IMPLEMENTATION OF BLOCK LEAST MEAN SQUARE ADAPTIVE ALGORITHM FOR EFFECTIVE NOISE CANCELLATION IN SPEECH SIGNAL

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ABSTRACT

Noise cancellation is a common occurrence in today telecommunication systems. Adaptive filter is one of the most important areas in digital signal processing. This paper explores the removal of noise from noise corrupted audio speech signals. An adaptive FIR filter with BLMS algorithm is developed to cancel the noise from the audio speech signal. The BLMS algorithm which is one of the most efficient criteria for determining the values of the adaptive noise cancellation coefficients are very important in communication systems. The major advantage of the proposed system is its ease of implementation and fast convergence. The proposed algorithm is applied to noise canceling problem of long distance communication channel. The algorithm was implemented in Mat lab and was tested for noise cancellation in speech signals.

Keywords: Noise cancellation, BLMS Adaptive Algorithm, Simulink Model.

I. INTRODUCTION

The speech signal which convey information from one to another with a band width of only 8 KHz. speech signal also carries the information with the emotion of a human voice. The speech signal has certain properties: It is a one-dimensional signal, with time as its independent variable, it is random in nature, it is non-stationary. The audible frequency range for the human being is 20Hz to 20 kHz, the human speech has significant frequency components only up to 8 kHz. The most common problem in speech processing is the effect of noise in speech signals. The unwanted noise masks the speech signal and reduces its Intelligibility. The unwanted noise can come from sources such as ventilation equipment, traffic, crowds and commonly, reverberation and echoes. It can also arise electronically from thermal noise, tape hiss or distortion products. If the sound system has unusually large peaks in its frequency response, the speech signal can even end up making itself. One relationship between the strength of the speech signal and the masking sound is called the signal-to-noise ratio, expressed in decibels. Ideally, the S/N ratio is greater than 0dB, indicating that the speech is louder than the noise. The signal type that is used is an 8 KHz audio speech signal which could be in the form of speech or music. Therefore, in order to accomplish the needs of project a digital filter is to be designed. That is, the input audio signal will be correlated with self-generated background noise and then the designed digital filter should try to eliminate or reduce the noise. Thus the project requires first to identify a suitable digital filter and update algorithm and then its implementation.

II. ADAPTIVE FILTER THEORY

An Adaptive filter is a filter that self adjusts its transfer function according to an optimization algorithm driven by an error signal .Because of the complexity of the optimization algorithm , most adaptive filter are digital. As the power of digital signal processing has

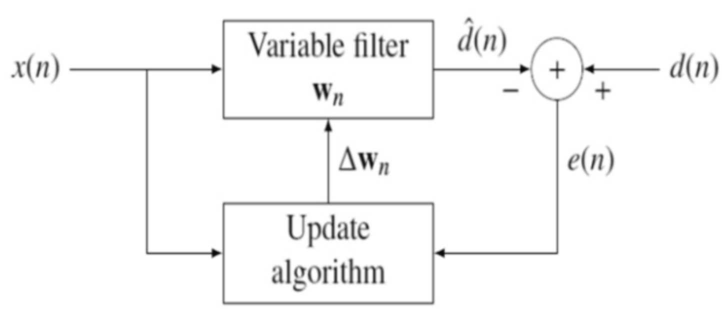


Fig. 1. General Block Diagram for Adaptive Filter

increased, adaptive filters have become much more common and are now routinely used in devices such as mobile phones and other communication devices, cam corders and digital cameras and medical monitoring equipment. The block diagram, shown in the following figure (1), serves as a foundation for particular adaptive filter realizations, such as LMS and RLS. The idea behind the block diagram is that a variable filter extracts an estimate of the desired signal.

