# Adaptive Noise Cancellation using Modified Normalized Least Mean Square Algorithm

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***Abstract-*** *This paper presents an efficient design of Adaptive filters which uses enhanced NLMS algorithm for eliminating noise added by mean of various communication media or any other noise sources. By using the appropriate weights, Adaptive filter estimates and remove the estimated noise signal from the available information. LMS and Normalized LMS are two most efficient algorithm for noise cancelation. Normalized form of LMS algorithm is considered more effective as it normalizes with the power of the input. The concept of memory preallocation for variables has been introduced to enhance the computational speed of filtering system. The proposed algorithm is successfully implemented using M Code and performance measures like SNR and MSE are calculated for varying step size and filter order. The SNR value ranges from 8.26 dB to 7.22 dB and MSE ranges from 0.087 to 0.1173.*

***Keywords:*** *Adaptive Filters, LMS Algorithm, MSE, NLMS Algorithm, Noise Cancelation, SNR.*

## I. INTRODUCTION

Most of the portable electronic devices such as cellular phones, personal digital assistants, and hearing aids require digital signal processing for high performance [1]. In the era of communication systems the processing of signal is of main concern as the transmission signals suffered from various interferences and noise because of their sinusoidal nature. Thus to improve the quality of communication, signal processing is required for the noise cancellation from the signal by using filters [2]. Adaptive filters are widely used in several signal processing applications for example system identification, equalization, interference and echo cancellation. These filters are usually made of finite-impulse-response (FIR) filters whose co efficient are updated using minimization criterion. The output of FIR filter is weighted sum of current and previous input samples [3].

Filter coefficients keep on changing with time in adaptive filtering to adapt to changing signal characteristics. After selecting a particular adaptive filter structure, the number of parameters and type of parameters for adjusting can be specified. DSPs have made many advancements over the past three decades in speed improvement, area and power consumption. Because of this reason the adaptive filtering algorithms are rapidly becoming more practical and necessary for the forthcoming of communications [4].The method of active noise control is applied for reducing the troubling noise and is most important communication systems. Over the years numerous techniques for processing of signal have been introduced to cancel out the uncorrelated noise. Because of the ability of handling the spatially varying noise present in desired signals, adaptive algorithms for signal processing have gained attractiveness in recent years [5].

Noise Cancellation is a variation of optimal filtering that includes creating an estimate of the noise by filtering the reference input and then deducting this noise estimate from the primary input comprising both signal and noise. In the process of adaptive noise cancellation the mixture of signal and noise is delivered through the filter, produces the output signal which is noise free without the variations in its parameters. Adaptive filters make use of negative feedback. When the input signal along with noise signal is given to the filter, a negative feedback is applied to the filter by adjusting the values of coefficients; this feedback is proportional to the noise signal present in the input. In this way the noise cancellation is done [6]. A prior knowledge of noise and signal characteristics is not required for an adaptive process. Although both FIR and IIR filters can be used but FIR filter is the mostly used and is more practical for adaptive filtering. The reason being that FIR filter provides the adjustable zeros because of this, it is free from the problems of stability which are present in adaptive IIR filters in which both zeros and poles are adjustable [7].

As shown in Figure 1, there are two inputs- one is signal input (primary signal) and other is reference signal. The primary input gets a signal X(n) from the input source which is corrupted by the presence of noise N(n) uncorrelated with the signal. The reference signal receives a noise Nr(n) which is uncorrelated with the input signal but in some way correlated with the noise signal N(n). The reference noise passes through an adaptive filter to produce an output F(n), which is close estimate of primary input noise N(n). This estimated noise is deducted from the corrupted signal to produce the output Y(n) of ANC system, which gives an estimate of the signal X(n).

