A noise reduction circuit for a hearing aid having an adaptive filter for producing a signal which estimates the noise components present in an input signal. The circuit includes a second filter for receiving the noise-estimating signal and modifying it as a function of a user's preference or as a function of an expected noise environment. The circuit also includes a gain control for adjusting the magnitude of the modified noise-estimating signal, thereby allowing for the adjustment of the magnitude of the circuit response. The circuit also includes a signal combiner for combining the input signal with the adjusted noise-estimating signal to produce a noise reduced output signal.
ADAPTIVE NOISE REDUCTION CIRCUIT FOR A SOUND REPRODUCTION SYSTEM

This invention was made with U.S. Government support under Veterans Administration Contract V674-P-857 and V674-P-1736 and National Aeronautics and Space Administration (NASA) Research Grant No. NAGl0-0040. The U.S. Government has certain rights in this invention.

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BACKGROUND OF THE INVENTION

The present invention relates to a noise reduction circuit for a sound reproduction system and, more particularly, to an adaptive noise reduction circuit for a hearing aid.

A common complaint of hearing aid users is their inability to understand speech in a noisy environment. In the past, hearing aid users were limited to listening-in-noise strategies such as adjusting the overall gain via a volume control, adjusting the frequency response, or simply removing the hearing aid. More recent hearing aids have used noise reduction techniques based on, for example, the modification of the low frequency gain in response to noise. Typically, however, these strategies and techniques have not achieved as complete a removal of noise components from the audible range of sounds as desired.

In addition to reducing noise effectively, a practical ear-level hearing aid design must accommodate the power, size and microphone placement limitations dictated by current commercial hearing aid designs. While powerful digital signal processing techniques are available, they require considerable space and power such that most are not suitable for use in a hearing aid. According, there is a need for a noise reduction circuit that requires modest computational resources, that uses only a single microphone input, that has a large range of responses for different noise inputs, and that allows for the customization of the noise reduction according to a particular user’s preferences.

SUMMARY OF THE INVENTION

Among the several objects of the present invention may be noted the provision of a noise reduction circuit which estimates the noise components in an input signal and reduces them; the provision of such a circuit which is small in size and which has minimal power requirements for use in a hearing aid; the provision of such a circuit having a frequency response which is adjustable according to a user’s preference; the provision of such a circuit having a gain which is adjustable according to an expected noise environment; the provision of such a circuit having a gain which is adjustable according to a user’s preference; the provision of such a circuit having a gain which is adjustable according to an existing noise environment; and the provision of such a circuit which produces a noise reduced output signal.

Generally, in one form the invention provides a noise reduction circuit for a sound reproduction system having a microphone for producing an input signal in response to sound in which noise components are present. The circuit includes an adaptive filter comprising a variable filter responsive to the input signal to produce a noise estimating signal and further comprising a first combining means responsive to the input signal and the noise-estimating signal to produce a composite signal. The parameters of the variable filter are varied in response to the composite signal to change its operating characteristics. The circuit further includes a second filter which responds to the noise-estimating signal to produce a modified noise-estimating signal and also includes means for delaying the input signal to produce a delayed signal. The circuit also includes a second combining means which is responsive to the delayed signal and the modified noise-estimating signal to produce a noise-reduced output signal. The variable filter may include means for continually sampling the input signal during predetermined time intervals to produce the noise-estimating signal. The circuit may be used with a digital input signal and may include a delaying means for delaying the input signal by an integer number of samples N to produce the delayed signal and may include a second filter comprising a symmetric FIR filter having a tap length of 2N+1 samples. The circuit may also include means for adjusting the amplitude of the modified noise-estimating signal.

Another form of the invention is a sound reproduction system having a microphone for producing an input signal in response to sound in which noise components are present and a variable filter which is responsive to the input signal to produce a noise-estimating signal. The system has a first combining means responsive to the input signal and the noise-estimating signal to produce a composite signal. The parameters of the variable filter are varied in response to the composite signal to change its operating characteristics. The system further comprises a second filter which responds responsive to the noise-estimating signal to produce a modified noise-estimating signal and also includes means for delaying the input signal to produce a delayed signal. The system additionally has a second combining means responsive to the delayed signal and the modified noise-estimating signal to produce a noise-reduced output signal and also has a transducer for producing sound with a reduced level of noise components as a function of the noise-reduced output signal. The variable filter may include means for continually sampling the input signal during predetermined time intervals to produce the noise-estimating signal. The system may be used with a digital input signal and may include a delaying means an for delaying the input signal by an integer number of samples N to produce the delayed signal and may include a second filter comprising a symmetric FIR filter having a tap length of 2N+1 samples. The system may also include means for adjusting the amplitude of the modified noise-estimating signal.

An additional form of the invention is a method of reducing noise components present in an input signal in the audible frequency range which comprises the steps of filtering the input signal with a variable filter to produce a noise-estimating signal and combining the input signal and the noise-estimating signal to produce a composite signal. The method further includes the steps of varying the parameters of the variable filter in response to the composite signal and filtering the noise-estimating signal according to predetermined parameters to produce a modified noise-estimating signal. The method
also includes the steps of delaying the input signal to produce a delayed signal and combining the delayed signal and the modified noise-estimating signal to produce a noise-reduced output signal. The method may include a filter parameter varying step comprising the step of continually sampling the input signal and varying the parameters of said variable filter during predetermined time intervals. The method may be used with a digital input signal and may include a delaying step comprising delaying the input signal by an integer number of samples N to produce the delayed signal and may include a noise-estimating signal filtering step comprising filtering the noise-estimating signal with a symmetric FIR filter having a tap length of 2N+1 samples. The method may also include the step of selectively adjusting the amplitude of the modified noise-estimating signal.

