Analysis of Noise Signal Cancellation using Adaptive Algorithms

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Abstract – Noise is an indispensable part in signal processing that we encounter every day. The study of reducing noise arises from the need to achieve stronger signal to noise ratios. It is any unwanted disturbance that hampers the desired response while keeping the source sound. The different sources may include speech, music played through a device such as a mobile, IPod, computer, or no sound at all. Active noise cancellation involves creating a supplementary signal that DE constructively interferes with the output ambient noise. The cancellation of noise can be efficiently accomplished by using adaptive algorithms. An adaptive filter is one that self-adjusts the coefficients of transfer function according to an algorithm driven by an error signal. The adaptive filter uses feedback in the form of an error signal to define its transfer function to match changing parameters. The adaptive filtering techniques can be used for a wide range of applications, including echo cancellation, adaptive channel equalization, adaptive line enhancer, and adaptive beam forming. In last few years, a lot of algorithms have been developed for eradicating the distortion from the signals. This paper presents analysis of two algorithms namely, Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and gives comparative study on various governing factors such as stability, computational complexity, filter order, robustness and rate of convergence. It further represents the effect of error with alteration in amplitude of noise signal fixating reference signal and desired signal. The algorithms are developed in MATLAB

Keywords– Anti noise, Adaptive filter, LMS, NLMS, Rate of convergence, Noise cancellation, Filter

I. INTRODUCTION

Noise is any unpleasant, objectionable, unexpected, undesired distortion in sound. In electronics, it may be defined as the random fluctuation in an electrical signal. Noise is all around us, right from radios and television, to lawn movers and washing machine. Normally, the sounds that we hear do not affect hearing, but too loud sounds may be harmful in the long run. Noise free signals give better signal to noise ratios as the absence of noise strengthens the signal to noise ratio. [1]

The technique employed to achieve this is active noise cancellation. The best approach to cancelling noise would be to take the noise signal, invert it, and add the input and inverted signals such that they add deconstructive. It is a highly recommended method because can block selectively and improves noise-control. It offers potential benefits such as size, cost, volume and effective attenuation of low frequency noise. The components of noise signal such as frequency, amplitude and phase are non-stationary and time varying; hence the use of adaptive filter helps us to deal effectively with the variations. Noise Cancellation utilizes the principle of destructive interference. When two sinusoidal waves superimpose in a way such that the amplitude, frequency and phase difference of the two waves are the governing factors of the resulting waveform and, if the two waves, the original and its inverse happen to meet at a joint, at the same instant, total cancellation occurs.[2]-[4]

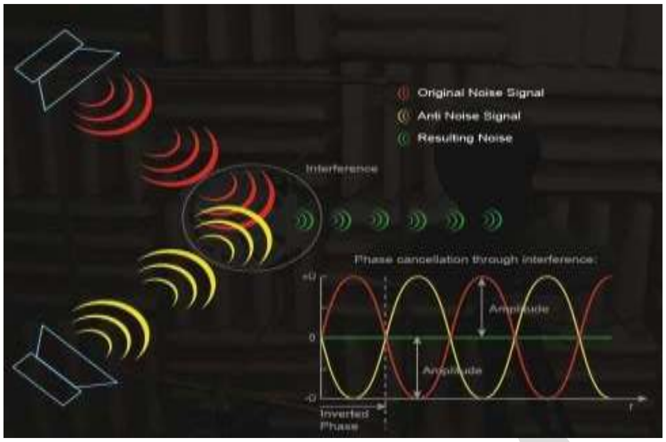


Fig. 1– Noise Cancellation through Destructive Interference

Noise elimination from an input signal could produce disastrous results, which is marked by an increase in the average power of the output noise. However when an adaptive process controls filtration and reduction, it is possible to achieve a superior system performance compared to direct filtering of the input signal.[5]

II. METHODS-

Methods used for executing the result of the research work is described below-

A. ADAPTIVE FILTER

An adaptive filter has the property of exhibiting self-modification in its frequency response with respect to time, allowing the filter to adapt the response to the input signal characteristics change enhancing performance and construction flexibility.

