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Performance Measures of NLMS Adaptive Filter for Noise Estimation and Cancellation

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Abstract *— Interference cancellation is a technique of utmost importance in the field of signal processing. It is especially essential for speech signal transmission and processing due to the ever-growing application of telephone and cellular communication. Interference cancellation can be achieved by an adaptive filter, a filter which self-adjusts its transfer function according to an optimizing algorithm. The most popular of such adaptive algorithms is the Least Mean Square (LMS) algorithm. The normalized Least Mean Square (NLMS) algorithm can be considered as a special case of the LMS recursion which takes into account the variation in the signal level at the filter output. The performance of these adaptive algorithms is dependent on their filter length and the selected convergence parameter μ. NLMS algorithm is a potentially faster converging algorithm compared to the LMS algorithm. Faster convergence, however, comes at a price of greater residual error. More recent studies that try to relax this trade-off have been directed towards adjustable step-size variations of the two algorithms. Rather than focusing on the convergence behavior, this paper focuses on NLMS Adaptive filter used to minimize the noise signal. Hence, the performance of the NLMS algorithms in interference cancellation has been presented in terms of the Simulink model of the input and output signals. The effects of the filter length and step size parameters have been analyzed to reveal the behavior of the algorithms.*

Keywords*- Matlab/Simulink, NLMS, Noise Estimation, Noise Cancellation*

I. INTRODUCTION

The purpose of filtering, which is a signal processing operation, is to manipulate the information present in the signal. Filter is a device which extracts desired information from the input signal by mapping input signal to another output signal. The digital format of discrete time signal can be processed by digital filter. With these filters desired spectral characteristics of signal can be achieved by which unwanted signal can be rejected, like noise or interference and bit rate can be reduced in transmission. For time-invariant filters the internal parameters and the structure of the filter are fixed and if the filter is linear the output signal is a linear function of the input signal. The problem which is not known in advance can be tackled by making the filter adaptive, changing the parameters of filter according to some algorithm. The characteristics of the signal, or the unwanted signal, or systems influence on the signal that one would like to compensate. The Adaptive filters are capable of adjusting to unknown environment and even track signal or system characteristics varies in time. Three steps are in designing time invariant linear filter, namely specification are approximated using rational transfer function, choice of defining the algorithm and choice of implementation form for the algorithm. An adaptive filter is required when either the fixed specifications are unknown or the specifications cannot be satisfied by time-invariant filters. In actual fact an adaptive filter is a nonlinear filter because its characteristics dependents on the input signal, consequently the homogeneity and additive conditions are not satisfied. In general, adaptive filters are one that varies through time because the characteristics of its inputs may be varying. That is why it separates itself from classical digital signal processing the digital system itself changes through time. When there is a need to process signal in an environment of unknown statistics adaptive filters will not perform better than fixed filter. This paper describes adaptive filtering and its simulation in MATLAB with Normalized LMS algorithm for noise cancellation in speech signals.

1.1 NLMS Algorithm

The LMS algorithm which uses an instantaneous estimate of the gradient vector of a cost function is an approximation of the steepest descent algorithm. Based on sample values of the tap-input vector and an error signal the gradient is estimated. The algorithm iterates each coefficient in the filter, moving it in the direction of the approximated gradient. For the LMS algorithm it is necessary to have a reference signal d[n] representing the desired filter output. The difference between the reference signal and the actual output of the transversal filter is the error signal which is given in the equation

$$e(n) = d(n) - c^h(n)x(n)$$

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One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. Even if we assume the only signal to be input to the adaptive system is signal, there are still many factors such as signal input power and amplitude which will affect its performance[1- 3].The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by calculating maximum step size value. Step size value is calculated by using the following formula.

Step size=1/dot product (input vector, input vector). This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x(n). This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix, R [4-6]

$$\mathrm{tr(R)} = \sum_{t=0}^{n-1} \mathrm{E}[x^2(\mathrm{n-i})]$$

$$= E\sum_{t=0}^{n-1} [x^2(\mathrm{n-i})]$$

The recursion formula for the NLMS algorithm is stated in equation

$$\mathrm{w(n+1)} = w(n) + \frac{1}{x^T n x(n)} \mathrm{e(n)x(n)}$$

The figure 1 shows the block diagram of the proposed adaptive filter for error estimation and a interference cancellation