Other objects and features will be in part apparent and in part pointed out hereinafter.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of a noise reduction circuit of the present invention.
FIG. 2 is a block diagram of a sound reproduction system of the present invention.
FIG. 3 illustrates the present invention embodied in a headset.
FIG. 4 illustrates a hardware implementation of the block diagram of FIG. 2.
FIG. 5 is a block diagram of an analog hearing aid adopted for use with the present invention.

DETAILED DESCRIPTION OF A PREFERRED EMBODIMENT

A noise reduction circuit of the present invention as it would be embodied in a hearing aid is generally indicated at reference numeral 10 in FIG. 1. Circuit 10 has an input 12 which may be any conventional source of an input signal such as a microphone, signal processor, or the like. Input 12 also includes an analog to digital converter (not shown) for analog inputs so that the signal transmitted over a line 14 is a digital signal. The input signal on line 14 is received by an N-sample delay circuit 16 for delaying the input signal by an integer number of samples N, an adaptive filter within dashed line 18, a delay 20 and a signal level adjuster 36.

Adaptive filter 18 includes a signal combiner 22, and a variable filter 24. Delay 20 receives the input signal from line 14 and outputs a signal on a line 26 which is similar to the input signal except that it is delayed by a predetermined number of samples. In practice, it has been found that the length of the delay introduced by delay 20 may be set according to a user’s preference or in anticipation of an expected noise environment. The delayed signal on line 26 is received by variable filter 24. Variable filter 24 continually samples each data bit in the delayed input signal to produce a noise-estimating signal on a line 28 which is an estimate of the noise components present in the input signal on line 14. Alternatively, if one desires to reduce the signal processing requirements of circuit 10, variable filter 24 may be set to sample only a percentage of the samples in the delayed input signal. Signal combiner 22 receives the input signal from line 14 and receives the noise-estimating signal on line 28. Signal combiner 22 combines the two signals to produce an error signal carried by a line 30. Signal combiner 22 preferably takes the difference between the two signals.

Variable filter 24 receives the error signal on line 30. Variable filter 24 responds to the error signal by varying the filter parameters according to an algorithm. If the product of the error and delayed sample is positive, the filter parameter corresponding to the delayed sample is increased. If this product is negative, the filter parameter is decreased. This is done for each parameter. Variable filter 24 preferably uses a version of the LMS filter algorithm for adjusting the filter parameters in response to the error signal. The LMS filter algorithm is commonly understood by those skilled in the art and is more fully described in Widrow, Glover, McCool, Kaunitz, Williams, Hearn, Ziedler, Dong and Goodlin, Adaptive Noise Cancelling: Principles and Applications, Proceedings of the IEEE, 63(12), 1692-1716 (1975), which is incorporated herein by reference. Those skilled in the art will recognize that other adaptive filters and algorithms could be used within the scope of the invention. The invention preferably embodies the binary version of the LMS algorithm. The binary version is similar to the traditional LMS algorithm with the exception that the binary version uses the sign of the error signal to update the filter parameters instead of the value of the error signal. In operation, variable filter 24 preferably has an adaption time constant on the order of several seconds. This time constant is used so that the output of variable filter 24 is an estimate of the persisting or stationary noise components present in the input signal on line 14. This time constant prevents the system from adapting and cancelling incoming transient signals and speech energy which change many times during the period of one time constant. The time constant is determined by the parameter update rate and parameter update value.

A filter 32 receives tile noise estimating signal from variable filter 24 and produces a modified noise-estimating signal. Filter 32 has preselected filter parameters which may be set as a function of the user’s hearing impairment or as a function of an expected noise environment. Filter 32 is used to select the frequencies over which circuit 10 operates to reduce noise. For example, if low frequencies cause trouble for the hearing impaired due to upward spread of masking, filter 32 may allow only the low frequency components of the noise estimating signal to pass. This would allow circuit 10 to remove the noise components through signal combiner 42 in the low frequencies. Likewise, if the user is troubled by higher frequencies, filter 32 may allow only the higher frequency components of the noise-estimating signal to pass which reduces the output via signal combiner 42. In practice, it has been found that there are few absolute rules and that the final setting of the parameters in filter 32 should be determined on the basis of the user’s preference.

When circuit 10 is used in a hearing aid, the parameters of filter 32 are determined according to the user’s preferences during tile fitting session for the hearing aid. The hearing aid preferably includes a connector and a data link as shown in FIG. 2 of U.S. Pat. No. 4,548,082 for setting the parameters of filter 32 during the fitting session. The fitting session is preferably conducted as more fully described in U.S. Pat. No. 4,548,082, which is incorporated herein by reference.

Filter 32 outputs the modified noise-estimating signal on a line 34 which is received by a signal level adjuster 36. Signal level adjuster 36 adjusts the amplitude of the modified noise-estimating signal to produce an amplitude adjusted signal on a line 38. If adjuster 36 is manu-
ally operated, the user can reduce the amplitude of the modified noise-estimating signal during quiet times when there is less need for circuit 10. Likewise, the user can allow the full modified-noise estimating signal to pass during noisy times. It is also within the scope of the invention to provide for the automatic control of signal level adjuster 36. This is done by having signal level adjuster 36 sense the minimum threshold level of the signal received from input 12 over line 14. When the minimum threshold level is large, it indicates a noisy environment which suggests full output of the modified noise-estimating signal. When the minimum threshold level is small, it indicates a quiet environment which suggests that the modified noise-estimating signal should be reduced. For intermediate conditions, intermediate adjustments are set for signal level adjuster 36.