An Adaptive Filter may be defined by following four aspects:

1. The signal being processed the by the filter.
2. The structure that defines how the output signal of the filter is computed from its input signal.
3. The parameters within this structure that can be iteratively changed to alter the filters input-output relationship.
4. The adaptive algorithm that describes how the parameters are adjusted from one time instant to the next

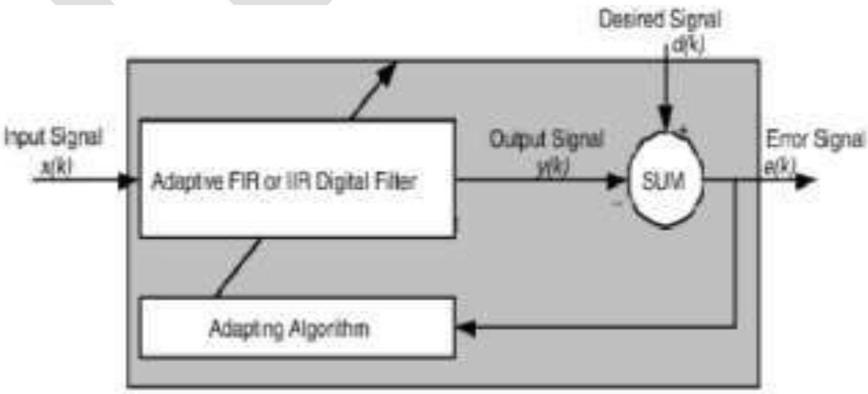


Fig.2. Block Diagram of General Adaptive Filter

The parameters of an adaptive filter are updated in each iterative step and hence it becomes data dependent. The adaptive filter is used when the parameters are not fixed or the specifications are unknown. Therefore, this implies the nonlinearity feature of the filter, as it fails to follow the principle of superposition and homogeneity. An adaptive filter is linear if the input output relation obeys the above principles and the filter parameters are fixed. As the parameters change continuously in order to meet a performance requirement, the adaptive filters are time varying in nature. In this sense, we can interpret an adaptive filter as a filter that performs the approximation step on-line. The performance criterion requires the existence of a reference signal that is usually hidden in the approximation step of fixed-filter design.

These filters are recommended because of their ease of stability and simplicity in implementation without any adjustment. Adaptive filtering, which concerns the choice of structures and algorithms for a filter that has its parameters (or coefficients) adapted, in order to improve a prescribed performance criterion. [1]-[7]

The adaptive filter adjusts coefficients to minimise following

Cost function J (n) = E [e^2(n)]

Where E[e^2(n)] is the expectation of e^2(n), and e^2(n) is the square of the error signal at time n.

The two algorithms used in the FIR Adaptive filter to control the adjustment of filter coefficients are:

1. Least Mean Square Algorithm (LMS)
2. Normalized Least Mean Square Algorithm (NLMS)

B. LEAST MEAN SQUARE:

Least Mean Squares (LMS) algorithm are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relates to producing the Least Mean Square of the error signal, that is, the deviation between the original signal and the desired signal. One of the feature of LMS filter algorithm is its simplicity. Further it neither requires measurement or knowledge of correlation function nor does it require Matrix Inversion (case of more than one μ = step size, a scaling factor which controls the incremental change applied to the tap weight vector of the filter from one iteration to next. To ensure stability, μ should satisfy the following condition.

$$0 < \mu < \frac{2}{LS_{max}}$$

Where,

L = filter length $S_{max=}$ is the maximum value of the power spectral density of the tap input x(n).