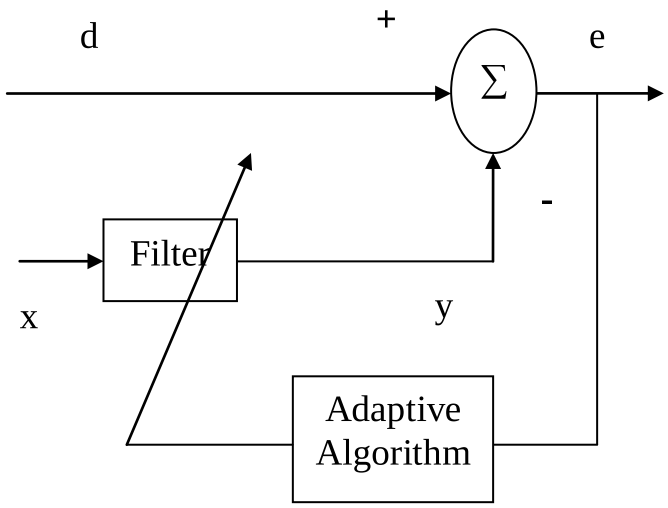

Fig.1 Adaptive filter structure

II. IMPLEMENTATION

The Fig.2 shows the simulink model for NLMS adaptive filter. First the input signal sine wave is applied as an input to add block. Another input to add block as from uniform noise generator, these two signals are combined together to form the noisy signal. These signalis appled as an input to the NLMS adaptive filter. NLMS adaptive filter is used to recover the origanal signal and minimize the noise singal. The scope shows the output of NLMS adaptivefilter to recover the orignal signal

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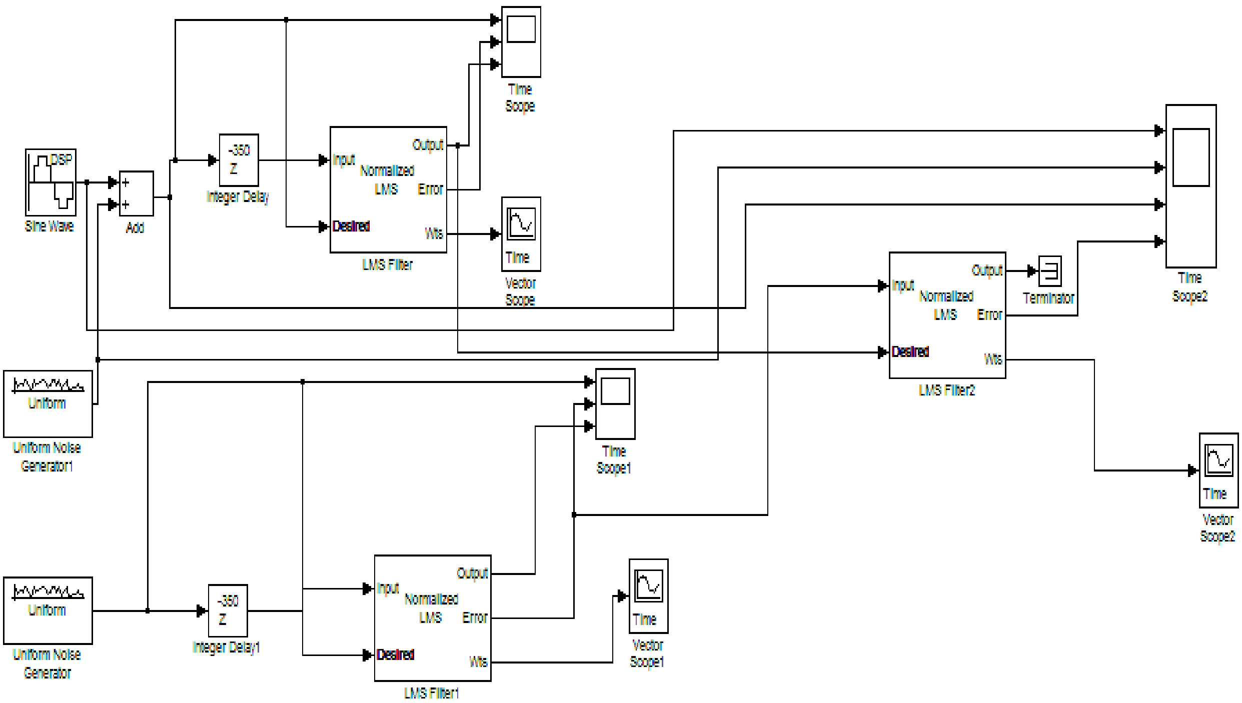

Fig.2 Simulink Model for NLMS Adaptive Filter for Noise Cancellation

III. RESULT

The Fig.3 ouputscope shows the output for NLMS adaptive filter.The first window shows the input signal Sine wave signal. The second window shows the uniform noise generator signal. This two signal are added and applied as an input to NLMS adaptive filter. The third window shows the input signal appied to the NLMS adaptive filter(noisy signal). The last window shows the output signal of NLMS filter, recovered original signal which is nearly same as the orignal input signal.