N-sample delay 16 receives the input signal from input 12 and outputs the signal delayed by N-samples on a line 40. A signal combiner 42 combines the delayed signal on line 40 with the amplitude adjusted signal on line 38 to produce a noise-reduced output signal via line 43 at an output 44. Signal combiner 42 preferably takes the difference between the two signals. This operation of signal combiner 42 cancels signal components that are present both in the N-sample delayed signal and the filtered signal on line 38. The numeric value of N in N-sample delay 16 is determined by the tap length of filter 32, which is a symmetric FIR filter with a delay of N-Samples. For a given tap length L, L = 2N + 1. The use of this equation ensures that proper timing is maintained between the output of N-sample delay 16 and the output of filter 32.

When used in a hearing aid, noise reduction circuit 10 may be connected in series with commonly found filters, amplifiers and signal processors. FIG. 2 shows a block diagram for using circuit 10 of FIG. 1 as the first signal processing stage in a hearing aid 100. Common reference numerals are used in the figures as appropriate. FIG. 2 shows a microphone 50 which is positioned to produce an input signal in response to sound external to the hearing aid 100 by conventional means. An analog to digital converter 52 receives the input signal and converts it to a digital signal. Noise reduction circuit 10 receives the digital signal and reduces the noise components in it as more fully described in FIG. 1 and the accompanying text. A signal processor 54 receives the noise reduced output signal from circuit 10. Signal processor 54 may be any one or more of the commonly available signal processing circuits available for processing digital signals in hearing aids. For example, signal processor 54 may include the filter-limit-filter structure disclosed in U.S. Patent No. 4,548,082. Signal processor 54 may also include any combination of the other commonly found amplifier or filter stages available for use in a hearing aid.

After the digital signal has passed through the final stage of signal processing, a digital to analog converter 56 converts the signal to an analog signal for use by a transducer 58 in producing sound as a function of the noise reduced signal.

In addition to use in a traditional hearing aid, the present invention may be used in other applications requiring the removal of stationary noise components from a signal. For example, the work environment in a factory may include background noise such as fan or motor noise. FIG. 3 shows circuit 10 of FIG. 1 installed in a headset 110 to be worn over the ears by a worker or in the worker’s helmet for reducing the fan or motor noise. Headset 110 includes a microphone 50 for detecting sound in the work place. Microphone 50 is connected by wires (not shown) to a circuit 112. Circuit 112 includes the analog to digital converter 52, noise reduction circuit 10 and digital to analog converter 56 of FIG. 2. Circuit 112 thereby reduces the noise components present in the signal produced by microphone 50. Those skilled in the art will recognize that circuit 112 may also include other signal processing as that found in signal processor 54 of FIG. 2. Headset 110 also includes a transducer 58 for producing sound as a function of the noise reduced signal produced by circuit 112.

FIG. 4 shows a hardware implementation 120 of an embodiment of the invention and, in particular, shows an implementation of the block diagram of FIG. 2, but simplified to unity gain function with the omission of signal processor 54. Hardware 120 includes a digital signal processing board 122 comprised of a TMS 32040 14-bit analog to digital and digital to analog converter 126, a TMS 32010 digital signal processor 128, and an EPROM and RAM memory 130, which operates in real time at a sampling rate of 12.5 kHz. Component 126 combines the functions of converters 52 and 56 of FIG. 2 while 128 is a digital signal processor that executes the program in EPROM program memory 130 to provide the noise reduction functions of the noise reduction circuitry 10. Hardware 120 includes an ear module 123 for inputting and outputting acoustic signals. Ear module 123 preferably comprises a Knowles EK 3024 microphone and preamplifier 124 and a Knowles ED 1932 receiver 134 packaged in a typical behind the ear hearing aid case. Thus microphone and preamplifier 124 and receiver 134 provide the functions of microphone 50 and transducer 58 of FIG. 2.

Circuit 130 includes EPROM program memory for implementing the noise reduction circuit 10 of FIG. 1 through computer program “NRDEF.320” which is set forth in Appendix A hereto and incorporated herein by reference. The NRDEF.320 program preferably uses linear arithmetic and linear adaptive coefficient quantization in processing the input signal. Control of the processing is accomplished using the serial port communication routines installed in the program.

In operation, the NRDEF.320 program implements noise reduction circuit 10 of FIG. 1 in software. The reference characters used in FIG. 1 are repeated in the following description of FIG. 4 to correlate the block from FIG. 1 with the corresponding software routine in the NRDEF.320 program which implements the block. Accordingly, the NRDEF.320 program implements a 6 tap variable filter 24 with a single delay 20 in the variable filter path. Variable filter 24 is driven by the error signal generated by subtracting the variable filter output from the input signal. Based on the signs of the error signal and corresponding data value, the coefficient of variable filter 24 to be updated is incremented or decremented by a single least significant bit. The error signal is used only to update the coefficients of variable filter 24, and is not used in further processing. The noise estimate output from the variable filter 24 is low pass filtered by an 11 tap linear phase filter 32. This lowpass filtered noise estimate is then scaled by a multiplier (default=1) and subtracted from the input signal delayed 5 samples to produce a noise-reduced output signal.

FIG. 5 illustrates the use of the present invention with a traditional analog hearing aid. FIG. 5 includes an analog to digital converter 52, an acoustic noise reduc-
As various changes could be made in the above constructions without departing from the scope of the invention, it is intended that all matter contained in the above description or shown in the accompanying drawings shall be interpreted as illustrative and not in a limiting sense.