The goal of the LMS method is to find the filter coefficients required to reduce the mean square error of the error signal. The error signal, which, is the difference present between the desired d(n) and output y(n). The filter will only conform to the error at the current time. In this algorithm, initially it assumes small weights (mostly zero), and at each iterative step, by finding the gradient of the mean square error, the filter parameters are updated. That is, if the MSE-gradient is positive, it implies that the error would keep increasing positively. If the same parameters are used for further iterations, which implies that we have to reduce the weights. Similarly, if the gradient is negative, we have to increase the weights. The mean square errors is a quadratic function which means it has only one extrema that minimises the mean square error, that is the optimal weight. The LMS thus, approaches towards this optimal weight by ascending/descending down the curve between the mean square error and filter weight. [3]-[7]

C. NORMALIZED LEAST MEAN SQUARE

The main drawback of the LMS algorithm is that it is sensitive the scaling factor, μ of the input signal. When it is large, the filter suffers from gradient noise amplification problem. This makes it difficult in choosing the appropriate step size for the filter to ensure the stability. The Normalized Least Mean Square algorithm is an improvement over the conventional LMS and solves this problem by normalising the power of input. [8]

The step size of NLMS filter is given as

$$\mu = \frac{\alpha}{||x(n)||^2}$$

where,α = Adaptation constant which is dimensionless and optimizes rate of convergence by satisfying the condition, $0 < \alpha < 2$

The weights of the filter are updated as the following step:

$$w(n+1) = w(n) + \mu . e^T(n) . x(n)$$

III. FACTORS DETERMINING THE BEHAVIOUR OF ALGORITHM

A. **Stability***:*

The term stability refers to the effects of finite precision on the algorithm that is used to find the solution of some problem of interest.

B. **Computational Requirement***:*

Basic requirement during computation are: Number of operations required to complete the whole algorithm in one iteration; memory required for storing the data and algorithm program and also investment required during the programming of the algorithm on computer.

C. **Rate of Convergence***:*

It explains the number of iteration required for the algorithm with respect to stationary inputs in order to converge close enough to the optimal wiener solution in the mean square error sense. Due to fast rate of convergence, which permit the algorithm to adapt rapidly to a stationary environment of unknown statistics.

In case of NLMS, when the input vector x(n) and x(n-1) are orthogonal to each other, i.e., the angle between them ±90°, then the rate of convergence is fastest. When the input vector x(n) and x(n-1) are in the same direction or in opposite direction such that the angle between them is ±180°, then the rate of convergence is slowest.

D. **Robustness:**

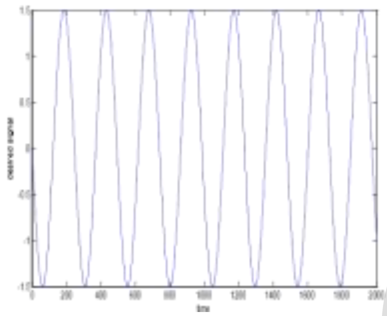
The ability of the system to cope up with errors is defined as robustness of the system. For a robust adaptive filter, small fluctuations results in errors. These deviations can be present as a result of various, internal or external factors of the filter. [9]-[12]

IV. RESULTS:

A. ANALYSIS I

These algorithms (LMS & NLMS) were formulated in MATLAB and following results were generated. The first figure shows the desired signal. The next figure represents the input signal which is composed of sinusoidal signal and

the noise that is incorporated, has random nature.



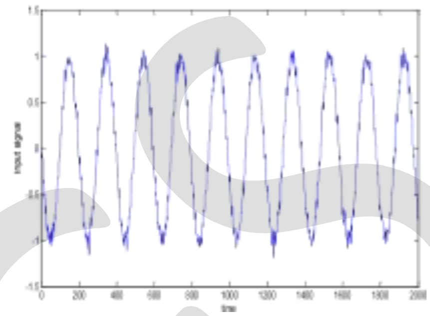


Fig. 3 – Desired Signal Fig. 4 – Input Signal

a. Results of LMS Algorithm

Figure 5 represents the filter output. The next figure gives the error generated. The filter order is 251 and the step size is 0.005.

