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ORIGINAL ARTICLE

SIMULATION OF ADAPTIVE NOISE CANCELLATION

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Abstract:

In numerous applications of signal processing, communications and biomedical we are faced with the necessity to remove noise and distortion from the signals. Adaptive filtering is one of the most important areas in digital signal processing to remove background noise and distortion. As received signal is continuously corrupted by noise where both received signal and noise signal both changes continuously, then this arise the need of adaptive filtering. In last few years various adaptive algorithms are developed for noise cancellation.

The normalized least mean square (NLMS) algorithm is an important variant of the classical LMS algorithm for adaptive linear filtering. It possesses many advantages over the LMS algorithm, including having a faster convergence and providing for an automatic time-varying choice of the LMS stepsize parameter that affects the stability, steady-state mean square error (MSE), and convergence speed of the algorithm. An auxiliary fixed step-size that is often introduced in the NLMS algorithm

has the advantage that its stability region (step-size range for algorithm stability) is independent of the signal statistics.

This paper describes the development of an adaptive noise cancellation algorithm like NLMS(Normalized Least Mean Square)for effective recognition of signal on MATLAB platform .We simulate the adaptive filter in MATLAB with noisy signal and obtained result shows that NLMS algorithm eliminates noise from noisy signal and get desired result at the output.

KEY WORDS:

Adaptive Noise Cancellation,Adaptive filtering, LMS Algorithm, NLMS Algorithm

1. INTRODUCTION

Nowadays, effective communication is necessary to keep up with the fast-developing world. Effective voice communication is the most important part of it. In the prevailing environment, the noise corrupts the speech signal to such an extent, sometimes, that it is almost impossible to recover the original voice message communicated. That noise is usually given the name of background noise, which affects the intelligibility of the speech signal.

Acoustic noise problems becomes more pronounce as increase in number of industrial equipment such as engines, transformers, compressors and blowers are in use. The traditional approach to acoustic noise cancellation uses passive techniques such as enclosures, barriers and silencers to remove the unwanted noise signal [1][2]. Silencers are important for noise cancellation over broad frequency range but ineffective and costly at low frequencies. Mechanical vibration is a type of noise that creates problems in all areas of communication and electronic appliances. Signals are carriers of information, both useful and unwanted.

Signal processing is an operation designed for extracting, enhancing, storing, and transmitting useful information. Hence signal processing tends to be application dependent. In contrast to the conventional filter design techniques, adaptive filters do not have constant filter coefficients and no priori information is known. Such a filter with adjustable parameters is called an adaptive filter. In the most of practical applications Adaptive filters are used and preferred over fixed digital filters because adaptive filters have the property of self-modifying its frequency response and allowing the filter to adapt the response as the input signal characteristics change. Adaptive filter adjust their coefficients to minimize an error signal and can be realized as finite impulse response (FIR), infinite impulse response (IIR), lattice and transform domain filter [3]. The most common form of adaptive filter is the transversal filter using least mean square (LMS) algorithm and NLMS algorithm.

In this paper we investigate the performance of an adaptive NLMS algorithm with the help of MATLAB simulation and tested for sinusoidal signal. The paper is organized as follows. Section II gives an idea of adaptive filter, in section III gives an idea of adaptive algorithm. Section IV describes Adaptive Noise Cancellation (ANC) model. Section V provide the simulation results and finally section VI concludes the work.

2.ADAPTIVE FILTER

An adaptive filter has the property of self-modifying its frequency response to change the behaviour in time, allowing the filter to adapt the response to the input signal characteristics change. Due to this capability, the overall performance and the construction flexibility, the adaptive filters have been employed in many different applications, some of the most important are: telephonic echo cancellation, radar signal processing, navigation systems, communications channel equalization and biomedical signals processing [4-6,2].The most common adaptive filters, which are used during the adaption process, are the finite impulse response (FIR) types.

Fig.1 illustrates the general configuration for an Adaptive filter [7]. The adaptive filter has two inputs: the primary input d(n), which represents the desired signal corrupted with undesired noise, and the reference signal x(n), which is the undesired noise to be filtered out of the system.

The goal of adaptive filtering systems is to reduce the noise portion, and to obtain the uncorrupted desired signal. In order to achieve this task, a reference of the noise signal is needed. That reference is fed to the system, and it is called a reference signal x(n). However, the reference signal is typically not the same signal as the noise portion of the primary signal - it can vary in amplitude, phase or time delay. Therefore the reference signal cannot be simply subtracted from the primary signal to obtain the desired portion at the output.

