaking concurrently as determined from the viewpoint of the echo canceller).

To reduce this sensitivity to double-talk conditions, it has been suggested to use both a non-adaptive response and an adaptive response filter in a single echo canceller. One such echo canceller is described in USPN 3,787,645, issued to Ochiai et al on January 22, 1974. Such an echo canceller is now commonly referred to as a dual-H echo canceller.

Another problem confronting echo canceller circuits is the possibility that the echo path response is non-linear. Such non-linear echo paths are often present in, for example, cellular telephone systems. The echo canceller must not only detect the non-linear echo response condition, it must also be able to reduce the effects of the non-linear response. The present inventors have recognized that the dual-H architecture may itself be employed to assist in detecting a non-linear echo path to thereby signal the echo canceller of the condition so that the echo canceller may respond in the appropriate manner. Further, the effects of the non-linearities are substantially reduced using a unique non-linear processor. The non-linear processor counters echo due to non-linear echo paths. Further, it also eliminates residual echo

due to limitations of the system (e.g., quantization error) and minor inhibitors (e.g., low levels of background noise).

BRIEF SUMMARY OF THE INVENTION

A non-linear processor for use in an echo canceller is set forth. The non-linear processor includes a center clipping digital filter receiving an echo compensated signal. The non-linear processor provides a center clipped output signal having non-linear thresholds at values of $+T_{NLP}$ and $-T_{NLP}$. The value of T_{NLP} is dynamically dependent, at least in part, on echo return loss measurements. To limit the processor's susceptibility to corruption from double-talk conditions, the non-linear processor inhibits the dynamic setting of the T_{NLP} value when a double-talk condition is present. Additionally, or in the alternative, the non-linear processor locks the value of the echo return loss measurement after a predetermined number of consecutive echo return loss measurements have values falling within a predetermined range of one another. Such locking further reduces the susceptibility of the non-linear processor to corruption from double-talk conditions.

BRIEF DESCRIPTION OF THE SEVERAL VIEWS OF THE DRAWINGS

Figure 1 is a block diagram of a conventional canceller.

Figure 2 is a schematic block diagram of an echo canceller that operates in accordance with one embodiment of the present invention.

Figure 3 is a flow chart illustrating one manner of carrying out coefficient transfers in accordance with one embodiment of the present invention.

Figure 4 is a flow chart illustrating a further manner of carrying out coefficient transfers in accordance with a further embodiment of the present invention.

Figure 5 illustrates the input and output of a filter that may be used to assist in measuring transfer density.

Figure 6 illustrates one manner of implementing non-linear echo path detection in the echo canceller operations previously described in connection with Figure 4.

Figure 7 illustrates tap coefficient time dispersion for the adaptive filter when responding to a linear echo path.

Figure 8 illustrates tap coefficient time dispersion for the adaptive filter when responding to a non-linear echo path.

Figure 9 illustrates one manner of implementing a non-linear detector using time dispersion as applied to the echo canceller operations of Figure 4.

Figure 10 is a graph of an input signal to the output signal of one embodiment of the non-linear processor.

Figures 11a and 11b illustrate the response of the non-linear processor to small signal and large signal inputs respectively.

Figure 12 illustrates various quantities from which T_{NLP} may be calculated.

Figure 13 illustrates one manner of implementing a double talk detector in the non-linear processor.

Figure 14 illustrates a signal input e' and the corresponding output signals from filters S, M, and L of Figure 13.

Figure 15 illustrates one manner of implementing an echo canceller system employing the present invention.

DETAILED DESCRIPTION OF THE INVENTION

Figure 2 illustrates one embodiment of a dual-h echo canceller suitable for use in implementing the present invention. As illustrated, the echo canceller, shown generally at 25, includes both a non-adaptive filter $\bar{h}$ and an adaptive filter $\hat{h}$ to model the echo response h. Each of the filters $\bar{h}$ and $\hat{h}$ are preferably implemented as digital filters, such as finite impulse response (FIR) filters comprising a plurality of taps each having a corresponding tap coefficient. This concept may be extended to IIR filters as well. If FIR filters are used, the duration of each of the FIR filters should be sufficient to cover the duration of the echo response of the channel in which the echo canceller 25 is disposed.

The output of the non-adaptive filter $\bar{h}$ is available at the line 30 while the output of the adaptive filter $\hat{h}$ is available at line 35. Each of the signals at lines 30 and 35 are subtracted from the signal-plus-echo signal of line 40 to generate echo compensated signals at lines 50 and 55, respectively. A switch 45, preferably a software switch, may be used to selectively provide either the output signal at the line 50 or the output signal at line 55 to the echo canceller output at line 60. The switch 45 may be used to provide the echo compensation based on the $\hat{h}$ filter during convergence and then be switched to provide the echo compensation based on the $\bar{h}$ filter after covergence. Further, the switch 45 is directed to provide the echo compensation based on the $\bar{h}$ filter in response to the detection of a double-talk condition.

A transfer controller 65 is used to transfer the tap coefficients of filter $\hat{h}$ to replace the tap coefficients of filter $\bar{h}$. As illustrated, the transfer controller 65 is connected to receive a number of system input signals. Of particular import with respect to the present invention, the transfer controller 65 receives the signal-plus-echo response y and each of the echo canceller signals $\bar{e}$ and $\hat{e}$ at lines 50 and 55, respectively. The transfer controller 65 is preferably implemented in the software of one or more digital signal processors used to implement the echo canceller 25.

As noted above, the art is substantially deficient of teachings with respect to the manner in which and conditions under which a transfer of tap coefficients from $\hat{h}$ to $\bar{h}$ is to occur. The present inventors have implemented a new process and, as such, a new echo canceller in which tap coefficient transfers are only made by the transfer controller 65 when selected criterion are met. The resulting echo canceller 25 has substantial advantages with respect to reduced double-talk sensitivity and increased double-talk detection capability. Further, it ensures a monotonic improvement in the estimates $\bar{h}$.

The foregoing system uses a parameter known as echo-return-loss-enhancement (ERLE) to measure and keep track of system performance. Two ERLE parameter values are used in the determination as to whether the transfer controller 65 transfers the tap coefficients from $\hat{h}$ to $\bar{h}$. The first parameter, $\bar{E}$, is defined in the following manner:

$$E = \frac{y}{\bar{e}}$$

Similarly, the parameter $\hat{E}$ is defined as follows:

$$\hat{E} = \frac{y}{\hat{e}}$$

Each of the values $\hat{E}$ and $\overline{E}$ may also be averaged over a predetermined number of samples to arrive at averaged $\hat{E}$ and $\overline{E}$ values used in the system for the transfer determinations.

Figure 3 illustrates one manner of implementing the echo canceller 25 using the parameters $\hat{E}$ and $\overline{E}$ to control tap coefficients transfers between filter $\hat{h}$ to $\overline{h}$. As illustrated, the echo canceller 25 provides a default $\overline{h}$ set of coefficients at step 80 during the initial portions of the call. After the tap coefficients values for $\overline{h}$ have been set, a measure of $\overline{E}$ is made at step 85 to measure the performance of the tap coefficient values of filter $\overline{h}$.

After the initialization sequence of steps 80 and 85, or concurrent therewith, the echo canceller 25 begins and continues to adapt the coefficients of $\hat{h}$ to more adequately match the echo response h of the overall system. As noted in Figure 3, this operation occurs at step 90. Preferably, the adaptation is made using a Normalized Least Mean Squares method, although other adaptive methods may also be used (e.g., LMS and RLS).

After a period of time has elapsed, preferably, a predetermined minimum period of time, the echo canceller 25 makes a measure of $\hat{E}$ at step 95. Preferably, this measurement is an averaged measurement. At step 100, the echo canceller 25 compares the value of $\hat{E}$ with the value of $\overline{E}$. If the value of $\hat{E}$ is greater than the

value of $\overline{E}$, the tap coefficients of filter $\hat{h}$ are transferred to replace the tap coefficients of filter $\overline{h}$ at step 105. If this criterion is not met, however, the echo canceller 25 will continue to adapt the coefficients of the adaptive filter $\hat{h}$ at step 90, periodically measure the value of $\hat{E}$ at 95, and make the comparison of step 100 until the condition is met.

Although not illustrated, other transfer conditions may be imposed in addition to the foregoing. For example, the echo canceller may impose a requirement that a far end signal exist before a transfer may occur. Additionally, transfers may be inhibited during a double-talk condition. Further conditions may also be imposed based on system requirements.

If the echo canceller 25 finds that $\hat{E}$ is greater than $\overline{E}$, the above-noted transfer takes place. Additionally, the echo canceller 25 stores the value of $\hat{E}$ as a value E_{max}. This operation is depicted at step 110 of the Figure 3. The value of E_{max} is thus the maximum value of ERLE that occurs over the duration of the call and at which a transfer has taken place. This further value is used thereafter, in addition to the $\hat{E}$ and $\overline{E}$ comparison, to control whether the tap coefficients of $\hat{h}$ are transferred by the transfer controller 65 to replace the tap coefficients of $\overline{h}$. This further process is illustrated that steps 115, 120, and 125 of Figure 3. In each instance, the tap coefficient transfer only occurs when both of the following two conditions are met: 1) $\hat{E}$ is greater than the current $\overline{E}$, and 2) $\hat{E}$ is greater than E_{max}. Each time that both criteria are met, the transfer controller 65 of echo

canceller 25 executes the tap coefficient transfer and replaces the previous E_{max} value with the current $\hat{E}$ value for future comparison.

Requiring that $\hat{E}$ be greater than any $\overline{E}$ value used over the course of the call before the coefficient transfer takes place has two beneficial and desirable effects. First, each transfer is likely to replace the prior tap coefficients of filter $\overline{h}$ with a better estimate of the echo path response. Second, this transfer requirement increases the double-talk protection of the echo canceller system. Although it is possible to have positive ERLE $\hat{E}$ during double-talk, the probability that $\hat{E}$ is greater than E_{max} during double-talk decreases as the value of E_{max} increases. Thus an undesirable coefficient transfer during double-talk becomes increasingly unlikely as the value of E_{max} increases throughout the duration of the call.

The echo canceller 25 may impose both an upper boundary and a lower boundary on the value of E_{max}. For example, E_{max} may have a lower bounded value of 6 dB and an upper bounded value of 24 dB. The purpose of the lower bound is to prevent normal transfers during double-talk conditions. It has been shown in simulations using speech inputs that during double-talk, a value of greater than 6 dB ERLE was a very low probability event, thus making it an appropriate value for the initial value of E_{max}. The upper bound on E_{max} is used to prevent a spuriously high measurement from setting E_{max} to a value at which further transfers become impossible.

The value of E_{max} should be set to, for example, the lower bound value at the beginning of each call. Failure to do so will prevent tap coefficient transfers on a new call until the echo cancellation response of the echo canceller 25 on the new

call surpasses the quality of the response existing at the end of the prior call. However, this criterion may never be met during the subsequent call thereby causing the echo canceller 25 to operate using sub-optimal tap coefficients values. Resetting the E_{max} value to a lower value increases the likelihood that a tap coefficient transfer will take place and, thereby, assists in ensuring that the $\bar{h}$ filter uses tap coefficients for echo cancellation that more closely correspond to the echo path response of the new call.