**APPENDIX A**

```
PROGRAM 'nrdef.320'

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818 S. Euclid
Saint Louis, Missouri 63110

This program is based on the 50 tap adaptive filter program 'nr'
In this program the noise estimate is low passed filtered with
X tap linear phase lowpass filter, scaled and used to cancel an
appropriately delayed input signal. The error term used in the
adaptive filter update remains the same. The coefficient update
uses a leaky coefficient form such that:

\[ w(k,\tau+1) = w(k,\tau) (1 - \text{leak}) + \text{delta} \]

where leak and delta are programmable.

This program also includes the serial port communication protocol
allow the program parameters to be adjusted through the serial
communication port.

The dc offset from the input is removed using and adaptive null
which subtracts an offset from the input to generate a zero mean
input stream.

50 tap adaptive filter using the sign-update method

This program implements a 50 tap (or smaller) adaptive filter using
the sign bit update method. The program is designed to use the
32010 DSP board with the AIC acting as both A/D and D/A.

The adaptive structure implemented is

\[
\begin{align*}
  x(n) \quad \text{--------------------------}\rightarrow & \quad \text{---------------------}\rightarrow \text{err} \\
  \quad \quad \text{--------------------------} & \quad \text{---------------------} \\
  \quad \quad \quad \text{--------------------------} & \quad \text{---------------------} \\
  \quad \quad \quad \quad \text{--------------------------} & \quad \text{---------------------} \\
  \quad \quad \quad \quad \quad \text{--------------------------} & \quad \text{---------------------} \\
  \quad \quad \quad \quad \quad \quad \text{--------------------------} & \quad \text{---------------------} \\
\end{align*}
\]

The output signal is
**The default conditions for this program are:**

- 6 tap adaptive filter
- non-leaking coefficients
- 1 LSB update of adaptive coefficients
- unity sensitivity term (32767 where 32768 is unity)

**DATA AREAS**

- page 0
  - 0 - 50 input samples
  - 51 - 100 adaptive filter coefficients
- page 1
  - 0 - 11 noise estimate samples

**Page 0 data locations**