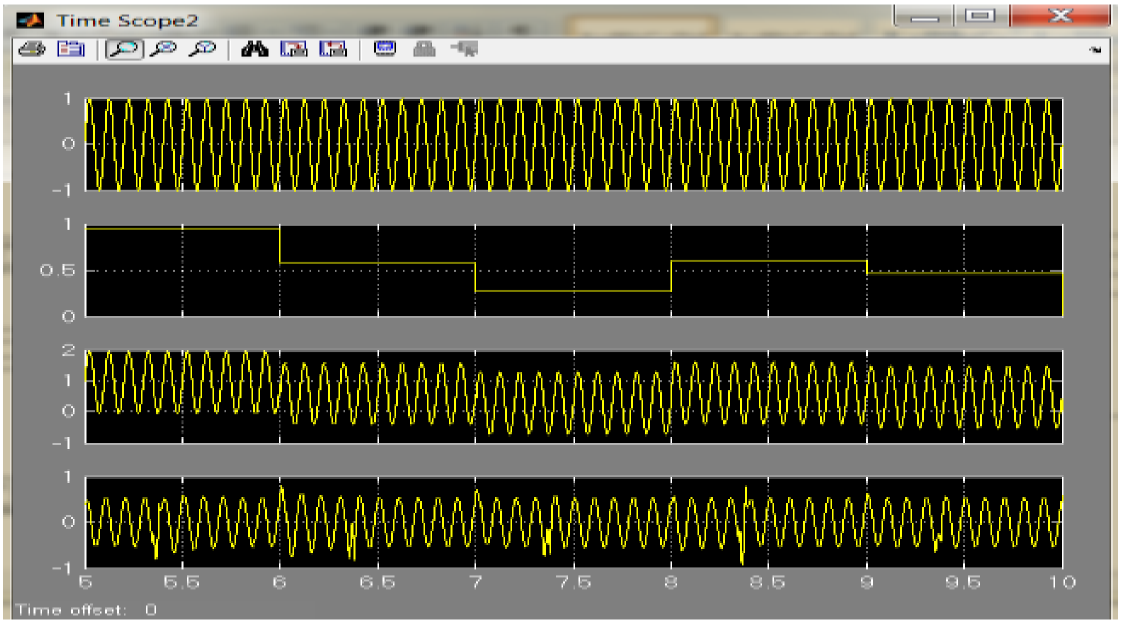

Fig. 3 Outptut Scope for NLMS Adaptive Filter

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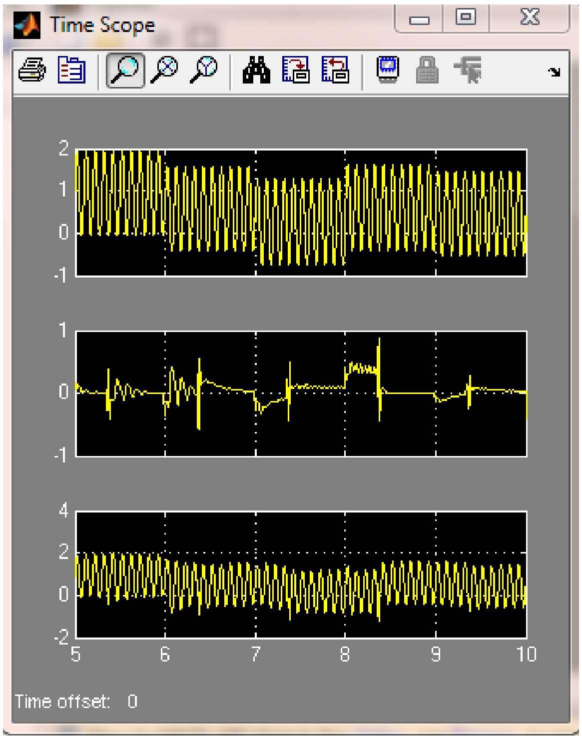

Fig 4 Output of First NLMS Filter

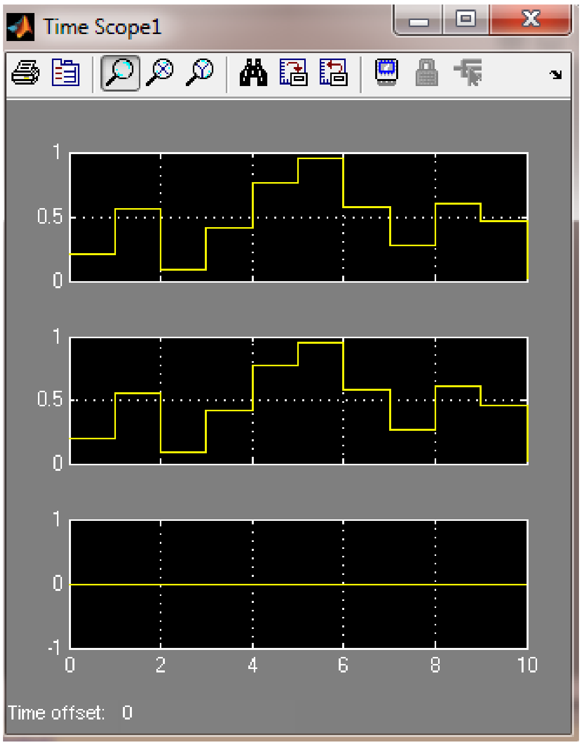

Fig 5 Output of Second NLMS Filter

IV. CONCLUSION

In this paper, the behavior of the NLMS algorithm were studied forinterference cancellation of input signals. Attempt was made to find out the effects ofthe filter length and the step size parameters of the algorithm. The results is presented in terms of the coefficients of the cross-correlation of the input and the filteredsignals.The analysis leads us to conclude that if the NLMS is chosen for interference cancellationof sine wave input signals, the step size should be smaller, in which case the filter length shouldalso be low. But if a larger step size is chosen, the filter length should be increased forbetter performance. By utilizing the findings of this study, new variable-step sizealgorithms can be developed in order to optimize filter performances.

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