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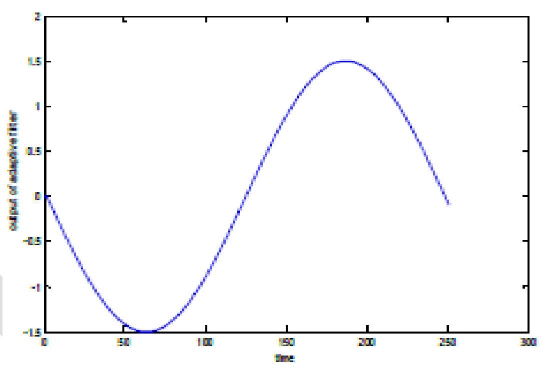


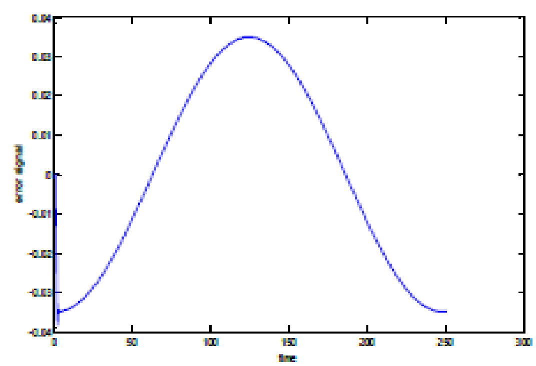
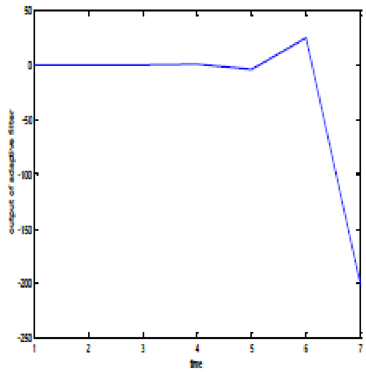


Fig. 5 – Adaptive Filter output Fig. 6 – Error signal

b. Results of NLMS Algorithm

Figure 7 represents the adaptive filter output. The next figure shows the error signal of NLMS algorithm. The filter order is 7 and α is fixed at 0.5.



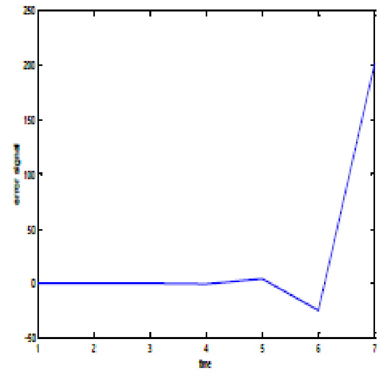


Fig. 7 - Adaptive filter outputFig. 8 – Error signal

Table.1 Comparison between LMS and NLMS on various factors

Serial No.	Factors	LMS	NLMS
1	Stability	Highly stable	Moderate stable
2	Computational Complexity	2N+1	3N+1
3	Rate of Convergence	Low and has slow implementation	High and has faster implementation
4	Robustness	Less robust	More robust

B. ANALYSIS II

In this section, we have observed the fluctuation in the error signal with the variation of amplitude of noise signal having constant reference signal and fixed desired signal. The input signal changes as it is the summation of reference and varied noise signal.

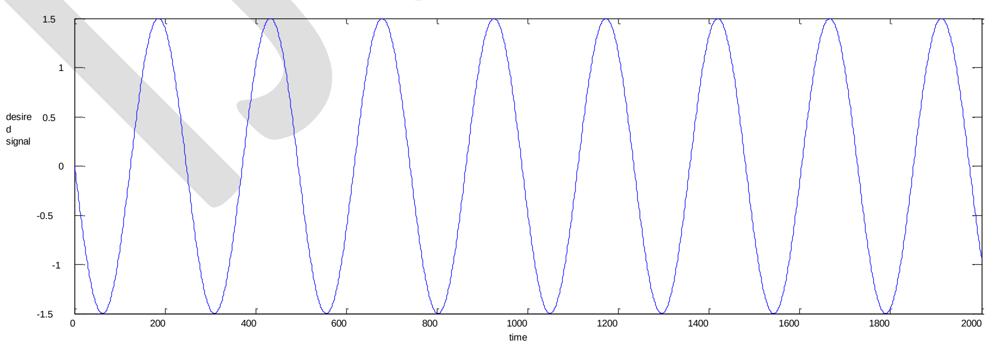
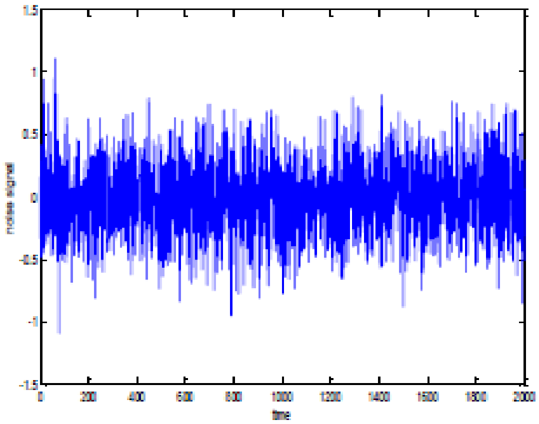