The input signal is the sum of a desired signal d(n) and noise signal v(n). ie., x(n) = d(n) + v(n).The variable filter has a finite impulse response (FIR) structure. For such structures the impulse response is equal to the filter coefficients. The coefficients for a filter of order P are defined as

$$Wn = [\ wn\,(o),\ wn\,(1) \ldots\ wn\,(p)]^T \qquad \ldots (1)$$

The error signal or cost function is the difference between the desired and the estimated signal

$$e\,(n) = d\,(n) - d\,(n) \qquad \ldots (2)$$

The variable filter estimates the desired signal by convolving the input signal with the impulse response. In vector notation this is expressed as

$$d\,(n) = wn^{*}\,x\,(n), \qquad \ldots (3)$$

Where x(n) $= [x\,(n), x\,(n-1), \ldots x\,(n-p)]^T$

is an input signal vector. Moreover, the variable filter updates the filter coefficients at every time instant

$$Wn + 1 = Wn + \Delta\ Wn \qquad \ldots (4)$$

Where Δ Wn is a correction factor for the filter coefficients. The adaptive algorithm generates this correction factor based on the input and error signals. LMS, RLS and BLMS are the different coefficient update algorithms.

III. BLOCK IMPLEMENTATION OF ADAPTIVE FILTER

A block of samples of the filter input and desired output are collected and then processed together to obtain a block of output samples. A good measure of computational complexity in a block processing system is given by the number of operations required to process one block of data divided by the block length.

A. Block Processing

Let us consider the data used for modifying the parameters is grouped in blocks of length Las shown if figure (2). The variables defined at time instants $n = kL + i$: the input signal $u\,(kL + i)$ the output of the filter

$$y\,(kL + i) = wT\,(k)\ u\,(kL + i) \qquad \ldots (5)$$

the error signal $e\,(kL + i)$ The parameter vector, w(k), is defined only at time instants Ki

B. Block Lms Algorithm

Let the given input signal samples be $\{\ U(1), U(2), U(3), \ldots\ldots\ \}$ and the desired signal samples be $\{\ d\,(1), d\,(2), d\,(3), \ldots\ldots\ \}$ Correlated with $\{\ U(1), U(2), U(3), \ldots\ldots\ \}$

The algorithm of Block LMS is as follows: Figure (3) shows the block diagram of Block LMS algorithm.

Step 1: Initialize the algorithm with an arbitrary parameter vector w(0), for example $w\,(0) = 0$.

Step 2: Iterate for $k = 0, 1, 2, 3\ldots\ldots\ kmax$ (K is the block index) and initialize $\phi = 0$

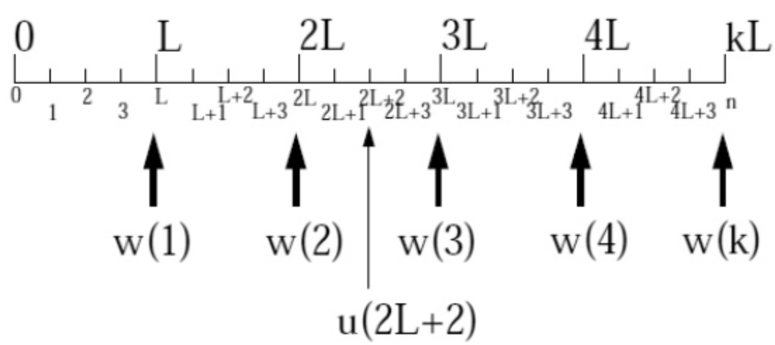


Fig. 2. Block Processing

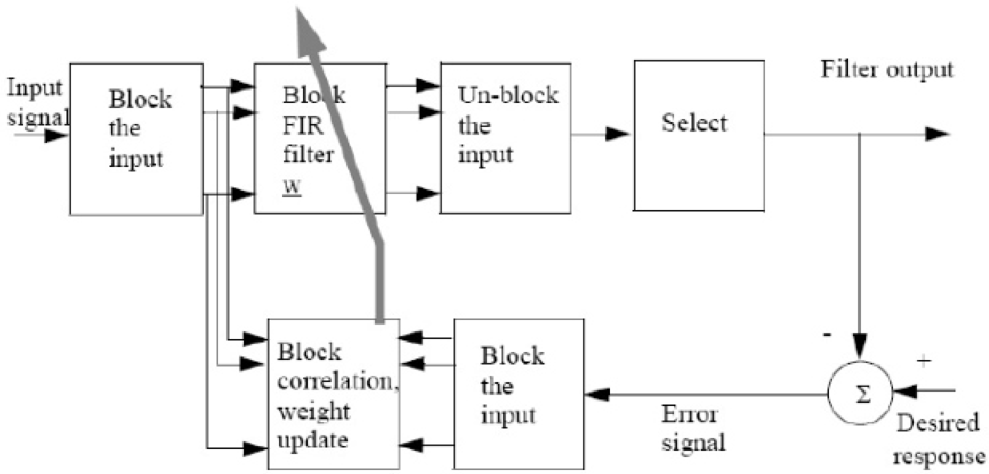


Fig. 3. Block diagram of Block LMS algorithm

Step 3: Iterate for $i = 0, 1, 2, 3 \ldots\ldots (L-1)$ generate a new data pair ,