One manner of implementing the E_{max} value change is illustrated in the echo canceller operations flow-chart of Figure 4. When all transfer conditions are met except $\hat{E}$ greater than E_{max}, and this condition persists for a predetermined duration of time, the echo canceller 25 will reset the E_{max} value to, for example, the lower bound value. In the exemplary operations shown in Figure 4, the echo canceller 25 determines whether $\hat{E}$ is greater than the lower bound of E_{max} at step 140 and less than the current value of E_{max} at step 145. If both of these conditions remain true for a predetermined period of time as determined at step 150, and all other transfer criterion have been met, the echo canceller 25 resets the E_{max} value to a lower value, for example, the lower bound of the E_{max} value, at step 155. This lowering of the E_{max} value increases the likelihood of a subsequent tap coefficient transfer.

Choosing values for the lower and upper bound of E_{max} other than 6 dB and 24 dB, respectively, is also possible in the present system. Choosing a lower bound of E_{max} smaller than 6 dB provides for a relatively prompt tap coefficient transfer after a reset operation or a new call, but sacrifices some double-talk protection. A

value greater than 6 dB, however, inhibits tap coefficient transfer for a longer period of time, but increases the double-talk immunity of the echo canceller. Similarly, varying the value of the predetermined wait time T before which E_{max} is reset may also be used to adjust echo canceller performance. A shorter predetermined wait time T produces faster reconvergence transfers, but may sacrifice some double-talk immunity. The opposite is true for larger predetermined wait time values.

A further modification of the foregoing echo canceller system relates to the value stored as E_{max} at the instant of tap coefficient transfer. Instead of setting E_{max} equal to the $\hat{E}$ value at the transfer instant, E_{max} may be set to a value equal to the value of $\hat{E}$ minus a constant value (e.g., one, three, or 6 dB). At no time, however, should the E_{max} value be set to a value that is below the lower bound value for E_{max}. Additionally, a further condition may be imposed in that a new softened E_{max} is not less than the prior value of E_{max}. The foregoing "softening" of the E_{max} value increases the number of transfers that occur and, further, provides more decision-making weight to the condition of $\hat{E}$ being larger than $\overline{E}$.

As will be readily recognized, the echo canceller of the present invention may be implemented in a wide range of manners. Preferably, the echo canceller system is implemented using one or more digital signal processors to carry out the filter and transfer operations. Digital-to-analog conversions of various signals are carried out in accordance with known techniques for use by the digital signal processors.

There are some circumstances when the foregoing transfer criterion should be defeated. For example, the transfer criterion is preferably defeated when 1) the long-term *ERLE* remains low, and 2) a small but measurable performance advantage of $\hat{h}$ over $\bar{h}$ is sustained over a long period of time.

One case in which it should be defeated is when the steady-state *ERLE* remains below the lower value of $E_{\max}$. Such a case may occur when there is a high-level, constant background noise entering from the near-end which may lower the measured *ERLE* considerably. Since the foregoing process prevents transfers from occurring unless the *ERLE* is greater than the lower bound of $E_{\max}$, no transfers are possible in low *ERLE* situations. Since the $\bar{h}$ may contain the solution to a previous call at the start of a new, low *ERLE* call, defeating the foregoing transfer criterion is preferable in some cases.

The first condition for defeating the foregoing transfer criterion is a sustained low *ERLE* measurement over a relatively long period of time (e.g., 150 to 500 msec) of adaptation. Since a low *ERLE* call will tend to have a smaller difference between the *ERLEs* of $\bar{h}$ and $\hat{h}$ (a 1 dB difference may be the largest difference observed), the required *ERLE* difference between $\hat{h}$ and $\bar{h}$ for a transfer to occur should be reduced (e.g. to 0 or 1 dB) once the long-term ERLE is confirmed to be low. To compensate, a requirement may be imposed whereby the small ERLE difference between $\bar{h}$ and $\hat{h}$ is maintained for a long period of time (e.g. 75 to 200 msec) before the transfer is allowed.

Each of the filters $\hat{h}$ and $\bar{h}$ preferably include one or more D.C. taps to compensate for non-linearities in the echo path response. The result is that the filters can model a D.C. shift quite accurately, and high ERLE can be achieved despite the presence of a D.C. non-linearity.

More complex non-linearities in the echo path response generally require more complex non-linear processing. One manner in which the echo canceller 25 can process such non-linearities is set forth in U.S.S.N. ___, titled _____ (Attorney Docket No. ____), the teachings of which are hereby incorporated by reference. A further manner in which the echo canceller may compensate for non-linearities is by directing switch 45 to use the $\hat{h}$ filter to cancel the echo. This is due to the fact that the $\hat{h}$ filter is more time responsive to the non-linearities since the NLMS adaptation process attempts to find the best short term solutions for the non-linearities in the echo path so as to maximize *ERLE* even where the short term solutions diverge from the long-term linear response.

The present inventors have recognized that a difference in the number of coefficient transfers from $\hat{h}$ to $\bar{h}$ occurs depending on whether or not the echo path has a non-linear component. When the echo path has an entirely linear response, the tap coefficients of $\hat{h}$ generally reflect the linear impulse response of the echo path. In a non-linear echo path having a non-linear residual echo signal, the residual echo will tend to increase when compared to the linear response. In an attempt to reduce this residual echo, the tap coefficients of $\hat{h}$ move, for short periods of time, away from the linear impulse response to maximize the short-term

ERLE. This occurs since the tap coefficients are adapted using a Least Mean Squares adaptation process which constantly tries to minimize the short term *ERLE*. As the non-linearity of the echo signal increases, the effect of the tap coefficients moving away from their original linear position to gain short term improvements becomes more pronounced.

For linear echo paths, the operational logic of the echo canceller 25 assumes that a number of transfers will occur at the beginning of the call as the echo canceller 25 adjusts the taps to the model echo path response and transfers that model to the $\bar{h}$ filter. After the initial convergence period, the expected number of transfers per unit time will become small if the echo path is truly non-linear. Conversely, the best cancellation solution varies with time for non-linear calls when using a linear echo canceller. As a result, the non-adaptive taps of the $\bar{h}$ filter quickly become sub-optimal when compared to the $\hat{E}$ value of the $\hat{h}$ filter response. For example, the $\bar{h}$ coefficients may become sub-optimal as compared to $\hat{h}$ within about 50 milliseconds. As a result, the transfer density, i.e. the number of transfers per unit time, becomes large and continues to stay large throughout the entire call.

With foregoing in mind, the echo canceller 25 measures the transfer density after a convergence period to determine whether the echo path response is linear or non-linear. To this end, the echo canceller 25 is programmed to store a transfer density threshold value TDT. The echo canceller 25 maintains a count of the number of transfers TC that occur over a known period of time. This count is compared to the transfer density threshold value TDT. If this count TC, and thus

the transfer density, exceeds the transfer density threshold value TDT, the echo canceller 25 declares the presence of a non-linear echo path response and executes the appropriate operations needed to cancel the echo signal. If the count TC, and thus the transfer density, is below the threshold value TDT, the echo canceller 25 handles the echo cancellation as a linear echo cancellation.

The value of TC may be calculated in a number of different manners. It may be calculated using a software counter that is read on a predetermined periodic basis at intervals T and that it is reset immediately after being read. The resulting count may be directly used and compared to the transfer density threshold value TDT. In such instances, the value of the transfer density threshold value TDT is selected based on the period used to read the software counter.

In a more complicated process for determining the value of TC, the time at which a software counter is read and the corresponding counter value are stored in memory. During a subsequent reading of the counter value, both the counter value and the time at which the counter is subsequently read are noted. The value of TC may then be calculated as the difference between the initial and final counter values divided by the difference between the initial and final time values.

The value of TC may also correspond to an averaged transfer density value. In such instances, the value TC may be calculated using a digital filter which calculates a moving average of the number of transfers per unit time. Such a filter preferably has, for example, a slow attack time constant and fast decay time constant, although this is not mandatory. A value of 1 is supplied to the averaging filter each time a transfer takes place during a frame having a predetermined period.

A value of 0 is supplied to the averaging filter each time a transfer does not take place during the frame. When the output of the averaging filter exceeds the threshold value TDV, a non-linear condition is declared.

Such a situation is illustrated in Figure 5. As shown, a value of 1 is applied to the filter input during each frame in which a transfer has taken place. A value of 0 is provided to the filter input if no transfer has taken place during the frame. The input value is illustrated at line 175 while the output value is illustrated at line 180. When this output value exceeds the threshold value TDV, designated at line 185, a non-linear condition is declared. The foregoing filter configuration may be implemented in hardware. However, it is preferably implemented in the software of one or more digital signal processors used to implement the echo canceller 25.

One manner of incorporating the foregoing threshold detection process in the operations of the process of Figure 4 is illustrated in Figure 6. As illustrated, the averaging filter is notified at step 190 that a transfer has taken place. The resulting filtered value is retrieved at step 195 and compared to the threshold value at step 200. If the threshold value TDV is exceeded, the E_{max} value is checked at step 205 to ensure that it is above a threshold value (e.g., 12dB) before a non-linear echo path condition is declared. If E_{max} is below this value, no such declaration is made. This check to ensure that the E_{max} value is greater than a predetermined value assists in ensuring that a non-linear echo path condition is not declared during periods of convergence and reconvergence of the adaptive filter. If E_{max} is above this value, a non-linear echo path response is declared at step 210.

A further manner for detecting non-linearities using one or more characteristics of the filter $\hat{h}$ is illustrated in connection with Figures 7 and 8. In accordance with this further manner, the time dispersion of the tap coefficient values is used to determine whether the echo path is linear or non-linear.

A graph of the tap coefficients of the $\hat{h}$ filter with respect to the time delay associated with each tap is illustrated in Figure 7 for a linear echo path response. As shown, the $\hat{h}$ response is comprised of a small number of large magnitude taps (shown here as those taps numbered 0 to 100) and many taps that are near 0 (shown here as taps numbered 101 to 511). Other linear echo paths generally yield similar results wherein a small percentage of the total number of taps are used to model the echo response and a large number of taps have coefficients that are approximately equal to 0. As such, the $\hat{h}$ coefficients have a generally small time dispersion value (i.e., the majority of the energy of the $\hat{h}$ coefficients is confined to a small window of time).

Figure 8 illustrates the coefficient values of $\hat{h}$ when attempting to model the echo response of a non-linear echo path. In this case, the ratio of the number of taps having large tap coefficients to the number of taps having low value tap coefficients is not as large as it was in the linear case of Figure 7. The energy of the taps of $\hat{h}$ is more widely dispersed among the taps. The relatively lower value tap coefficients are much larger in the non-linear case since they are playing an active role in finding short-term solutions for canceling the non-linear components of the echo. As such, the $\hat{h}$ coefficients have a generally large time dispersion

value when attempting to model a non-linear echo path (i.e., the tap energy is dispersed over a large number of taps).

In view of the foregoing characteristics recognized by the present inventors, the echo canceller 25 calculates the time dispersion of the tap coefficients of the $\hat{h}$ filter to determine whether the echo path is linear or non-linear. The time dispersion value is compared to a threshold. The threshold depends on the method of measuring the time dispersion and can be chosen experimentally. When the time dispersion value moves above the threshold, a non-linear echo path is declared. Otherwise, the echo path is assumed to be linear. Hysteresis can be used in making this determination.

The time dispersion value can be calculated in many ways. In accordance with one manner, the echo canceller may find the inverse of the fraction of the total tap energy which can be attributed to the largest M taps, where M is a small number compared to the total number of taps in the $\hat{h}$ filter. In accordance with a further manner of calculating time dispersion, the echo canceller may measure the ratio of the L lowest tap coefficients to the number M of the largest tap coefficients. However it is measured, the time dispersion will be larger for non-linear calls than for linear calls.