```plaintext
d0  equ   0  input data x(n)
d5  equ   5  input data x(n-5)
d49 equ  49  input data x(n-49)
d50 equ  50  input data x(n-50)
w0  equ  51  adaptive FIR coefficient w(0)
w49 equ 100  adaptive FIR coefficient w(49)
y   equ  101 adaptive filter output (estimate)
err equ  102 estimate error \( \cdot err = x(n) - y(n) \)
temp equ  103 temporary working location
delta equ  104 coefficient update magnitude / 2
lpest equ  105 low pass filtered noise estimate
sens equ  106 noise reduction sensitivity term
dcoff equ  107 adaptive dc offset nulling term
taps equ  108 number of adaptive filter taps - 1
leak equ  109 leaky coefficient multiplier
```

**Serial communication locations**
* serial equ 118 serial input data from uart
serout equ 119 serial output data to uart
cad6 equ 120 hex value of valid input
cdata equ 122 data from serial port communication
cdata equ 123 data memory address containing 1
* page 1 data locations
* y0 equ 0 current noise estimate y(n)
y10 equ 10 noise estimate y(n-10)
* * AORG 0 start hard reset vector
* * AIC interrupt routine
* sint in din,0 read a/d input sample
out dout,0 output d/a sample
pop load return address into accumulator
add one,1 add offset to return address
push store new return address
eint enable interrupts and clear interf
ret return from interrupt call
* * bmask data >fff8 output bit mask
fsrta data >01c8 r/a data for 12.25 kHz sampling
fsrth data >448a r/b data for 12.25 kHz sampling
ksens data 32767 default noise reduction sensitivity
* * Program initialization
* * start dint disable interrupts from AIC
ldpk 0 load data page pointer to page 0
sovrm set overflow clipping mode
lack ksens default noise reduction sensitivity
tb1r sens read noise reduction sensitivity
lack 2 load coefficient delta value
sacl delta store coefficient delta value
lack 5 load number of taps - 1
sacl taps store the desired number of taps - 1
lack >0 default coefficient leak term [1 + leak/2^16]sacl leak store default leak term
* * clear coefficients and data areas
* (start at cdat to clear filter taps without resetting
* model parameters)
cdat cld larp 0 use aux reg. 0
lack 0,100 set word counter to 100
zac clear accumulator
sacl * clear lower 100 data locations
banz cld branch until all locations clear
* lark 0,50 initialize ARO to 50
lark 1,0 initialize ARI to 0

*  
* start point for resetting parameters  
* (this does not set delta, sens, or the number of taps)  
* (does not clear filter taps)  
*  
* start  
int 0 disable interrupts from AIC  
link 0 load data page pointer to page 0  
sovm set overflow clipping mode  
ack bmask output bit mask  
tblr mask read bit mask  
ack 1 load one (1) in accumulator  
sacl one store value of 1 in one  

This code is used to set the sampling rate and AIC configuration  

*  
zac  clear accumulator  
sacl dout zero output data to AIC  
cut dout,0 clear AIC serial register  
cut dout,7 reset AIC  
cut dout,7 reset AIC  
out dout,0 clear AIC serial register  

*  
eint enable interrupts  

h1 b hl ignore first interrupt  

*  
lack 3 data to initiate secondary communication  
sacl dout store data in interrupt region  
c0 b c0 wait for interrupt  
lack fsta ta/ra settings  
tblr dout read ta/ra settings  
c1 b cl wait for interrupt  
lack 3 data to initiate secondary communication  
sacl dout store data in interrupt region  
c2 b c2 wait for interrupt  
lack faitb tb/rb settings  
tblr dout read tb/rb settings  
c3 b c3 wait for interrupt  
lack 3 data to initiate secondary communication  
sacl dout store data in interrupt region  
c4 b c4 wait for interrupt  
lack >63 AIC data for no aac / 3V FS / in+ input  
sacl dout store AIC settings  
c5 b c5 wait for interrupt  
zac  clear accumulator  
sacl dout store output sample of 0  
c6 b c6 wait for interrupt  

This is the region in which the main program sampling loop is executed.  

*  
null the input dc offset  

loop  
lac din.12 load new input sample  
sub dcoff,3 subtract dc offset  
sean din.4 store input with dc term nulled  
bgt inoff branch if offset input signal positive  

*  
lac dcoff load adaptive dc offset term  
sub one reduce offset term  
sacl dcoff store new offset
5,412,735

* filter branch to adaptive filter code

incoff lac dcoff load adaptive dc offset term
add one increase offset term
sacl dcoff store new offset

* calculate the adaptive filter output

filter sac clear accumulator
lt d49 load x(n-49) into T register
mpy w49 P reg. = x(n-49)*w(49)
ltd 48 load x(n-48) in T reg., accumulate, Z^*-1
mpy 99 P reg. = x(n-48)*w(48)
ltd 98
mpy 97
ltu 45
mpy 96
lti 44
mpy 95
lti 43
mpy 94
lti 42
mpy 93
lti 41
mpy 92
lti 40
mpy 91
lti 39
mpy 90
lti 38
mpy 89
lti 37
mpy 88
lti 36
mpy 87
lti 35
mpy 86
lti 34
mpy 85
lti 33
mpy 84
lti 32
mpy 83
lti 31
mpy 82
lti 30
mpy 81
lti 29
mpy 80
lti 28
mpy 79
lti 27
mpy 78
lti 26
mpy 77
lti 25
mpy 76
lti 24
mpy 75
lti 23
mpy 74
lti 22
mpy 73
lti 21
calculate estimate error (assume delay of one)

lac din load current input x(n+1)
sacl d0 store new input sample in array
sub y subtract estimate err = x(n+1) - y(n)
sacl err store error

update a single filter coefficient using the sign bit method

-AR0 counts from 50 to 1, w(k) to be updated has address <AR0> + 50, applicable data x(n-k) has address <AR0>

sar 0,temp store x(n-k) pointer in location temp
lack 50 load w(k) offset in accumulator
-addr 1, temp add coefficient pointer value
sacl temp store w(k) coefficient address in temp
lar 1,temp load w(k) address in A1
lt  *,1 load x(n-k) in to T register, set ARP=1
mpy err  err * x(n-k) in P reg.
adc  nprd load accumulator with product
bit nprd branch if err * x(n-k) is negative

add delta to w(k)

lac delta,15 coefficient delta in accumulator
upat b updat branch to update code

subtract delta from w(k)

nprd zac clear accumulator
sub delta,15 negative coefficient delta in accumlat

update w(k) using address stored in AR1

updat add  *,15 add ,w(k) to current delta
add  *,15 add w(k) again to make use of overflow processi
lt  * load w(k) in T reg. for leak term
mpy leak multiply by leak term
sapc subc scaled w(k) for leak
zac  *,0,0 store updated w(k), set ARP=0

update the coefficient pointer AR0

mar  *,0 subtract one from AR0 to offset count (49-0)
banz cntok branch if coefficient counter not zero
lar 0,taps reset coefficient counter
cntok  mar  *,+o add one to AR0 to use again as address pointer

low pass filter and scale the noise estimate

lac y load current noise estimate in accumulator
ldp 1 change to data page 1
sacl y0 store current noise estimate in page 1

lowpass filter ( 1 kHz SW, -40 dB at 3kHz)

zac clear accumulator
lt  y10 load y(n-10) in T register
mpy  -59 multiply by h(10)
lit  9 load y(n-9) in T register, accumulate, Z**-1
mpy  -68 multiply by h(9)
lit  8
mpy  113
lit  7
mpy  545
lit  6
mpy  1036
lit  5
mpy  1255
lit  4
mpy  1036
lit  3
mpy  545
lit  2
mpy  113
lit  1
mpy  -68
lit  y0 load y(n) in T register, accumulate, Z**-1
mpy  -59 multiply by h(0)
apc accumulate last product
ldp  0return to data page 0
program gencom.320

This program contains routines for communication via an RS232 line and the TMS32010 board. It contains routines to read and write to the data and program memory, and begin execution at a given location.