$$(u(kL+i), d(kL+i)) \quad \text{... (6)}$$

Filter output:

$$y(kL+i) = w(k)\, T(u(kL+i) = \sum_{i=0}^{M-1} \begin{matrix} Wj(k) \\ u(kL+i-j) \end{matrix} \quad \text{... (7)}$$

Output error:

$$e(kL+i) = d(kL+i) \ldots y(kL+i). \quad \text{... (8)}$$

$$\text{Accumulate } \phi + \mu\, e(kL+i)\, u(kL+i) \quad \text{... (9)}$$

Step 4: parameter adaptation:

$$w(K+1) = w(k) + \phi \quad \text{... (10)}$$

Step 5: Complexity of the algorithm: $2M+1$ multiplication and $2M+M/L$ addition per iteration.

C. Block LMS Algorithm By Approximating the gradient by time average

The criterion

$$J = E\, e^2(n) = E(d(n) - w(n)\, T\, u(n))^2 \quad \text{... (11)}$$

has the gradient with respect to the parameter vector w(n),

$$\nabla\, w(n) J = -2\, Ee(n)\, u(n) \quad \text{... (12)}$$

The adaptation of parameter in the Block LMS algorithm is

$$w(k+1) = w(k) + \mu \sum_{i=0}^{L-1} u(kL+i)\, ew(k)$$

$$= (kL+1) \quad \text{... (13)}$$

and denoting $\mu B = \mu L$, The adaptation can be rewritten

$$w(k+1) = w(k) + \mu B \left[\frac{1}{L} \sum_{i=0}^{L-1\; u(kL+i)\, ew(k)} (kL+1) \right]$$

$$= w(k) - \mu B \frac{1}{2} \nabla\, w(k)\, J \quad \text{... (14)}$$

$$\text{Where } \nabla\, w(k)\, J = \frac{1}{L} \sum_{i=0}^{L-1\; u(kL+i)\, ew(K)} (kL+1) \quad \text{... (15)}$$

This shows that expectation in the expression of the gradient is replaced by time average.

D. Convergence Properties of the Block Lmsalgorithm

1. The convergence behavior of the BLMS is governed by the Eigen values of the correlation matrix

$$R = E[X(n)\, X^T(n)]. \quad \text{... (16)}$$

2. The BLMS algorithm has N modes of convergence which are characterized by the time constants

$$T_{Bi} = \frac{1}{4\,\mu\, B\,\lambda_i} \text{ for } i = 0, 1, 2\ldots\ldots N-1 \qquad \ldots (17)$$

Where $\lambda\, i\, S$ are eigen values of R.The time constants are in the "unit of iteration (block) interval" Averaging the instantaneous gradient vectors as was done in BLMS algorithm results in gradient vector with a lower variance, as compared with that in the conventional LMS algorithm. For block length L, comparable or less than the filter length, N, the mis adjustment, MB, of the BLMS can be approximated by the following expression:

$$M\,B = \{\,\mu\, B/L\; tr\,[R]\,\}. \qquad \ldots (18)$$

If we let M B » M, where M denotes the misadjustment of the conventional LMS algorithm, we obtain

$$T_{Bi} = 1/4L\,\mu\,\lambda\, i \qquad \ldots (19)$$

Block Interval and

$$T_{Bi} = 1/4\,\mu\,\lambda\, i \qquad \ldots (20)$$

Sample Interval .Thus, the convergence behavior of BLMS and the conventional LMS algorithm are the same.