The time dispersion process of the echo canceller 25 can be augmented with a few other processes to improve its accuracy. For example, the echo canceller 25 may require that the time dispersion value remain above the threshold for a predetermined period of time before a non-linear echo path is declared. Such a requirement assists in preventing a period of convergence and reconvergence, which

also might be characterized with large time dispersion values, from falsely tripping the non-linear detector process. Still further, the echo canceller may require that a measurement of the background noise be made prior to declaring a non-linear echo path. High levels of background noise may increase the time dispersion thereby causing a false indication of non-linearity. The echo canceller 25 may be programmed to require both a low background noise condition and a high time dispersion value above the threshold before declaring the presence of a non-linear echo path.

One manner of implementing the time dispersion process in the process shown in Figure 4 is illustrated in Figure 9.

When a non-linear echo path is detected, the echo canceller 25 preferably enters a time varying mode. In this mode, the output decision logic is biased to use the $\hat{h}$ filter to compensate for the echo response. This may be accomplished, for example, by transferring switch 45 so that the $\hat{h}$ output is consistently used for the echo compensation. Alternatively, this bias may be accomplished by varying the $\hat{h}$ to $\bar{h}$ transfer criterion discussed above so as to make the transfers more likely to occur. Additionally, the adaptation gain used to adapt the tap coefficients of $\hat{h}$ is preferably lowered so as to reduce the rate at which the tap coefficients change and prevent overshoot of short term solutions. Further, any split adaptation of the tap coefficients is inhibited in time varying mode. Such split adaptation is shown and described and U.S.S.N. ____ (Attorney Docket No. ___) titled "DUAL-H ECHO CANCELLER HAVING SPLIT ADAPTATION OF ADAPTIVE COEFFICIENTS", filed on even date herewith.

A further manner of detecting non-linearities based on the adapted coefficients is to monitor the variance of each of the taps over time. A large tap variance indicates non-linearities while a low tap variance indicates a linear echo path response.

Irrespective of the manner used to declare the presence of a non-linear echo path response in the channel, some manner of canceling non-linearities in the echo response is needed. To this end, the echo canceller 25 is provided with a non-linear processor, shown generally at 300 of FIG. 2.

The NLP 300 works by reducing small input signals to zero while leaving larger input signals relatively unchanged. To accomplish this, the NLP 300 includes a center clipping filter. The output of the NLP 300 as a function of the input thereto is illustrated in Figure 10. An input signal having an amplitude less than or greater than a calculated NLP threshold T_{NLP} will be reduced to zero at the output of the NLP 300. Larger amplitude signals are modified only near their zero crossings. Alternatively, the NLP 300 may be designed in accordance with the response of the non-linear processor of USPN 5,274,705, titles "Nonlinear Processor for an Echo Canceller", which is hereby incorporated by reference.

Figure 11a illustrates a small signal input to the NLP 300 at 305 and the resulting output signal at 310. Similarly, Figure 11b illustrates a large signal input to the NLP 300 at 315 and the resulting output signal at 320. Note that the resulting output signal is clipped at amplitudes between $-T_{NLP}$ and $+T_{NLP}$.

To be generally effective, the NLP 300 should set T_{NLP} larger than the amount of non-cancellable echo. This provides the end user with the experience of

"echo-free" performance. However, the value of T_{NLP} should not be set to a value any greater than necessary so as to minimize distortion of larger signals.

The value of T_{NLP} may be obtained by estimating the power of $\sigma^2_{e'}$ from measurable echo canceller quantities (note that $\sigma^2_{e'}$ = powe(e')). Such quantities are set forth in Figure 12 in connection with a standard linear echo canceller. To this end, the echo canceller 25 measures the power of the X, σ^2_x signal and subtracts the difference in power across lines 340 from the measured quantities (e.g., the echo return loss or "*ERL*"). The echo canceller 25 then subtracts the power decrease due to the echo canceller 25, which is *ERLE*, from the calculated difference. The result is an estimate of $\sigma^2_{e'}$, which is expressed as $\tilde{\sigma}^2_{e'}$ (note that $\tilde{\sigma}^2_{e'}$ = pow(e)) far. This corresponds to the following equations:

$$\tilde{\sigma}^2_{e'} = \sigma^2_{e'} \cdot \underbrace{\frac{\sigma^2_y}{\sigma^2_x}}_{\text{ERL}^{-1}} \cdot \underbrace{\frac{\sigma^2_{e'}}{\sigma^2_y}}_{\text{ERL}^{-1}} \qquad \text{(assuming } v = \text{O)}$$

or in terms of dB

$$\tilde{\sigma}^2_{e'} \text{ (in dB)} = \sigma^2_x \text{ (in dB - ERL (in dB) - ERLE (in dB)}$$

Once the estimate $\tilde{\sigma}^2_{e'}$ has been made, the echo canceller 25 may then set the value of T_{NLP} equal to or slightly higher than $\tilde{\sigma}^2_{e'}$. Assuming that T_{NLP} is set to slightly overbound $\tilde{\sigma}^2_{e'}$, the far-end user should hear no residual echo with only minimal distortion to the near end speech.

It is possible, but not necessary, to use E_{MAX}^{-1} as an estimate of $ERLE^{-1}$ in determining $\tilde{\sigma}^2_{e'}$. In calculating *ERL*, it is possible, but not necessary to instead calculate *ERL* using the following equation:

$$ERL = pow(x_{\max}) / pow(y)$$

where $x_{\max}$ is the maximum value of x over the last L seconds, where L is a constant.

Measurements of *ERL* and *ERLE* and, thus, $\tilde{\sigma}^2_{e'}$, will likely be erroneous if a double-talk condition exists when these measurements are taken. During periods of double-talk, the measured *ERLE* $\approx$ 0 dB and the measured *ERL* is very small, perhaps negative. The resulting $\tilde{\sigma}^2_{e'}$ would be much larger than desired, thus setting T_{NLP} too high and causing severe distortion of the near-end speaker's voice as perceived at the far-end.

Steps are taken to ensure that the T_{NLP} value is not modified during double-talk. Preferably, double-talk detection is implemented in accordance with the teachings of U.S.S.N. ___, titled ____ (Attorney Docket No. ___), filed on even

date herewith. Even more preferably, the double-talk detection is implemented in the manner set forth in Figure 13.

The double-talk detector 400 compares long-term and short-term averages of the error power. The long-term average is computed in such a way that it can be considered a steady-state measure of the residual error. When the short-term measure of the error power noticeably exceeds the long term measure (e.g., by 3 or 6 dB), it is assumed that e' signal no longer contains only residual echo, but in addition, near-end speech. This event thus triggers a double talk event.

As illustrated in Figure 13, the double-talk detector 400 employs at least two measurements of the average echo power that are compared with one another wherein that each measurement corresponds to a different averaging time. In the illustrated embodiment, three measurements of the average echo power are employed, each using a different window of time in which it performs its averaging function. From an implementation standpoint, it is convenient to translate these various averaging window lengths into the time constants of averaging filters: longer averaging windows translate into longer time constants. To this end, the double-talk detector 400 employs a short time constant filter S, a medium time constant filter M, and a long time constant filter L.

The S filter simply uses a short time constant, e.g., 2 to 7 msec, to average e' which is supplied at input line 405. The M filter uses a medium time constant, e.g., 12 to 36 msec, to average e'. The computation of L is slightly more complex. In the illustrated embodiment, L averages M, not e' directly. When $M \geq L$, L averages M with a very long time constant (e.g., 500-2000 msec).

When $M < L$, then L averages M with a much smaller time constant. The net effect is that L and M form a dual time constant filter in which the output of L increases very slowly but decreases very quickly. The resulting filter is a "slow attack, fast decay" average of M.

Figure 14 illustrates a signal input e' and the corresponding output signals from filters S, M, and L. The output signals of filters S and L are provided to the input of comparator 410. The output of filter L serves to maintain the lower bound of e'. When the comparator 410 detects that the output signal of filter S is noticeably larger than the output signal of filter L, a double-talk condition is declared.

The foregoing double-talk detector is preferably implemented in software code of one or more digital signal processors used to implement the echo canceller 25. When a double-talk condition is declared, the T_{NLP} value is preferably locked and no further modifications thereof are made until the double-talk condition is no longer present.

As noted above, the echo return loss *ERL* measurement is used to ultimately arrive at a value for T_{NLP}. However, this measurement may be significantly corrupted during a double-talk condition thereby impacting the value of T_{NLP}. To further reduce the likelihood of such corruption, the non-linear processor 300 of the echo canceller 25 preferably locks the *ERL* value without further calculation thereof once the processor 300 has determined that the *ERL* is at a generally static value.

One method that may be used to measure *ERL* involves mapping the measured value to a magnitude in a look-up table. The magnitude of the x signal is

divided by the magnitude of the y signal. This ratio is then quantized to the closest member of a pre-defined set by way of the look-up table. For example, let the pre-defined set by {0, 3, 6, 9, . . ., 30} dB, and let $\tilde{\sigma}_x^2 / \tilde{\sigma}_y^2 = 7.1$ dB. Then the measured *ERL* would map to the set member 6 dB. On the next instance, let $\tilde{\sigma}_x^2 / \tilde{\sigma}_y^2 = 5.61$ dB. This too would map to 6 dB.

Now consider the situation where several consecutive amplitude ratios map to the same value. If this continues for a period of time, the echo canceller 25 may accept that the estimate of the *ERL* is valid for the remainder of the call. Locking in this estimate of *ERL* means that future instances of double-talk cannot corrupt the estimate of *ERL*, and thus T_{NLP} is less likely to be corrupted during double-talk.

To this end, a counter is incremented if the current table look-up produces the same result as the previous look-up. This counter is cleared if any of the following conditions occur: 1) a different look-up result occurs; 2) the far-end goes silent for a substantial period of time; 3) double-talk is detected (e.g., by the averaging filter comparison above); or 4) other conditions are detected that will likely skew the *ERL* measurement.

If the counter reaches a predetermined large value (e.g., 500 to 2000 msec), an *ERL* Lock condition is declared and the current *ERL* estimate is used without further look-ups or calculations thereof. The ERL Lock is lifted if an endpath switch is detected by the echo canceller indicating that a call has terminated.

Figure 15 illustrates one embodiment of an echo canceller system, shown generally at 700, that maybe used to cancel echos in multi-channel communication

transmissions. As illustrated, the system 700 includes an input 705 that is connected to receive a multi-channel communications data, such as a T1 transmission. A central controller 710 deinterleaves the various channels of the transmission and provides them to respective convolution processors 715 over a data bus 720. It is within the convolution processors 715 that a majority of the foregoing operations take place. Each convolution processor 715 is designed to process at least one channel of the transmission at line 730. After each convolution processor 715 has processed its respective channel(s), the resulting data is placed on the data bus 720. The central controller 710 multiplexes the data into the proper multichannel format (e.g., T1) for retransmission at line 735. User interface 740 is provided to set various user programmable parameters of the system.

Numerous modifications may be made to the foregoing system without departing from the basic teachings thereof. Although the present invention has been described in substantial detail with reference to one or more specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the scope and spirit of the invention as set forth in the appended claims.

CLAIMS

1. A non-linear processor for use in an echo canceller, the non-linear processor comprising:

a center clipping digital filter receiving an echo compensated signal and providing a center clipped output signal, the center clipping digital filter having non-linear thresholds at values of $+T_{NLP}$ and $-T_{NLP}$, the value of T_{NLP} being dynamically dependent, at least in part, on echo return loss measurements;

means for inhibiting the dynamic setting of the T_{NLP} value when a double-talk condition is present.

2. A non-linear processor as claimed in claim 1 wherein the non-linear processor further comprises means for locking the value of the echo return loss measurement after a predetermined number of consecutive echo return loss measurements have values falling within a predetermined range of one another.