The command formats are as follows:

start execution at address xxxx
write data to program memory starting at address xxxx
read data from program memory address
write data to data memory starting at address xxxx
read data from data memory address
write data xxx to WDHA interface
read data XXX from WDHA interface
read WDHA serial output line, 0000 if low, 0001 if high

communication routines for the log DEA evaluation system

At this point a character has been received through the serial interrupting program execution. The subroutine used to service the serial port will be called. If program control returns to this point from 'getch' a character other than '/' has been received. Further program execution will halt until a valid character has been received.

disable AIC interrupts
get character input routine
wait for valid '/' character

This portion begins the command interpretation portion of the program control passes to this point whenever an '/' character received.

get command character
load received command value
branch to execute routine
check for 1 command
branch to load program memory
check for 2 command
branch to read program memory
check for 3 command
branch to load data memory routine
check for 4 command
branch to read data memory routine
check for 5 command
branch to write wdha routine
check for 6 command
branch to read wdha routine
check for 7 command
branch to check wdha serial output bit
branch to get valid control sequence

call word input routine to get address
load starting address
jump to desired starting location

call word input routine to get address
load new word
store command address

call word input to get data
load new word
store command data
load write address
write data
increment address
store new address
branch for new word

call word input routine to get address
load address in accumulator
read memory contents
send word to host
read next command

load address in accumulator
store starting address for write to mem
load data into accumulator
select aux register 1
load program memory address in aux reg.
store new data increment, increment add
store updated address in cadd
select aux register 0
branch for next data input

branch for new word

load address in aux. reg. 1

call word input routine to get address
load address in aux. reg. 1
25

| LARP | 1            | select aux reg. 1           |
| LA C | *            | read data memory location  |
| SA C L | word        | store data from memory location |
| LARP | 0            | select aux reg. 0           |
| CALL | sword        | call send word routine     |
| b    | charin       | read next command          |

* * *

write to wdha routine

**wwdha**

| CALL | gword        | word input routine to get data for wdha |
| LAC  | one,15      | set wdha data in high for leading 1 |
| SA C L | cadd        | use cadd for working location |
| OUT  | cadd,6      | clear wdha clocks to 0 |
| LAC  | one,15      | set wdha data in high for leading 1 |
| ADD  | one,14      | set wdha clkin high |
| SA C L | cadd        | store wdha output signals |
| OUT  | cadd,6      | clock in leading 1 |
| ZAC  |            | clear accumulator |
| SA C L | cadd        | low clock signals |
| OUT  | cadd,6      | output low clock signals |
| LARP | 1           | select aux reg 0 |
| LARK | one,15      | store bit shift counter |
| WR0  |            | mask for data bit |
| LAC  | one,15      | mask off high order bit |
| SA C L | cdata       | store output data bit |
| OUT  | cdata,6     | output data bit to wdha, clkin low |
| LAC  | one,14      | set clkin high |
| OR   | cdata       | add data bit |
| SA C L | cdata       | store data bit, clkin high |
| OUT  | cdata,6     | clock in data to wdha |
| LAC  | word,1      | shift data word |
| SA C L | word        | store shifted output word |
| BANZ | wr0         | branch for next bit output |
| LARP | 0           | select aux. register 0 |
| b    | charin      | branch for next command |

* * *

wdha read word routine

**rwdha**

| ZAC  |            | clear accumulator |
| SA C L | word       | clear input data word |
| OUT  | word,6     | set clk out low |
| LARP | 1           | select aux reg 0 |
| LARK | one,15      | store bit shift counter |
| RO   |            | shift building input word |
| SA C L | word       | store shifted word |
| IN   | cdata,6    | read data out bit |
| LAC  | cdata,1    | shift data by 1 left |
| SA C H | cdata       | store new bit |
| LAC  | one         | set low order bit |
| AND  | cdata       | mask off new bit |
| OR   | word        | add bit to low order bit of word |
| SA C L | word       | store word |
| LAC  | one,13      | set clk out bit |
| SA C L | cdata       | store clk out bit |
| OUT  | cdata,6    | set clk out high, generate leading edge |
| ZAC  |            | clear accumulator |
| SA C L | cdata       | clear clk out bit |
| OUT  | cdata,5    | set clk out low |
| BANZ | r0          | branch until all bits read |
| LARP | 0           | select aux reg. 0 |
| CALL | sword       | call word send routine |
| b    | charin      | wait for next command |

* * *

check wdha serial output bit
cwdha in cdata,6 read wdha serial output bit
and cdata check serial input bit
bz bitlow branch if bit low
lac one,15 mask for wdha serial bit
lac one load one in accumulator
sacl word store 0001 in output word
b cw0 branch to send word out

bitlow zac clear accumulator
sacl word store 0000 in output word
cw0 call sword call word send routine
b charin wait for next command

* word send routine (output word passed in word)
*
sword lac word,4 shift first nibble into upper accumulat
sach cdata store nibble
lack 13 4 low order bit mask
and cdata mask nibble
* sacl cdata store nibble to be output
call sendch call send character routine
lac word,3 shift second nibble into upper accumula
sach cdata store nibble
lack 15 4 low order bit mask
and cdata mask nibble
sacl cdata store nibble to be output
call sendch call send character routine
lac word,12 shift third nibble into upper accumulat
sach cdata store nibble
lack 15 4 low order bit mask
and cdata mask nibble
sacl cdata store nibble to be output
call sendch call send character routine
ret return from sword

* send character routine (output nibble in cdata)
*
sendch larp 1 load auxiliary pointer to 1 for delay
lack 9 load 9 in accumulator
sub cdata check for chars 0-9
biz saf branch if value A-F
lack 48 base ascii offset for 0-9
add cdata prepare ascii character
sacl cdata store ascii code for 0-9
b sc0 branch to serial output processing
saf lack 55 base ascii offset for A-F
add cdata prepare ascii character
sacl cdata store ascii code for A-F
b sc0 branch to serial output processing
delay lack 1,40 delay counter for trans buffer to empty
del0 banz del0 delay loop
larp 0 select aux reg. 0
sc0 bicz tbeck check for pending input character
b charin check for new command
tbeck in serin,1 read serial input register
lac one,10 mask for the bit
and serin check the bit
bz delay if buffer full branch to delay
out cdata,1 output character to UART
ret return from sendch

* word construct routine (results returned in word)
serial input routine