IV. SIMULINK MODEL

The model which is built for this paper is used to generate a input of 8KHz audio speech signal and noise is added from the acoustic environment ie., the noise can be for mild, moderate and severe and it shown in the port 1 (Exterior Mic) and mixed output is shown in port 2 (pilot mic) . The extracted original speech signal is shown in time scope. The adaptive BLMS algorithm is used to reduce the error and also this error will be used to update the next coefficient. The Simulink model is shown in figure (4&5)

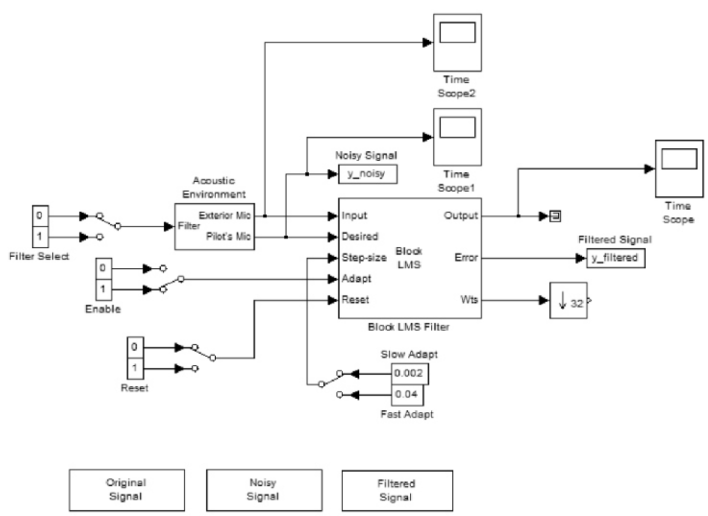


Fig. 4. Block LMS Simulink model

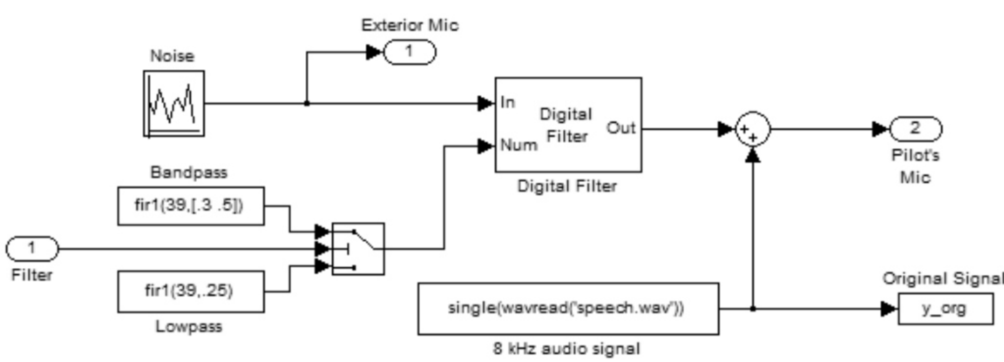


Fig. 5. Internal Acoustic Environmental Simulink model

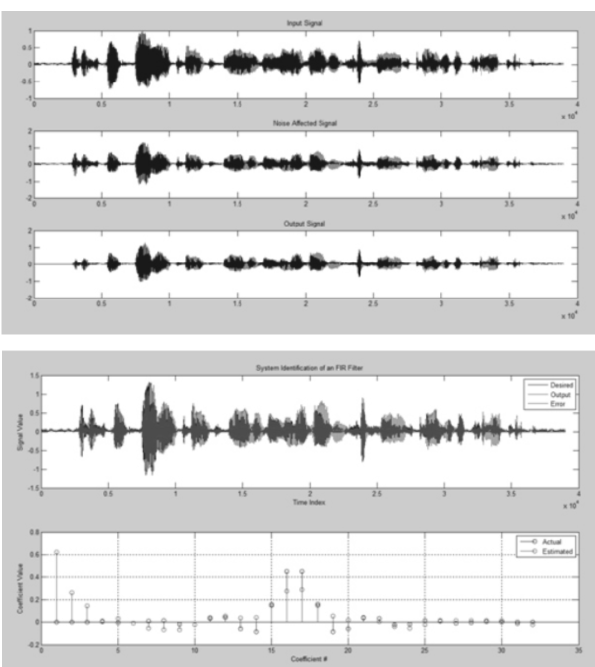
V. SIMULATION AND EXPERIMENTAL RESULTS

For an input of speech signal three different variation of noise, ie, mild, moderate and severe noise is added with an input and the corrupted noise output is traced for three different noises and the original speech signal is also adapted and traced by using Simulink modeling. To perform the system improvement signal to noise ratio is also determined and which shows the system effectiveness. The first figure will for Mild Noise which comprises the input speech signal, followed by mild noise and original signal. The second figure will for Mild Noise which comprises the input speech signal, followed by mild noise and original signal. The Third figure will for Mild