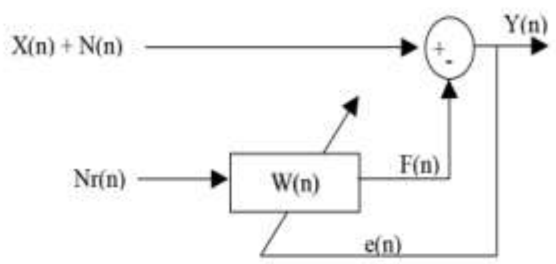


Fig. 1. Adaptive Noise Cancellation (ANC)

The main objective of a noise cancellation system is to generate an output that is a best fit in the least squares sense to the signal x, which can be done by applying feedback to the adaptive filter then filter coefficients are being adjusted using the LMS based adaptive algorithm to diminish the output noise power.LMS mainly modify the filter Weight coefficients by modifying the different variables of filter thus attain the minimum difference between desired signal and obtained output of filter [8]. Until the desired performance is achieved, the number of iterations are needed to be performed by changing the various parameters again and again. But there is a problem associated with "pure" LMS algorithm, that for the input scaling it is very sensitive, thus it is difficult to select the step size that warranties stability of LMS algorithm.

Thus Normalized least mean squares filter (NLMS), which is modified version of Least Mean Square algorithm and remove the instability problem of LMS by applying normalization to the input power [9].Normalized Least Mean Square (LMS) Algorithm is used in this paper. The filter length and step size of an adaptive filter along with the algorithm affect the convergence speed. An appropriate step size and filter length must be chosen to ensure that the convergence speed of the adaptive filter satisfies the application requirements. After getting the output, SNR and MSE can be calculated.

## II. ADAPTIVE FILTER ALGORITHMS

Adaptive filtering involves the changing of filter parameters over time, to adapt to changing signal characteristics. When the signal into the filter continues, the adaptive filter coefficients modify themselves to achieve the desired result, such as identifying an unknown filter or canceling noise in the input signal. An adaptive algorithm provides the impulse response of filter and weights adjustment by using adaptive control mechanism, noise reduction can be done with minute risk of distorting the signal. The higher level noise rejection is possible by performing adaptive noise cancelation, otherwise this level of rejection of noise is difficult to achieve by using direct filtering methods. It depends on type of application that what type of error signal to be used[10]. Some of the criteria which can be used are minimization of the mean square error and the temporal average of the least squares error. Various algorithms can be employed for the purpose of minimization criteria such as, least mean square algorithm, NLMS algorithm, the recursive least square algorithm etc. In this paper to understand the concept of cancellation of adaptive noise, the NLMS algorithm is used which further uses the minimum mean-square error criterion.

### *A. Least Mean Square Algorithm*

LMS algorithm was first proposed by Widrow and Holf in 1960. It is the most widely used adaptive filter algorithm. In this algorithm weight coefficients are adjusted from sample to sample in such a way to minimize the mean square error (MSE). This algorithm is considered very reliable for noise cancellation in various communication system applications.
LMS algorithm is best suited for the problems of linear estimation to find out the excess mean square error [11]. Aim of this algorithm is reduction of the error signal which is calculated in between output of filter and the desired signal by estimation process. The complete process can be fragmented in to two parts.

Filtration part: Under this part the output signal is computed for the adaptive filter, pertaining to the applied input signal to the filter and thus error signal is computed by taking the difference between actual output signal and desired output signal.

Adaptation part: In this part the adjustment of the filter weights($Wt_{(n+1)}$) is being done with the help of estimated error signal.