```
getch bios getch wait for serial input
larp 1 select aux reg 1
lark 1,10 store delay counter
cwait banz cwait wait for uart registers
larp 0 select aux reg 0
* in serin,1 read serial input register
* check for '/ ' ([ESC])
* lack >ff load 8 bit low order mask
and serin load input data into accumulator
sacl serin store data only
sacl serout store input data (prepare for echo)
lack 47 load '/ ' ([ESC]) code in accumulator
sub serin compare input
bs escin branch if '/ ' ([ESC]) command character
* check for 0-9 hex character
* lack 48 ascii code for 0
sacl temp store ascii offset
lac serin load serin in accumulator
sub temp subtract offset for ascii 0
biz insert branch (<0) to invalid character routine
sacl serin store shifted serin
lack 9 ascii code offset for 9
...sacl temp store ascii offset
...lac serin load input data
sub temp subtract 9
bgsz not09 branch if serin > 9
lac serin load value 0-9 in accumulator
sacl value store input character value
b good branch to character echo routine
* * check for A-F hex character
```
not09 lack 17 additional offset for A-F
sacl temp
lac serin load input data
sub temp subtract new offset
biz inerr branch (<0) to invalid character routine
sac1 serin store shifted serin
lack 5 ascii code offset
sac1 temp store ascii offset
lac serin load input data
sub temp subtract 5
bgz inerr branch if serin > 5
lack 10 load value for hex A
add serin add input data
sac1 value store input character value
b good branch to character echo routine

* valid character echo
* good out serout,1 output valid character
ret return from character input
*
* invalid character echo
* inerr lack 33 ascii code for !
sac1 serout store character to be echoed
out serout,1 output character
zap one -1 in accumulator
sac1 value store -1 in value
ret return from character input
*
* '/' character echo
* escin out serout,1 output '/' character
pop b comman branch to command interpretation
*
* bell larp 1 select aux reg. 1
lack 1,127 store delay counter
waits banz waitb wait for uart registers
larp 0 select aux reg. 0
*
* b10z bell2 branch if no pending character
b charin branch to serial input handler
bell2 in serin,1 read serial input register
lac one,10 mask for the bit
and serin check the bit
bz bell if buffer full branch to bell
*
* lack 7 ascii bell in accumulator
sacl serout store bell character
out serout,1 send bell character
b bell send another bell
*
end
What is claimed is:

1. A noise reduction circuit for a sound reproduction system having a microphone for producing an input signal in response to sound in which a noise component is present, said circuit comprising:
   an adaptive filter including a variable filter responsive to the input signal for producing a noise-estimating signal and further including a first combining means responsive to the input signal and the noise-estimating signal for producing a composite signal;
   said variable filter having parameters which are varied in response to the composite signal to change the operating characteristics thereof;
   a second filter for filtering the noise-estimating signal to produce a filtered noise-estimating signal;
   means for delaying the input signal to produce a delayed signal; and
   second combining means for combining the delayed signal and the filtered noise-estimating signal to attenuate noise components in the delayed signal and for producing a noise-reduced output signal.

2. The circuit of claim 1 wherein the variable filter comprises means for sampling a percentage of the input signal to produce the noise-estimating signal which is a function of the noise components during said time intervals.

3. The circuit of claim 1 or 2 wherein the input signal is a digital signal; wherein the delaying means comprises means for delaying the input signal by an integer number of samples N to produce the delayed signal; and wherein the second filter comprises a symmetric FIR filter having a tap length of 2N+1 samples.

4. The circuit of claim 1 or 2 further comprising means for adjusting the amplitude of the filtered noise-estimating signal to produce an amplitude adjusted signal, and wherein the second combining means is responsive to the delayed input signal and the amplitude adjusted signal.

5. The circuit of claim 4 wherein the input signal is a digital signal and wherein the circuit further comprises means for delaying the input signal by a preset number of samples to produce a preset delayed signal; and wherein the variable filter is responsive to the preset delayed signal to produce the noise-estimating signal.

6. The circuit of claim 1 or 2 wherein the first combining means comprises means for taking the difference between the input signal and the noise-estimating signal and wherein the second combining means comprises means for taking the difference between the delayed input signal and the filtered noise-estimating signal.

7. The circuit of claim 1 or 2 wherein the input signal is a digital signal and wherein the circuit further comprises means for delaying the input signal by a preset number of samples to produce a preset delayed signal, and wherein the variable filter is responsive to the preset delayed signal to produce the noise-estimating signal.

8. The circuit of claim 1 or 2 wherein the sound reproduction system is a hearing aid for use by the hearing impaired and wherein the second filter has filter parameters which are selected as a function of a user's hearing impairment.

9. The circuit of claim 1 or 2 wherein the second filter has filter parameters which are selected as a function of expected noise components.

10. A sound reproduction system comprising:
    a microphone for producing an input signal in response to sound in which noise components are present;
    a variable filter responsive to the input signal to produce a noise-estimating signal;
    a first combining means responsive to the input signal and the noise-estimating signal for producing a composite signal;
    said variable filter having parameters which are varied in response to the composite signal to change the operating characteristics thereof;
    a second filter for filtering the noise-estimating signal to produce a filtered noise-estimating signal;
    means for delaying the input signal to produce a delayed signal;

11. The system of claim 10 wherein the variable filter comprises means for sampling a percentage of the input signal to produce the noise-estimating signal which is a function of the noise component during said time intervals.

12. The system of claim 10 or 11 wherein the input signal is a digital signal; wherein the delaying means comprises means for delaying the input signal by an integer number of samples N to produce the delayed signal; and wherein the second filter comprises a symmetric FIR filter having a tap length of 2N+1 samples.

13. The system of claim 10 or 11 further comprising means for adjusting the amplitude of the filtered noise-estimating signal to produce an amplitude adjusted signal, and wherein the second combining means is responsive to the delayed input signal and the amplitude adjusted signal.

14. The system of claim 13 wherein the input signal is a digital signal and wherein the system further comprises means for delaying the input signal by one sample to produce a predetermined delayed signal; and wherein the variable filter is responsive to the predetermined delayed signal to produce the noise-estimating signal.

15. The system of claim 10 or 11 wherein the first combining means comprises means for taking the difference between the input signal and the noise-estimating signal and wherein the second combining means comprises means for taking the difference between the delayed input signal and the filtered noise-estimating signal.