Fig. 9 – Desired Signal

This is desired signal of amplitude 1.5.

We have fixated the order of the filter at 251.

1. When the scaling factor of random noise is 0.3, the noise signal, input signal and error signal are generated as follows.



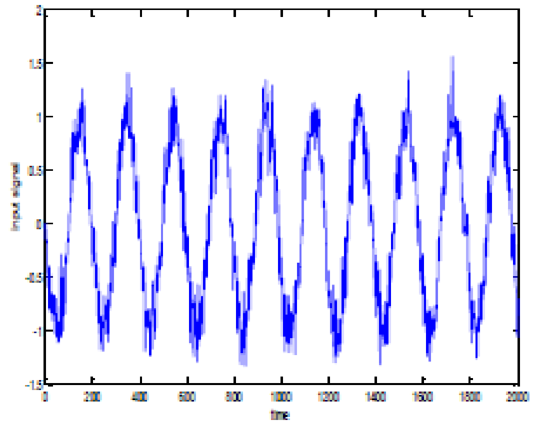


Fig. 10 – Noise signal

Fig. 11 – Input signal

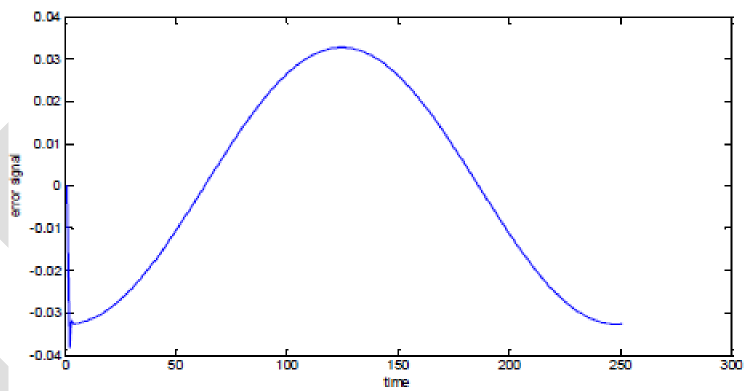
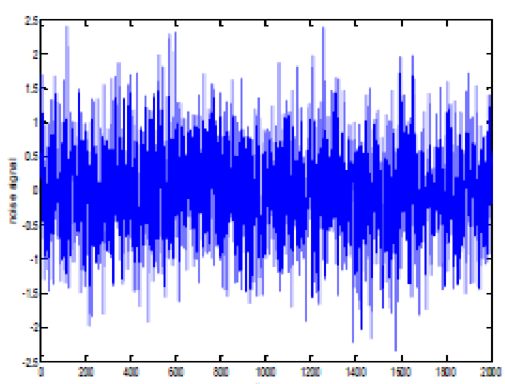