3. A non-linear processor as claimed in claim 2 wherein the means for inhibiting comprises means for detecting a double-talk condition.

4. A non-linear processor as claimed in claim 3 wherein the means for detecting a double-talk condition comprises:

a first averaging filter having a first time constant, the first averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a second averaging filter having a second time constant greater than the first time constant, the second averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a third averaging filter having a filter input receiving the time averaged filter output of the second averaging filter and a filter output to a time averaged value of the signal output of the second averaging filter, the third averaging filter using a third time constant when an immediately prior filter output signal of the third averaging filter is greater than the filter output signal of the second averaging filter, the third averaging filter using a fourth time constant when an immediately prior filter output signal of the third averaging filter is less than the filter output signal of the second averaging filter, the fourth time constant being greater than the third time constant;

a comparator for comparing the filter output of the first averaging filter and the filter output of the third averaging filter, the

comparator declaring a double-talk condition when the filter output of the first averaging filter exceeds the filter output of the third averaging filter by a predetermined value.

5. A non-linear processor for use in an echo canceller, the non-linear processor comprising:

a center clipping digital filter receiving an echo compensated signal and providing a center clipped output signal, the center clipping digital filter having non-linear thresholds at values of $+T_{NLP}$ and $-T_{NLP}$, the value of T_{NLP} being dynamically dependent, at least in part, on echo return loss measurements;

means for locking the value of the echo return loss measurement after a predetermined number of consecutive echo return loss measurements have values falling within a predetermined range of one another.

6. A non-linear processor as claimed in claim 5 wherein the non-linear processor further comprises means for inhibiting the dynamic setting of the T_{NLP} value when a double-talk condition is present.

7. A non-linear processor as claimed in claim 5 wherein the means for inhibiting comprises means for detecting a double-talk condition.

8. A non-linear processor as claimed in claim 7 wherein the means for detecting a double-talk condition comprises:

a first averaging filter having a first time constant, the first averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a second averaging filter having a second time constant greater than the first time constant, the second averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a third averaging filter having a filter input receiving the time averaged filter output of the second averaging filter and a filter output to a time averaged value of the signal output of the second averaging filter, the third averaging filter using a third time constant when an immediately prior filter output signal of the third averaging filter is greater than the filter output signal of the second averaging filter, the third averaging filter using a fourth time constant when an immediately prior filter output signal of the third averaging filter is less than the filter output signal of the second averaging filter, the fourth time constant being greater than the third time constant;

a comparator for comparing the filter output of the first averaging filter and the filter output of the third averaging filter, the comparator declaring a double-talk condition when the filter output of the first averaging filter exceeds the filter output of the third averaging filter by a predetermined value.

9. An echo canceller comprising:

at least one input for receiving a far-end signal of a call;

at least one input for receiving a signal-plus-echo signal of the call, the signal-plus-echo single having a signal component corresponding to an echo response of a transmission medium carrying the call;

a first digital filter receiving the far-end signal and having non-adaptive tap coefficients to simulate the echo response;

a summer circuit for subtracting the filtered far-end output signal of the first digital filter from the signal-plus-echo signal to generate an echo compensated signal for transmission to a far-end;

a second digital filter receiving the far-end signal and having adaptive tap coefficients to simulate the echo response, the adaptive tap coefficients being updated during the call;

a coefficient transfer controller disposed to transfer the adaptive tap coefficients of the second digital filter to replace the tap

coefficients of the first digital filter when a set of one or more predetermined conditions exists;

a center clipping digital filter receiving an echo compensated signal from the output of the summer circuit and providing a center clipped output signal, the center clipping digital filter having non-linear thresholds at values of $+T_{NLP}$ and $-T_{NLP}$, the value of T_{NLP} being dynamically dependent, at least in part, on echo return loss measurements;

means for inhibiting the dynamic setting of the T_{NLP} value when a double-talk condition is present.

10. An echo canceller as claimed in claim 9 and further comprising means for locking the value of the echo return loss measurement after a predetermined number of consecutive echo return loss measurements have values falling within a predetermined range of one another.

11. An echo canceller as claimed in claim 9 wherein the means for inhibiting comprises means for detecting a double-talk condition.

12. An echo canceller as claimed in claim 11 wherein the means for detecting a double-talk condition comprises:

a first averaging filter having a first time constant, the first averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a second averaging filter having a second time constant greater than the first time constant, the second averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a third averaging filter having a filter input receiving the time averaged filter output of the second averaging filter and a filter output to a time averaged value of the signal output of the second averaging filter, the third averaging filter using a third time constant when an immediately prior filter output signal of the third averaging filter is greater than the filter output signal of the second averaging filter, the third averaging filter using a fourth time constant when an immediately prior filter output signal of the third averaging filter is less than the filter output signal of the second averaging filter, the fourth time constant being greater than the third time constant;

a comparator for comparing the filter output of the first averaging filter and the filter output of the third averaging filter, the

comparator declaring a double-talk condition when the filter output of the first averaging filter exceeds the filter output of the third averaging filter by a predetermined value.

13. An echo canceller comprising:

at least one input for receiving a far-end signal of a call;

at least one input for receiving a signal-plus-echo signal of the call, the signal-plus-echo single having a signal component corresponding to an echo response of a transmission medium carrying the call;

a first digital filter receiving the far-end signal and having non-adaptive tap coefficients to simulate the echo response;

a summer circuit for subtracting the filtered far-end output signal of the first digital filter from the signal-plus-echo signal to generate an echo compensated signal for transmission to a far-end;

a second digital filter receiving the far-end signal and having adaptive tap coefficients to simulate the echo response, the adaptive tap coefficients being updated during the call;

a coefficient transfer controller disposed to transfer the adaptive tap coefficients of the second digital filter to replace the tap coefficients of the first digital filter when a set of one or more predetermined conditions exists;

a center clipping digital filter receiving an echo compensated signal from the output of the summer circuit and providing a center clipped output signal, the center clipping digital filter having non-linear thresholds at values of $+T_{NLP}$ and $-T_{NLP}$, the value of T_{NLP} being dynamically dependent, at least in part, on echo return loss measurements; and

means for locking the value of the echo return loss measurement after a predetermined number of consecutive echo return loss measurements have values falling within a predetermined range of one another.

14. An echo canceller as claimed in claim 13 and further comprising means for inhibiting the dynamic setting of the T_{NLP} value when a double-talk condition is present.

15. A non-linear processor as claimed in claim 14 wherein the means for inhibiting comprises means for detecting a double-talk condition.

16. A non-linear processor as claimed in claim 15 wherein the means for detecting a double-talk condition comprises:

a first averaging filter having a first time constant, the first averaging filter having a filter input receiving the echo compensated

signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a second averaging filter having a second time constant greater than the first time constant, the second averaging filter having a filter input receiving the echo compensated signal and a filter output corresponding to a time averaged value of the echo compensated signal;

a third averaging filter having a filter input receiving the time averaged filter output of the second averaging filter and a filter output to a time averaged value of the signal output of the second averaging filter, the third averaging filter using a third time constant when an immediately prior filter output signal of the third averaging filter is greater than the filter output signal of the second averaging filter, the third averaging filter using a fourth time constant when an immediately prior filter output signal of the third averaging filter is less than the filter output signal of the second averaging filter, the third time constant being greater than the fourth time constant;

a comparator for comparing the filter output of the first averaging filter and the filter output of the third averaging filter, the comparator declaring a double-talk condition when the filter

output of the third averaging filter exceeds the filter output of the first averaging filter by a predetermined value.

FIG. 1

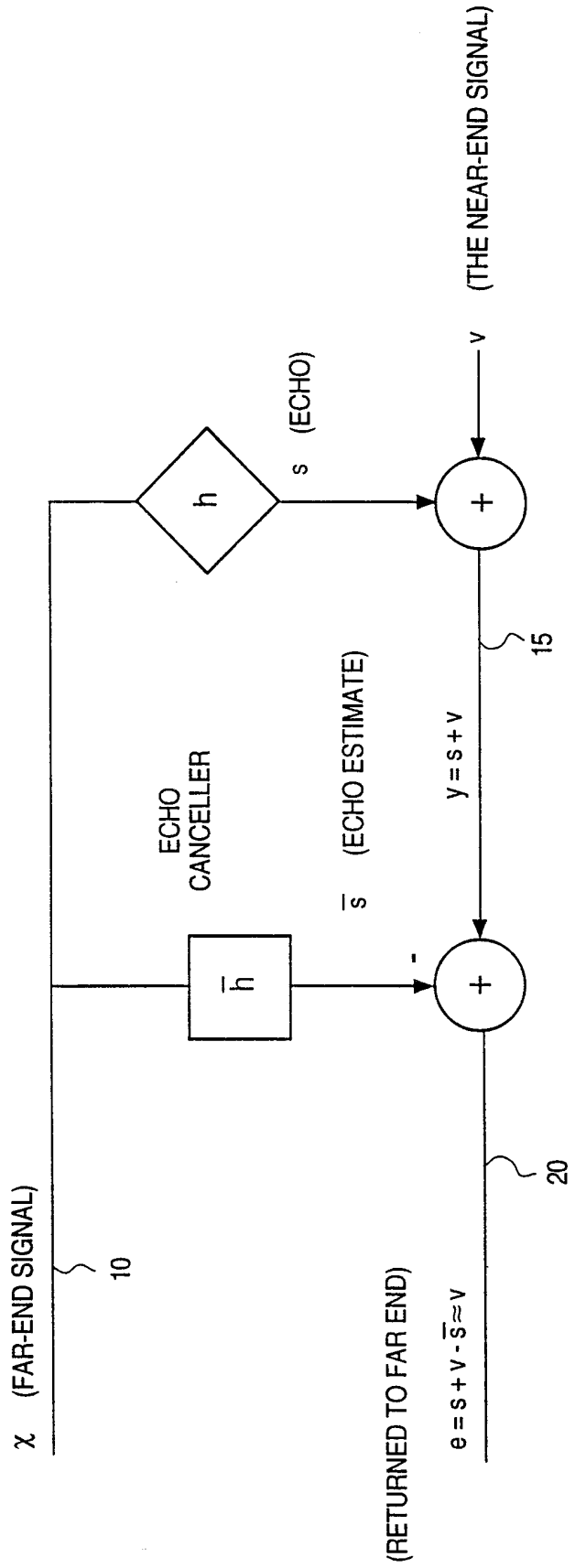

x (FAR-END SIGNAL)
10
h
s (ECHO)
v (THE NEAR-END SIGNAL)
+
15
y = s + v
ECHO CANCELLER
$\bar{s}$ (ECHO ESTIMATE)
$\bar{h}$
-
+
20
e = s + v - $\bar{s}$ ≈ v
(RETURNED TO FAR END)