16. The system of claim 10 or 11 wherein the input signal is a digital signal and wherein the system further comprises means for delaying the input signal by one sample to produce a predetermined delayed signal; and wherein the variable filter is responsive to the predetermined delayed signal to produce the noise-estimating signal.

17. The system of claim 10 or 11 wherein the sound reproduction system is a hearing aid for use by the hearing impaired and wherein the second filter has filter parameters which are selected as a function of a user's hearing impairment.

18. The system of claim 10 or 11 wherein the second filter has filter parameters which are selected as a function of expected noise components.
19. A method of reducing noise components present in an input signal in the audible frequency range comprising the steps of:
in an input signal in the audible frequency range coming signal which is a function of the noise components between the input signal and the noise-estimating signal to produce a filtered noise-estimating signal; delaying the input signal to produce a delayed signal; and combining the delayed signal and the filtered noise-estimating signal to attenuate noise components in the delayed signal to produce a noise-reduced output signal.

20. The method of claim 19 wherein the filter parameter varying step comprises the step of continually sampling the input signal and varying the parameters of said variable filter during predetermined time intervals, whereby said variable filter produces the noise-estimating signal which is a function of the noise components during said time intervals.

21. The method of claim 19 or 20 wherein the input signal is a digital signal; wherein the delaying step comprises delaying the input signal by an integer number of samples N to produce the delayed signal; and wherein the noise-estimating signal filtering step comprises filtering the noise-estimating signal with a symmetric FIR filter having a tap length of 2N+1 samples.

22. The method of claim 19 or 20 further comprising the step of selectively adjusting the amplitude of the filtered noise-estimating signal to produce an amplitude-adjusted signal, and wherein the second stated combining step comprises combining the delayed signal and the amplitude-adjusted signal.

23. The method of claim 22 wherein the input signal is a digital signal and wherein the method further comprises the step of delaying the input signal by a predetermined number of samples to produce a predetermined delayed signal; and wherein the first stated filtering step comprises filtering the predetermined delayed signal to produce the noise-estimating signal.

24. The method of claim 19 or 20 wherein the first stated combining step comprises taking the difference between the input signal and the noise-estimating signal and wherein the second stated combining step comprises taking the difference between the delayed input signal and the filtered noise-estimating signal.

25. The method of claim 19 or 20 wherein the input signal is a digital signal and wherein the method further comprises the step of delaying the input signal by a predetermined number of samples to produce a predetermined delayed signal; and wherein the first stated filtering step comprises filtering the predetermined delayed signal to produce the noise-estimating signal.

26. The method of claim 19 or 20 as utilized in a sound reproduction system for use by the hearing impaired and wherein the noise-estimating signal filtering step comprises selecting the predetermined filter parameters as a function of a user's hearing impairment.

27. The method of claim 19 or 20 wherein the noise-estimating signal filtering step comprises selecting the predetermined filter parameters as a function of expected noise components.

28. The method of claim 22 wherein the step of adjusting the amplitude of the filtered noise-estimating signal comprises the step of making the adjustment as a function of the amplitude of the input signal.

29. The system of claim 10 or 11 further comprising a headband for a user's head and wherein the transducer is positioned on the headband adjacent the user's ear.

30. A hearing aid comprising:
a microphone for producing an input signal in response to sound in which noise components are present; a variable filter responsive to the input signal to produce a noise-estimating signal; a first combining means responsive to the input signal and the noise-estimating signal for producing a noise-reduced output signal.

31. The hearing aid of claim 30 wherein the variable filter comprises means for selecting the predetermined parameters as a function of a user's hearing impairment.

32. The hearing aid of claim 30 or 31 wherein the input signal is a digital signal; wherein the delaying means comprises means for delaying the input signal by an integer number of samples N to produce the delayed signal; and wherein the second filter comprises a symmetric FIR filter having a tap length of 2N+1 samples.

33. The hearing aid of claim 30 or 31 further comprising means for adjusting the amplitude of the filtered noise-estimating signal to produce an amplitude adjusted signal, and wherein the second combining means is responsive to the delayed input signal and the amplitude adjusted signal.

34. The hearing aid of claim 33 wherein the input signal is a digital signal and wherein the hearing aid further comprises means for delaying the input signal by one sample to produce a predetermined delayed signal; and wherein the variable filter is responsive to the predetermined delayed signal to produce the noise-estimating signal.

35. The hearing aid of claim 30 or 31 wherein the first combining means comprises means for taking the difference between the input signal and the noise-estimating signal and wherein the second combining means comprises means for taking the difference between the delayed input signal and the filtered noise-estimating signal.

36. The hearing aid of claim 30 or 31 wherein the input signal is a digital signal and wherein the hearing aid further comprises means for delaying the input signal by one sample to produce a predetermined delayed signal; and wherein the variable filter is responsive to the predetermined delayed signal to produce the noise-estimating signal.
37. The hearing aid of claim 30 or 31 for use by the hearing impaired and wherein the second filter has filter parameters which are selected as a function of a user's hearing impairment.

38. The hearing aid of claim 30 or 31 wherein the second filter has filter parameters which are selected as a function of expected noise components.

39. A noise reduction circuit for a sound reproduction system having a microphone for producing an input signal in response to sound in which a noise component is present, said circuit comprising:

   - an adaptive filter including a variable filter responsive to the input signal for producing a noise-estimating signal and further including a first combining means responsive to the input signal and the noise-estimating signal for producing a composite signal; said variable filter having parameters which are varied in response to the composite signal to change the operating characteristics thereof;
   - means for adjusting the amplitude of the noise-estimating signal to produce an amplitude adjusted signal; and
   - second combining means for combining the input signal and the amplitude adjusted signal to attenuate noise components in the input signal and for producing a noise-reduced output signal.

* * * * *