Following equation represents the weight updating part:

$$Wt_{(n+1)} = Wt_{(n)} + \mu S_{(n)} X_{(n)} \qquad (1)$$

Where $Wt_{(n)}$signifies co efficient, $Wt_{(n+1)}$signifies updated coefficient vector , µis parameter used for step size , $X_{(n)}$is Input signal with time delay and $S_{(n)}$is the value of estimated error signal. All the parameters in LMS algorithm are affected by step size. Filter circuit can become unstable if step size value is very small because more time will be required for the convergence of filter. The filter output is given by the equation:

$$X_{(n)} = Wt^{T}_{(n)} X_{(n)} \qquad (2)$$

As shown in equation 2, to obtain the filter output, the transpose of previously calculated weight is multiplied with the input signal. The Adaption equation of Weight Vector is given as:

$$Wt_{(n+1)} = Wt_{(n)} + 2\mu S_{(n)} X_{(n)} \qquad (3)$$

### *B. Normalized Least Mean Square Algorithm*

Because of Least Mean Square Algorithm has some limitations such as instability when the power of input signal changes, as the step size value varies resulting in change of the rate of convergence, thus Normalized Least Mean Square Algorithm is evolved from Least Mean Square algorithm to overcome these limitations. The most significant parameter of LMS responsible for the stability of the circuit is step size (μ). But in case of NLMS minimal disturbance is imposed during the change of iteration for which the weight vector W(n) should be changed so as to get minimum error in the obtained output signal [12]. The suitable solution for above problem is to normalizing the power of input. For which, the value of varying step size can be given as:

$$\mu_{(n)} = \frac{\mu}{c+||x(n)^2||} \qquad (4)$$

As per the above equation variations in step size for Normalized Least Mean Square depends on Adaption constant μ, whose range should lie in between 0 <μ< 2 and c specifies the constant of Normalization which must be less than 1.This type of variation in Step size provides improved rate of convergence. So the equation of Weights for NLMS can be written as:

$$W_{(n+1)} = W_{(n)} + \frac{\mu}{c+||X(n)^2||} S_{(n)}\, X_{(n)} \qquad (5)$$

These modified equations of step size and weight vector are used to attain high rate of convergence and provides the flexibility of NLMS algorithm.

## III. PROPOSED NLMS ALGORITHM

In this section we have presented our proposed methodology for enhance NLMS algorithm. In our proposed model we have replaced traditional NLMS algorithm with modified NLMS algorithm for adaptive filtering which uses memory preallocation for variables to optimize overall run time for the whole process. Figure 2 represents the proposed model for NLMS algorithm.

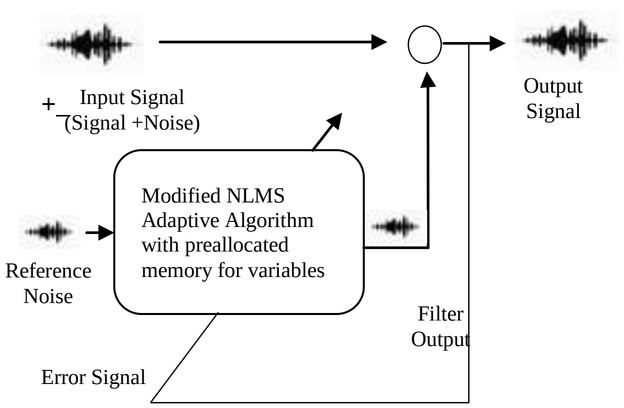


Fig. 2. Modified NLMS Adaptive Filtering

Following notations have been used to explain pseudo code of proposed adaptive algorithm.

Z(m,n) : Convolution matrix for input signal X(n)
D(n) : Desired Signal
$C_n$ : Normalization Constant
Err(n) : Error Signal for sample n
W(m,n) : Weight Coefficient Matrix
O(n) : Filter Output
μ : Step Size
h : Impulse Response

The following steps are followed during error calculation and weight updation process for the modified algorithm.

Step 1: Determination of Convolution Matrix of Input Signal
X(n) and impulse response.

Z(m,n) = conv(X(n),h)

Step 2: Calculation of Normalization Constant using convolution matrix.

$C_n$=Z(n,:)*Z(n,:)' + 0.0001

Step 3: Error calculation using previous weight coefficient.

Err(n) = D(n) - W(n-1,:)*Z(n,:)'

Step 4: Weight matrix updation.