FIG. 2

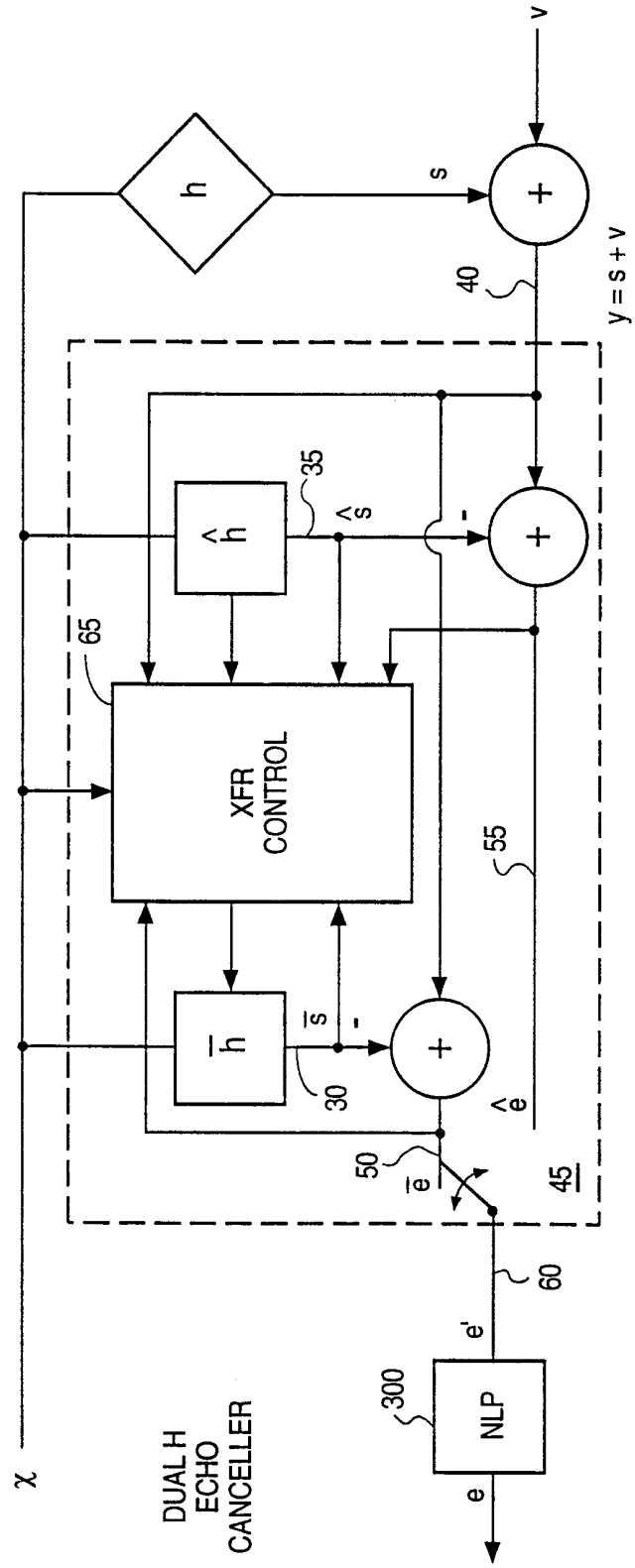

v
h
s
+
40
y = s + v
25
35
ŝ
-
65
XFR CONTROL
55
s̄
30
ê
50
ē
45
60
e'
300
NLP
e
x
DUAL H ECHO CANCELLER

FIG. 3

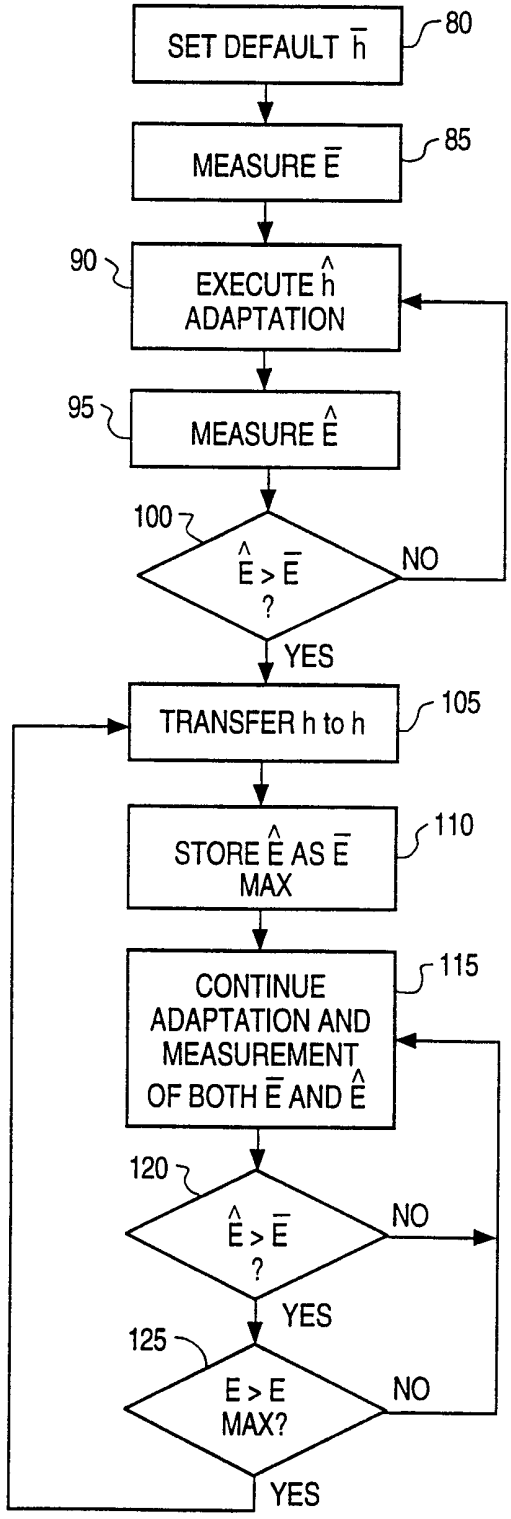

SET DEFAULT $\bar{h}$
80
MEASURE $\bar{E}$
85
90
EXECUTE $\hat{h}$ ADAPTATION
95
MEASURE $\hat{E}$
100
$\hat{E} > \bar{E}$?
NO
YES
TRANSFER h to h
105
STORE $\hat{E}$ AS $\bar{E}$ MAX
110
CONTINUE ADAPTATION AND MEASUREMENT OF BOTH $\bar{E}$ AND $\hat{E}$
115
120
$\hat{E} > \bar{E}$?
NO
YES
125
E > E MAX?
NO
YES

FIG. 4

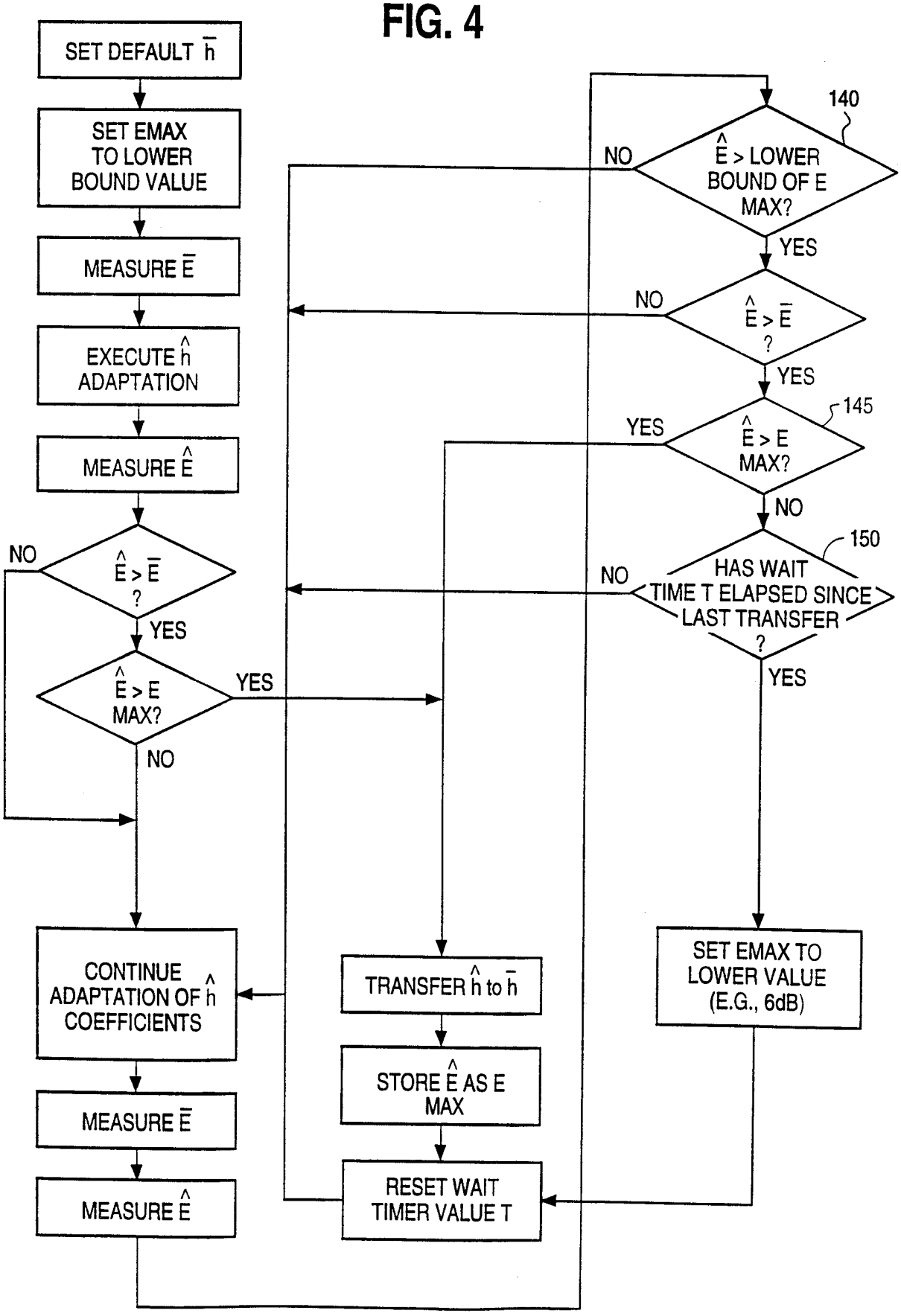

SET DEFAULT $\bar{h}$
SET EMAX TO LOWER BOUND VALUE
MEASURE $\bar{E}$
EXECUTE $\hat{h}$ ADAPTATION
MEASURE $\hat{E}$
$\hat{E} > \bar{E}$?
NO
YES
$\hat{E} > E$ MAX?
YES
NO
CONTINUE ADAPTATION OF $\hat{h}$ COEFFICIENTS
MEASURE $\bar{E}$
MEASURE $\hat{E}$
TRANSFER $\hat{h}$ to $\bar{h}$
STORE $\hat{E}$ AS E MAX
RESET WAIT TIMER VALUE T
140
$\hat{E}$ > LOWER BOUND OF E MAX?
NO
YES
$\hat{E} > \bar{E}$?
NO
YES
145
$\hat{E} > E$ MAX?
YES
NO
150
HAS WAIT TIME T ELAPSED SINCE LAST TRANSFER ?
NO
YES
SET EMAX TO LOWER VALUE (E.G., 6dB)