Fig.12-Error Signal

2. When the scaling factor of random noise is 0.7, the noisesignal, input signal and error signal are produced as below



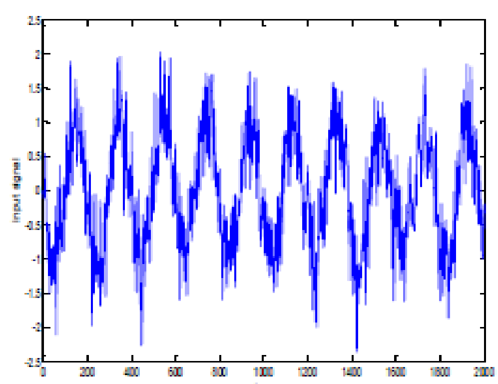


Fig. 13 – Noise Signal Fig. 14 – Input Signal

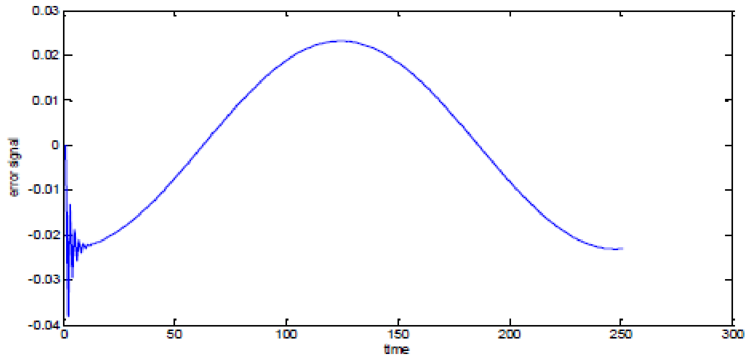


Fig. 15 – Error Signal

2. When the scaling factor of random noise is 1 and 1.1 the error signals are given below in figure 16 and 17 respectively.

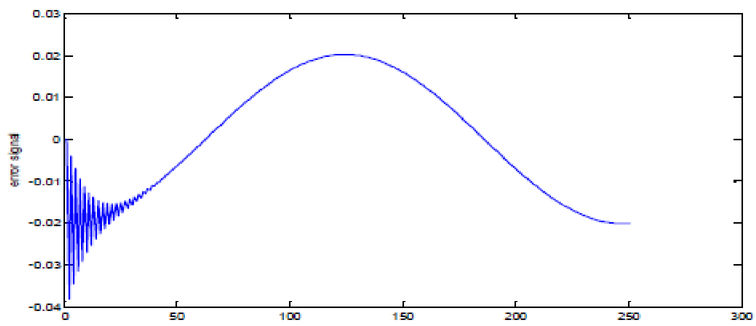


Fig.16- Error signal for scalinging factor 1

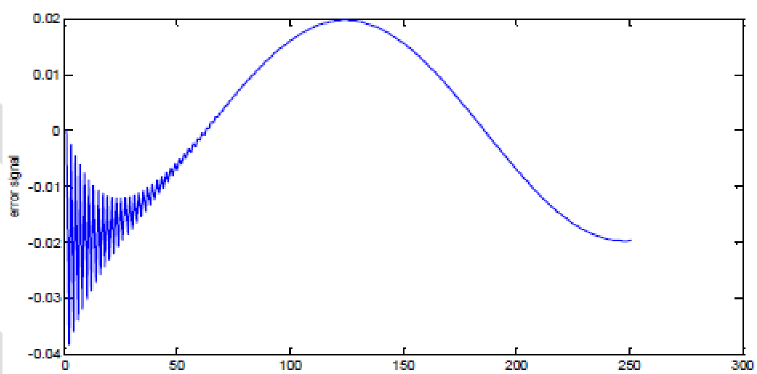


Fig.17-Error signal for scaling factor 1.1

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VI. CONCLUSION

In first analysis, we see that the LMS is preferred because of its stability, simplicity and additional adjustment is required, but has a slow rate of convergence, whereas NLMS though less stable, offers better robustness, converges faster than conventional LMS and has good performance because of its properties.

In second analysis, we have observed that with increase in the scaling factor of random noise, the fluctuation in the error signal increases and it takes little longer to optimize the error in accordance to the desired signal. As verified above, when the scaling factor is taken as 0.3, 0.7, 1 and 1.1, the adaptive filter output, initially deviates from the desired signal and gradually the filter output approaches the desired signal.

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