W(n,:) = W(n-1,:) + μ/$C_n$*E(n)*conj(Z(n,:))

Step 5: Filter Output Calculation

O(:,n) = W(n,:)*Z(n,:)'

The filter output O(n) is subtracted from the input signal X(n) for final output of the adaptive noise cancellation system. The above algorithm is implemented using M codes to find out various results corresponding to different systems in terms of efficiency and reliability.

## IV.RESULTS AND DISCUSSON

The proposed algorithm for NLMS adaptive filtering is implemented and simulated using Matlab code. Filter parameters like order and step size are changed time to time to find out optimum values of these parameters for different communication systems.For the simulation purpose we have taken input signal X(n) with sampling frequency $f_s$=48000 Hz for 2 sec.

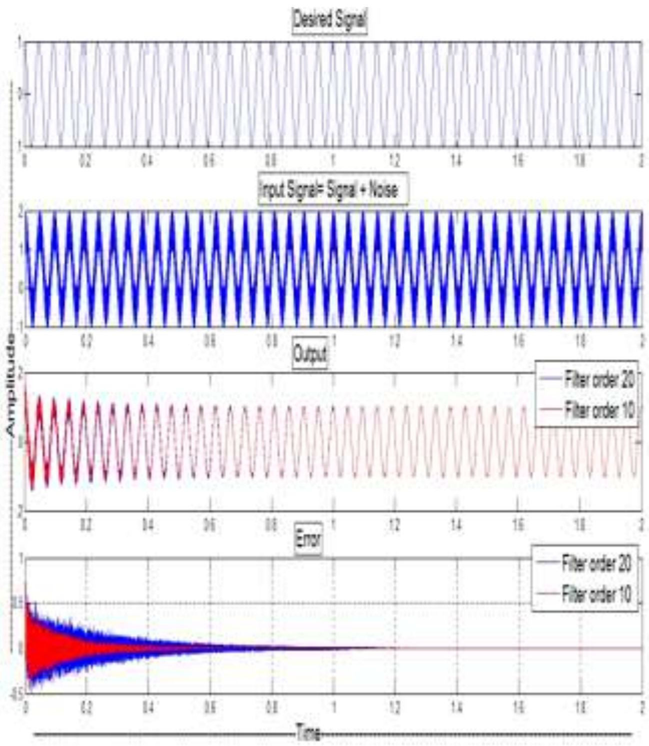


Fig.3.Output and Error signals.

The signal X(n) contains a random noise within it. Figure 3, shows the different wave forms of the proposed system observed for filter order 10 and 20. The proposed system is also exposed to series of varying step sizes (μ) ranging .0001<μ<.01 for which $SNR_{db}$(Signal to Noise Ratio)and MSE (Mean Square Error) values are calculated for filter order 10 and 20. Table 1, contains some of these values with which we can observe the behavior of the proposed system in different environment.

Table 1: MSE & SNR Analysis.

| Sr. No. | Step Size | Filter Order 10 | | Filter Order 20 | |
|---|---|---|---|---|---|
| | | MSE | $SNR_{db}$ | MSE | $SNR_{db}$ |
| 1. | 0.0001 | 0.0888 | 8.2187 | 0.0877 | 8.2643 |
| 2. | 0.0003 | 0.0917 | 8.1010 | 0.0877 | 8.2633 |
| 3. | 0.0005 | 0.0983 | 7.8512 | 0.0913 | 8.1203 |
| 4. | 0.002 | 0.1072 | 7.5433 | 0.1006 | 7.7702 |
| 5. | 0.003 | 0.1107 | 7.4296 | 0.1066 | 7.5644 |
| 6. | 0.004 | 0.1125 | 7.3737 | 0.1102 | 7.4450 |
| 7. | 0.005 | 0.1135 | 7.3403 | 0.1126 | 7.3706 |
| 8. | 0.006 | 0.1142 | 7.3180 | 0.1141 | 7.3207 |
| 9. | 0.007 | 0.1147 | 7.3020 | 0.1153 | 7.2852 |
| 10. | 0.008 | 0.1151 | 7.2899 | 0.1161 | 7.2587 |
| 11. | 0.009 | 0.1154 | 7.2805 | 0.1168 | 7.2381 |
| 12. | 0.01 | 0.1156 | 7.2729 | 0.1173 | 7.2218 |

From table 1 it can be analyzed that as the step size is increased gradually, the SNR values decreases and the Mean Square error increases between desired output and actual output.