FIG. 5a

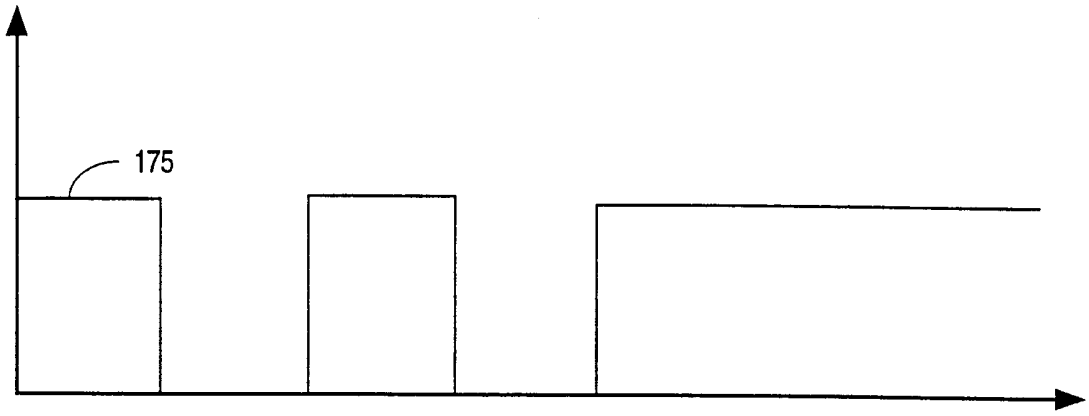


FIG. 5b

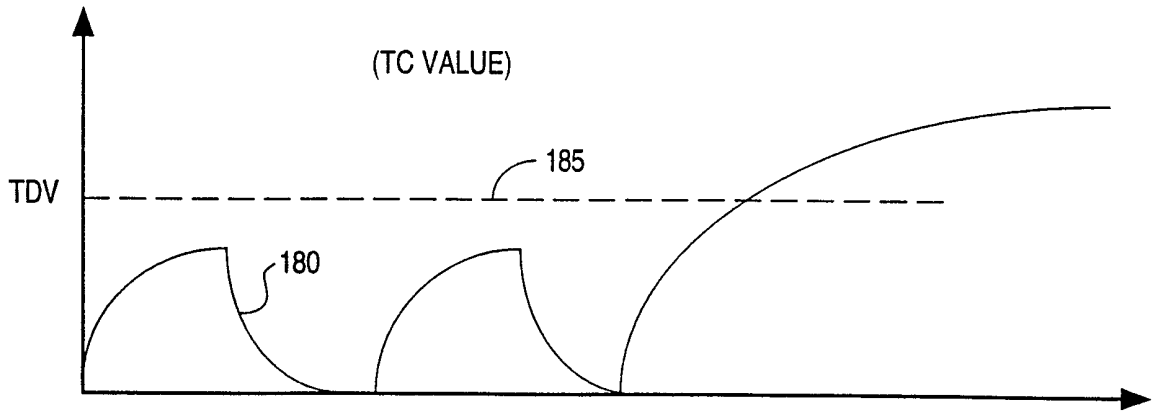


FIG. 6

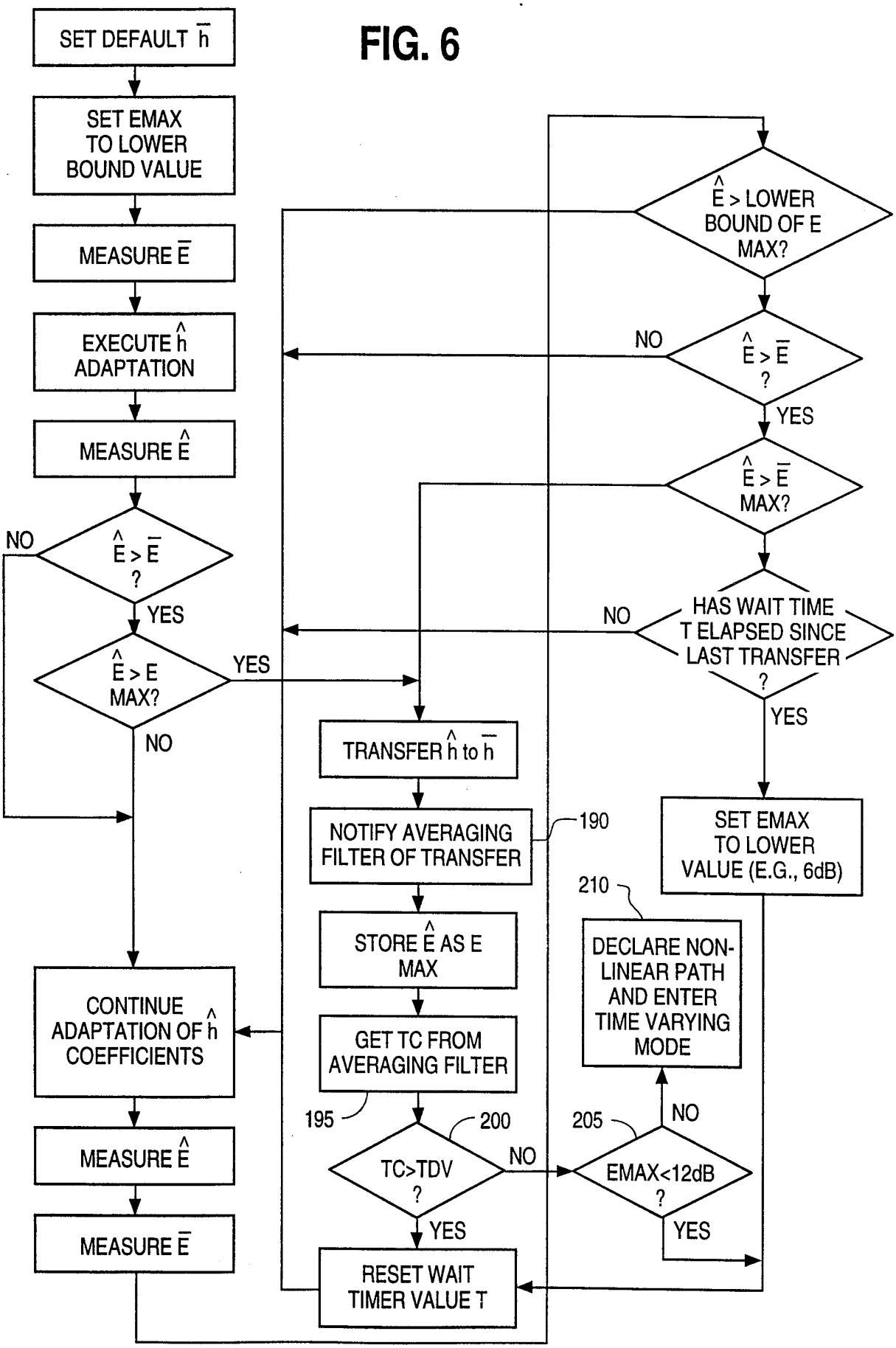

SET DEFAULT $\bar{h}$
SET EMAX TO LOWER BOUND VALUE
MEASURE $\bar{E}$
EXECUTE $\hat{h}$ ADAPTATION
MEASURE $\hat{E}$
$\hat{E} > \bar{E}$?
NO
YES
$\hat{E} > E$ MAX?
YES
NO
CONTINUE ADAPTATION OF $\hat{h}$ COEFFICIENTS
MEASURE $\hat{E}$
MEASURE $\bar{E}$
TRANSFER $\hat{h}$ to $\bar{h}$
NOTIFY AVERAGING FILTER OF TRANSFER
190
STORE $\hat{E}$ AS E MAX
GET TC FROM AVERAGING FILTER
195
200
TC>TDV ?
NO
YES
RESET WAIT TIMER VALUE T
$\hat{E} >$ LOWER BOUND OF E MAX?
NO
$\hat{E} > \bar{E}$?
YES
$\hat{E} > \bar{E}$ MAX?
NO
HAS WAIT TIME T ELAPSED SINCE LAST TRANSFER ?
YES
SET EMAX TO LOWER VALUE (E.G., 6dB)
210
DECLARE NON-LINEAR PATH AND ENTER TIME VARYING MODE
205
NO
EMAX<12dB ?
YES