Noise which comprises the input speech signal, followed by mild noise and original signal. Finally each experiment has the filter performance and their spectrum analysis. Table 1 refers the signal to noise ratio of the speech signal using BLMS algorithm

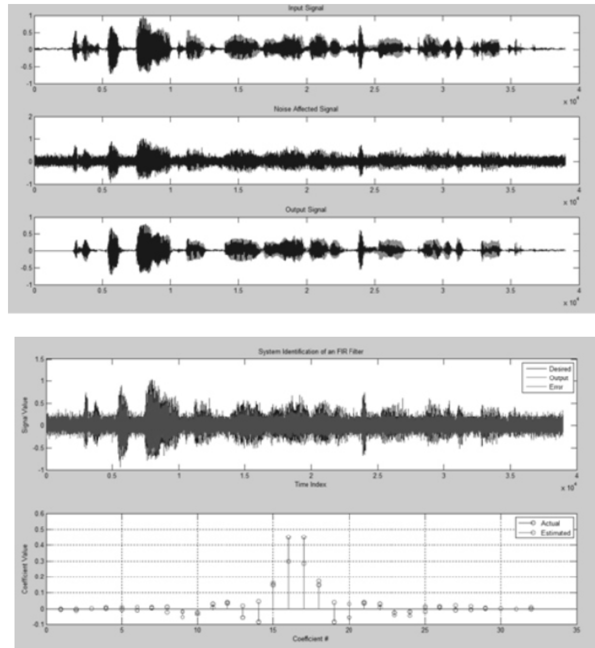
Table 1. Refers the signal to noise ratio

S.No	Different types of Noise	SNR in dB
1	Mild Noise	15.5
2.	Moderate Noise	15
3.	Severe Noise	9

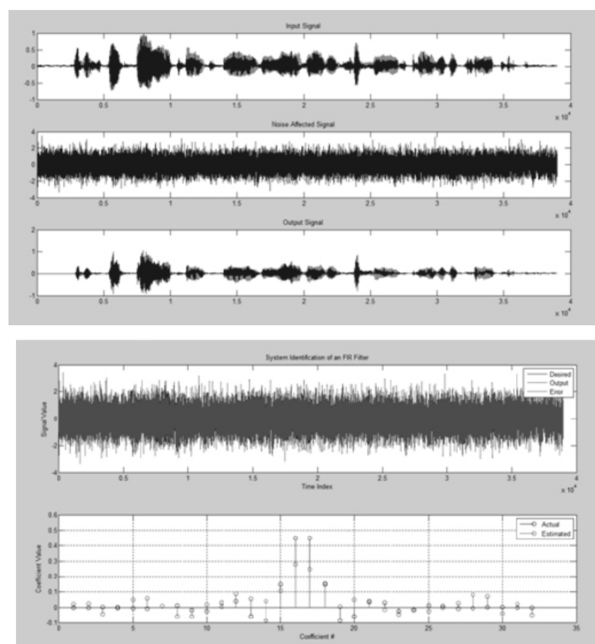
First experiment: Mild Noise with speech signal



Second experiment: Moderate Noise with speech signal



Third experiment: Severe Noise with speech signal



VI. CONCLUSION

The three different variations of noises with input were used to adjudge about the system performance in software and hardware .Besides mild and moderate noises were also used interchangeably to test the system on hardware during its running time. The background noises for speech signal were eliminated adequately with reasonable rate for all the tested noises. In mild and moderate noises the system showed SNR improvement up to 15dB and the severe noise test 9dB of SNR improvement achieved. The major advantage of the proposed system is its ease of implementation and fast convergence. The proposed algorithm is applied to noise canceling problem of long distance communication channel. The simulation results showed that the proposed model is effectiveness.

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