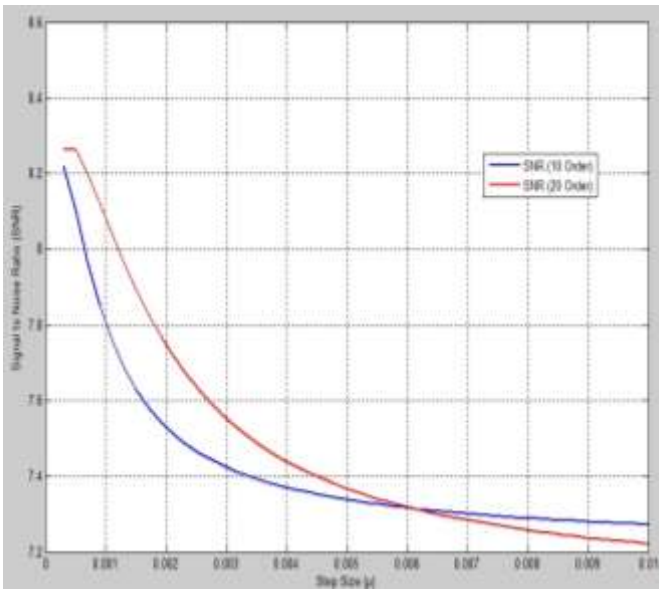

Fig.4.SNR comparison.

Figure 4, represents SNR values calculated for various step sizes for filter order 10 and 20. On observing above figure the behavior of proposed system can be analyzed under different conditions. Mean Square Error has also been calculated for same conditions and represented in Figure 5. Filter order is a parameter which determine the area, cost and complexity of a filter design. So on the basis of above observations, we have to select best possible combination of filter order, step size and normalization constant for different communication systems.

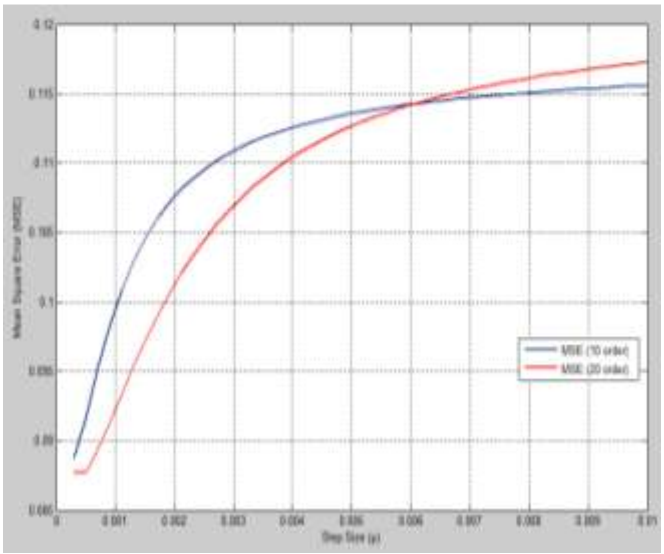

Fig.5.MSE comparison.

## V.CONCLUSION

Enhanced NLMS Adaptive Algorithm is implemented with reallocated memory for variables carrying different signals. Simulation and testing is done in MATLAB with different filter parameters. Performance measures like SNR and MSE for the proposed system are being calculated. We are getting SNR values ranging between 8.26 dBto 7.22 dB and MSE ranging from 0.087 to 0.1173, which indicates an efficient adaptive algorithm.

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