FIG. 7

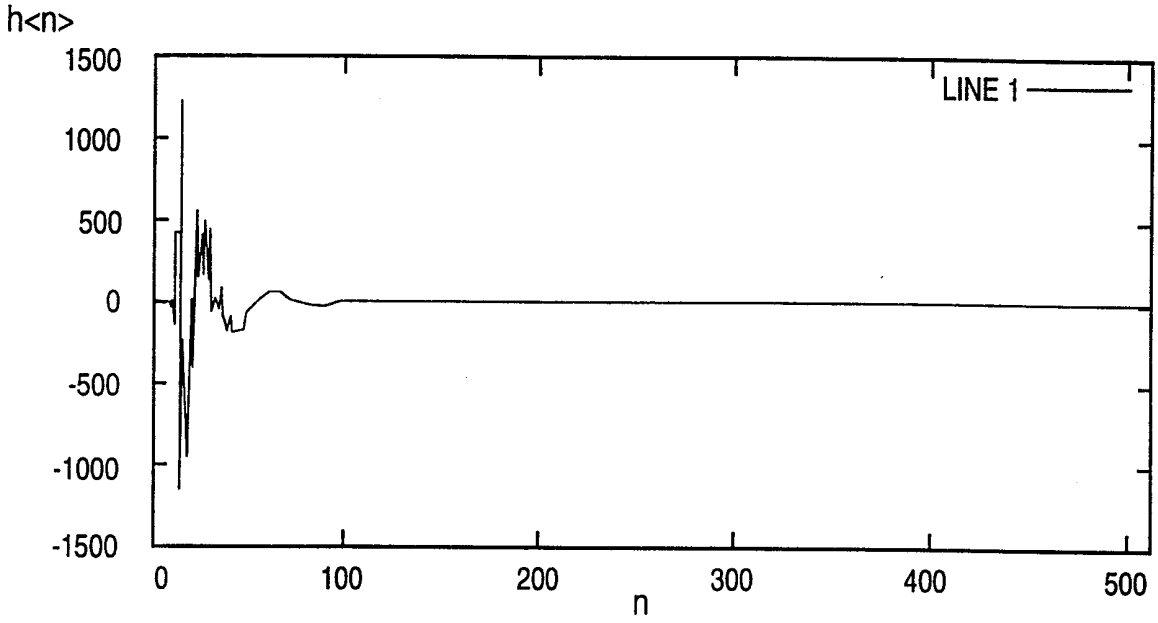

h<n>
1500
1000
500
0
-500
-1000
-1500
0
100
200
300
400
500
n
LINE 1

FIG. 8

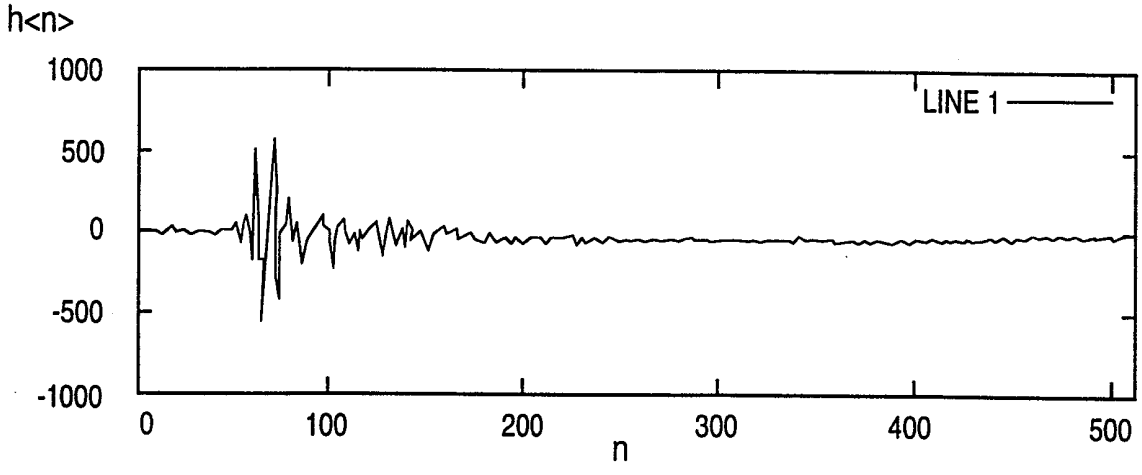

h<n>
1000
500
0
-500
-1000
0
100
200
300
400
500
n
LINE 1

FIG. 9

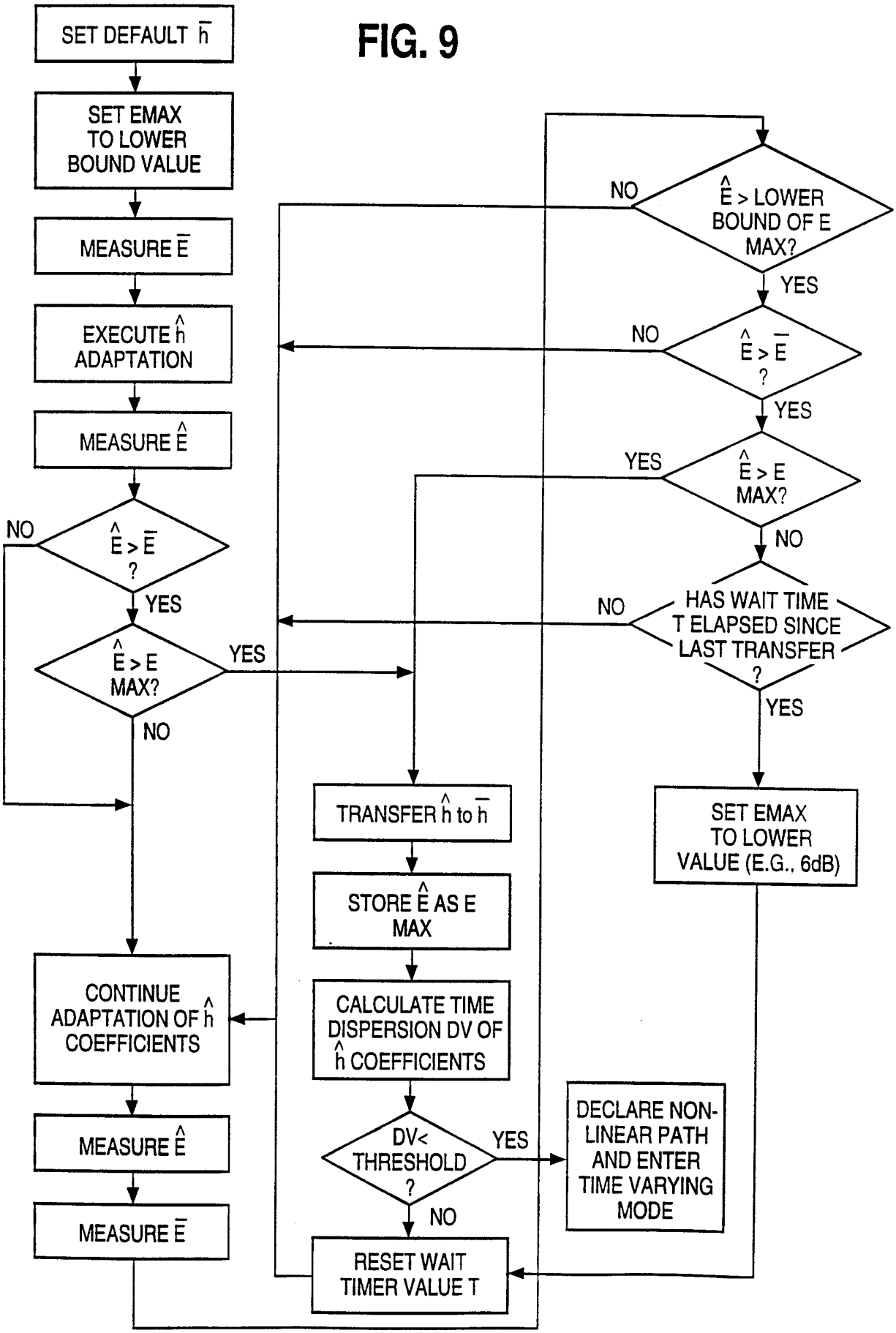

SET DEFAULT $\bar{h}$
SET EMAX TO LOWER BOUND VALUE
MEASURE $\bar{E}$
EXECUTE $\hat{h}$ ADAPTATION
MEASURE $\hat{E}$
$\hat{E} > \bar{E}$?
NO
YES
$\hat{E} >$ E MAX?
YES
NO
CONTINUE ADAPTATION OF $\hat{h}$ COEFFICIENTS
MEASURE $\hat{E}$
MEASURE $\bar{E}$
TRANSFER $\hat{h}$ to $\bar{h}$
STORE $\hat{E}$ AS E MAX
CALCULATE TIME DISPERSION DV OF $\hat{h}$ COEFFICIENTS
DV< THRESHOLD ?
YES
NO
DECLARE NON-LINEAR PATH AND ENTER TIME VARYING MODE
RESET WAIT TIMER VALUE T
$\hat{E} >$ LOWER BOUND OF E MAX?
NO
YES
$\hat{E} > \bar{E}$?
NO
YES
$\hat{E} >$ E MAX?
YES
NO
HAS WAIT TIME T ELAPSED SINCE LAST TRANSFER ?
NO
YES
SET EMAX TO LOWER VALUE (E.G., 6dB)

Fig. 10

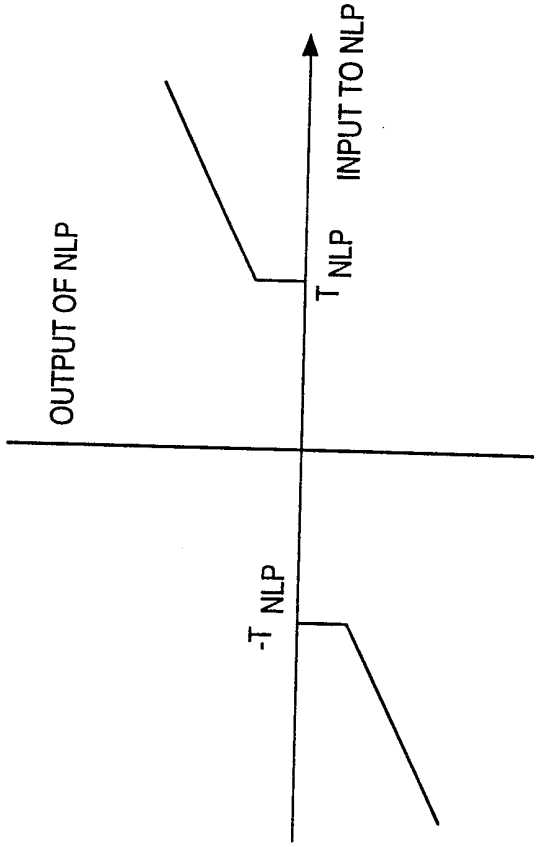


Fig. 11a-2

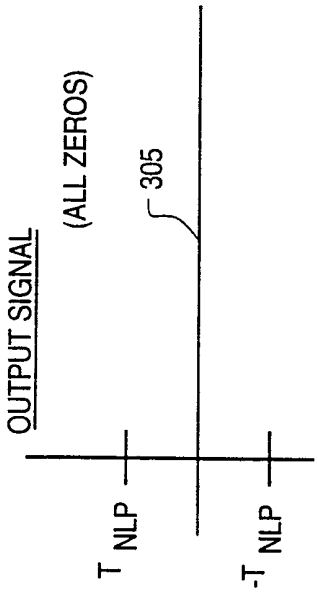


Fig. 11b-2

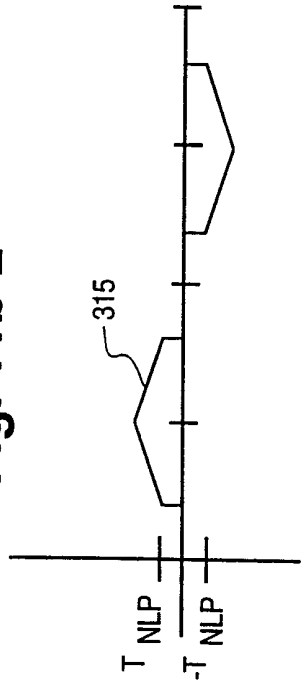


Fig. 11a-1

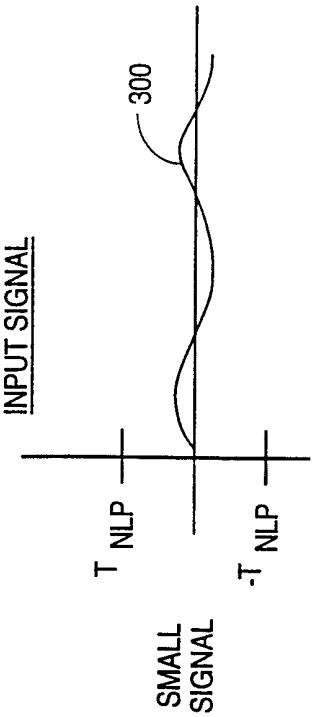


Fig. 11b-1

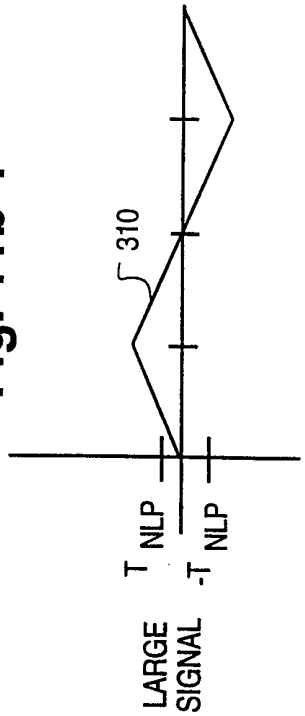


Fig. 12

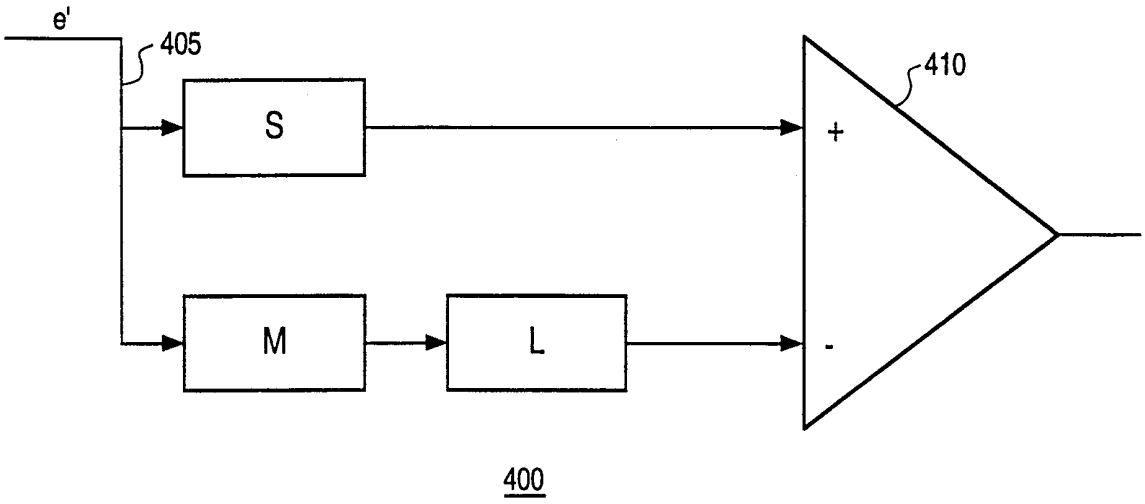


Fig. 13

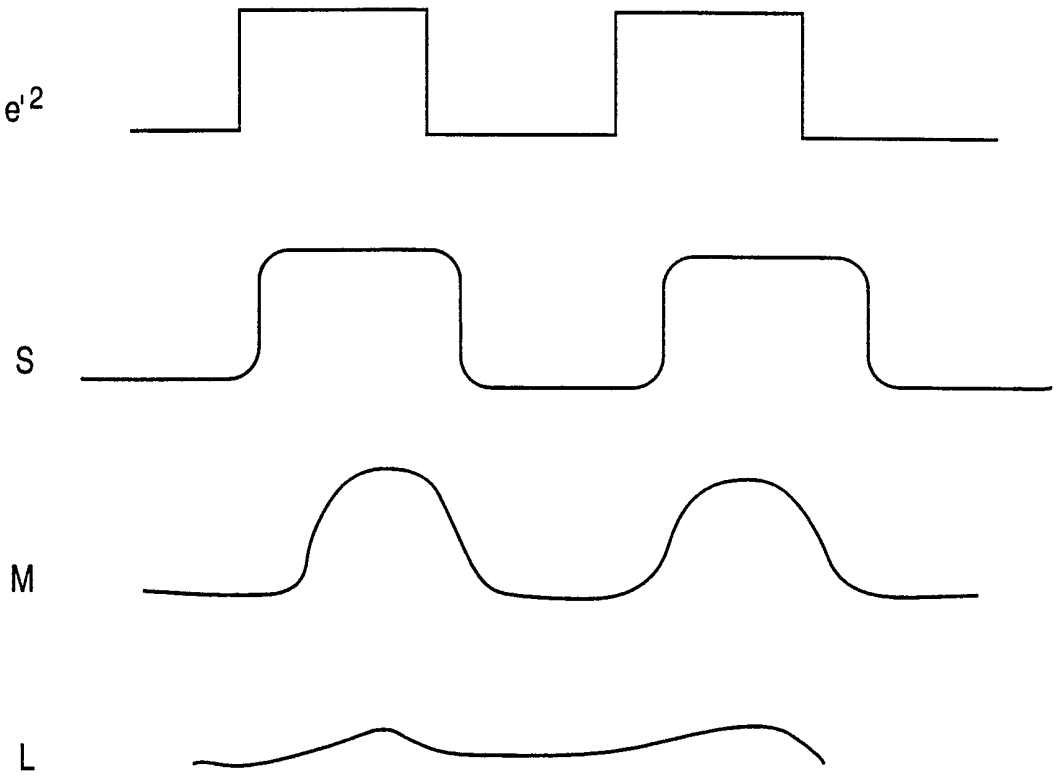


FIG. 14

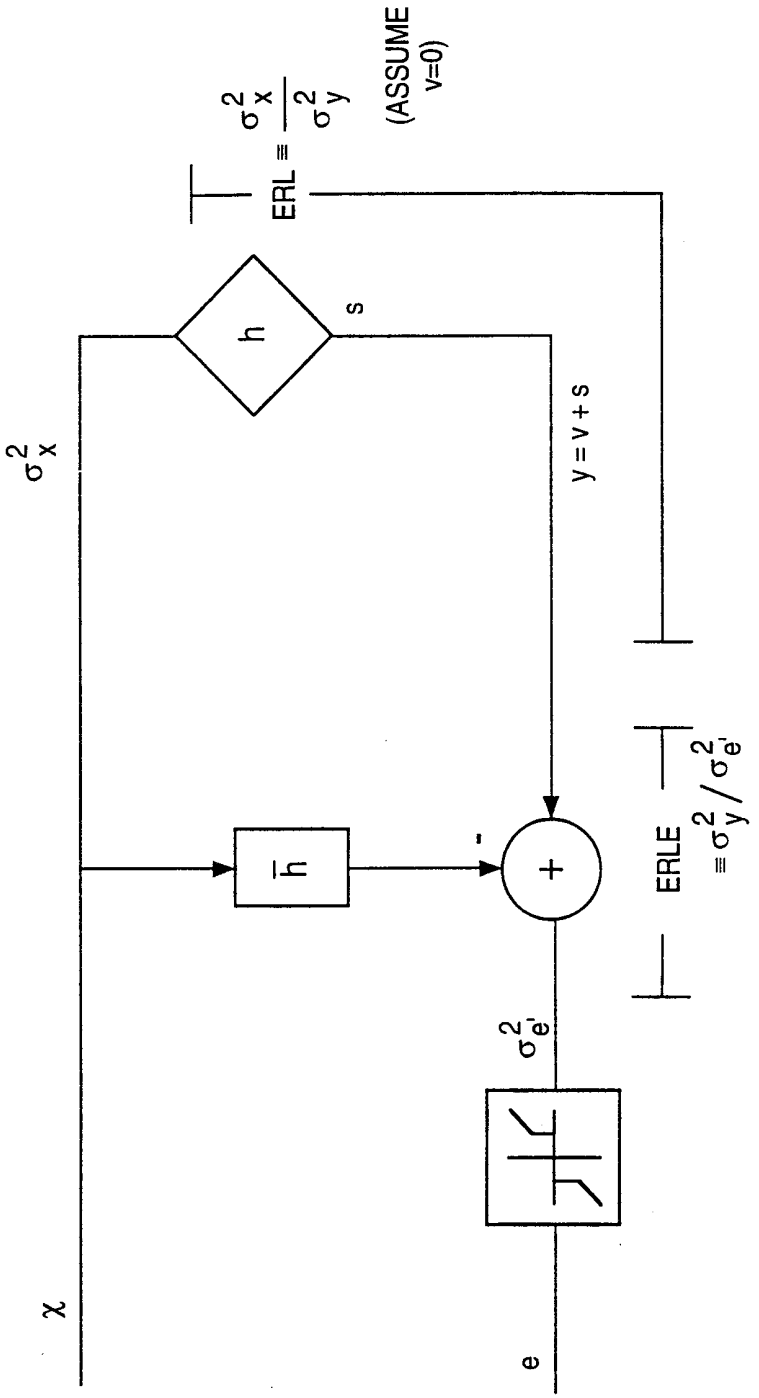
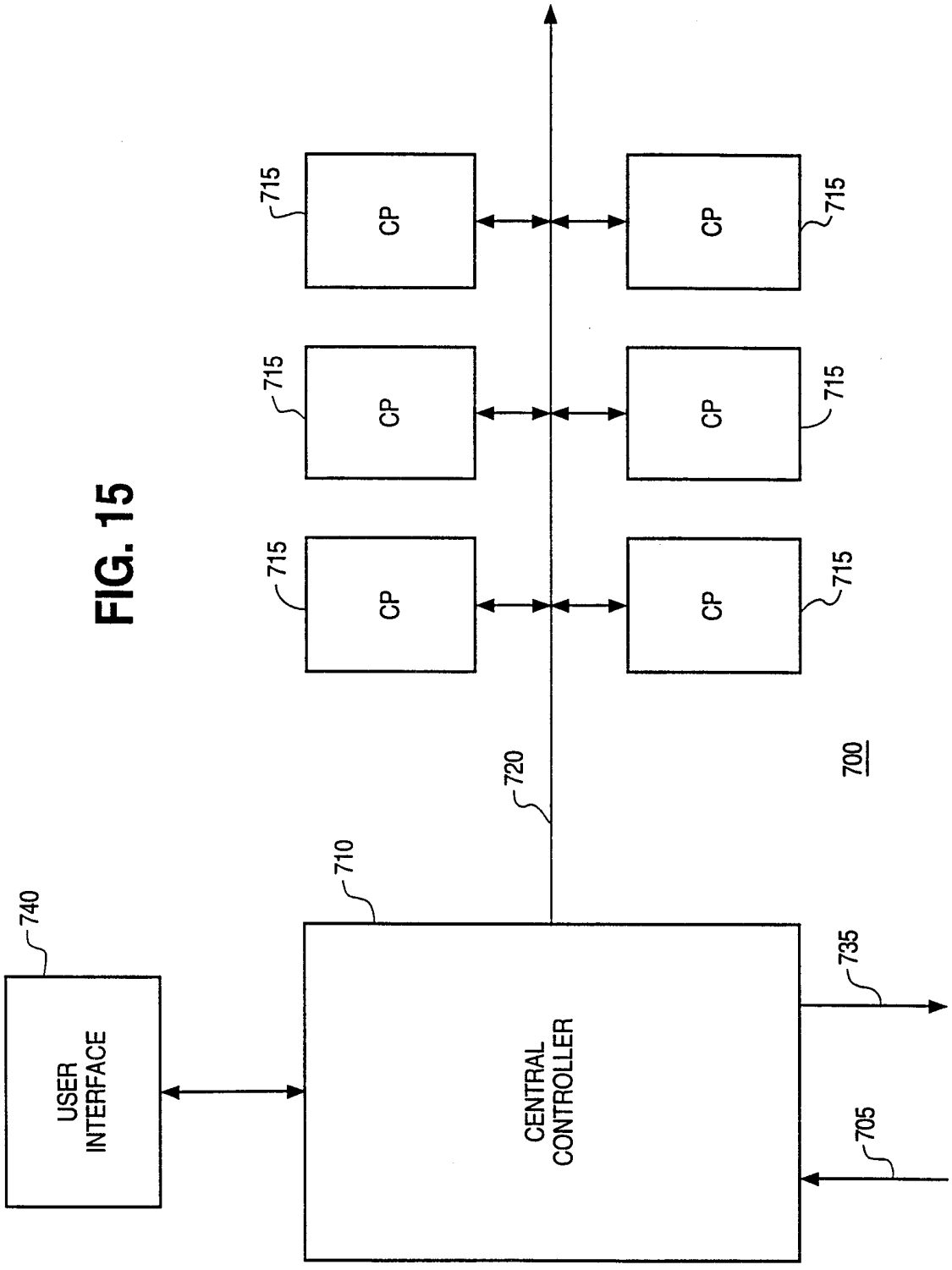


FIG. 15


715
CP
715
CP
715
CP
715
CP
715
CP
715
CP
720
700
710
740
USER INTERFACE
CENTRAL CONTROLLER
735
705

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US98/24346

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :HO4M 9/08; H04B 3/20
US CL :379/410, 406, 407, 408, 409, 410, 411; 364/724.19
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 379/410, 406, 407, 408, 409, 410, 411; 364/724.19

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

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

APS

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US, 5,274,705 A (YOUNCE et al.) 28 December 1993, col. 7, lines 31-68).	1
--		-----
Y		9 ,11
Y	US 3,787,645 A (OCHIAI et al.) 22 January 1974, col. 2, lines 40-65	9,11
Y	US 5,664,011 A (CROCHIERE et al.) 02 September 1997	9 , 11
A	US Re. 28,919 A (BERKLEY et al.) 27 July 1976	1-16
A	US 5,475,731 A (RASMUSSON) 12 DECEMBER 1995	1-16
A	US 5,315,585 A (IIZUKA et al.) 24 May 1994	1-16

[X] Further documents are listed in the continuation of Box C. [ ] See patent family annex.

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

Date of the actual completion of the international search 11 JANUARY 1999	Date of mailing of the international search report 15 MAR 1999
Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231 Facsimile No. (703) 305-3230	Authorized officer James R. Matthews JACQUES M. SAINT-SURIN Telephone No. (703) 305-47600

Form PCT/ISA/210 (second sheet)(July 1992)*

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US98/24346

C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 5,283,784 A (GENTER) 01 February 1994	1-16
A	US 5,329,586 A (AGAZZI) 12 July 1994	1-16
A	US 5,327,495 A (SHENOI et al.) 05 July 1994	1-16
A	US 3,699,271 A (BERKLEY et al.) 17 October 1972	1-16
A	US 4,897,832 A (SUZUKI et al.) 30 January 1990	1-16
A	US 5,315,585 A (IIZUKA et al.) 24 May 1994	1-16