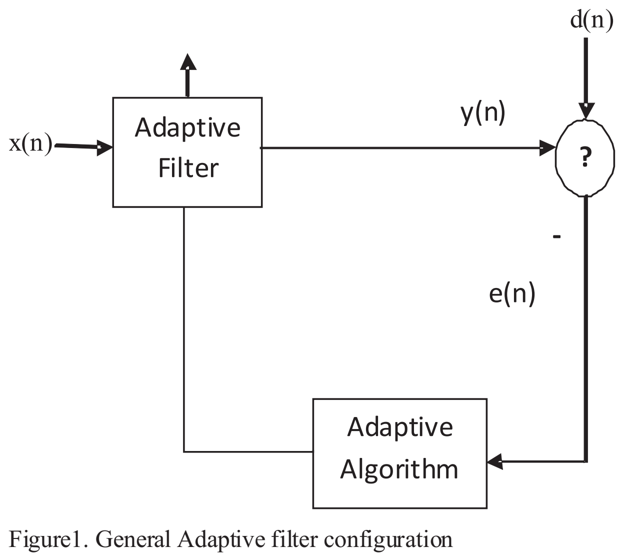

Figure1. General Adaptive filter configuration

The basic idea for the adaptive filter is to predict the amount of noise in the primary signal, and then subtract that noise from it. The prediction is based on filtering the reference signal x(n), which contains a solid reference of the noise present in the primary signal. The noise in the reference signal is filtered to compensate for the amplitude, phase and time delay, and then subtracted from the primary signal. This filtered noise is the system's prediction of the noise portion of the primary signal, y(n). The resulting signal is called error signal e(n), and it presents the output of the system. Ideally, the resulting error signal would be only the desired portion of the primary signal.

The adaptive filter can be realize on DSP Processors because they have huge number of applications in today's life, such as audio signal processing i.e. noise cancellation, system identification, equalization and etc. Besides audio signal processing, digital signal processing is also used in other kind of signal processing applications such as image processing, statistical signal processing, biomedical signal processing etc. DSP is widely used in high speed modems and mobile phones also due to availability of low cost DSP chips that can perform extensive computation in real time.

III.ADAPTIVE ALGORITHMS

The procedure for the algorithm is to adjust the filter coefficients for the adaptive filter in order to reduce a prescribed criterion. The algorithm is determined by minimization algorithm, the objective function, and the nature of error signal. The algorithms used to perform the adaptation, and the configuration of the filter depends directly on the use of the filter. However, the basic computational engine that performs the adaptation of the filter coefficients can be the same for different algorithms, and it is based on the statistics of the input signals to the system. The two classes of adaptive filtering algorithms namely Least Mean Squared (LMS) and Recursive Least Squares (RLS) are capable of performing the adaptation of the filter coefficients. The LMS based algorithms are simple to understand and easy to implement whereas RLS based algorithm are complex and requires so much memory for implementation. However, LMS suffers from a slow rate of convergence. Further, its implementation requires the choice of an appropriate value for the step-size that affects the stability, steady-state MSE, and convergence speed of the algorithm. So in this work we have focuses on NLMS based algorithms.

A. Least Mean Square Algorithm

To make exact measurements of the gradient vector $j(n)$ at each iteration n, and if the step-size parameter μ is suitability chosen then the tap-weight vector computed by using the steepest descent algorithm would converge to the optimum wiener solution. The exact measurements of the gradient vector are not possible and since that would require prior knowledge of both the autocorrelation matrix R of the tap inputs and the cross correlation vector p between the tap inputs and the desired response, the optimum wiener solution could not be reached [8]. Consequently, the gradient vector must be estimated from the available data when we operate in an unknown environment.

After estimating the gradient vector we get a relation by which we can update the tap weight vector recursively as:

$$W(n+1)=w(n)+\mu\, u(n)[d^*(n)-uH(n)w(n)] \qquad (1)$$

Where μ=step size parameter
$u^H(n)$=Hermit of a matrix u
$d^*(n)$=Complex conjugate of d(n)

Here x(n) is the input vector of time delayed input values,

$$X(n)=[x(n)x(n-1)x(n-2)\ldots..x(n-N+1)]^T \qquad (2)$$

The vector $w(n)=[w_0(n)w_1(n)\ w_2(n)\ldots..\ wN-1(n)]T$ represent the coefficients of the adaptive FIR filter tap weight vector at time n.

We may write the result in the form of three basic relations as follows:

1.Filter output

$$y(n) = w^H(n)\,u(n) \qquad (3)$$

2.Estimation error or error signal:

$$e(n) = d(n) - y(n) \qquad (4)$$

3. Tap weight adaption:

$$w(n+1) = w(n) + \mu\, u(n)\, e^*(n) \qquad (5)$$

Equations (3) and (4) define the estimation error e(n), the computation of which is based on the current estimate of the tap weight vector w(n). Note that the second term, u(n)e*(n) on the right hand side of equation (5) represents the adjustments that are applied to the current estimate of the tap weight vector w(n) The iterative procedure is started with an initial guess w(0). The algorithm described by equations (3) and (4) is the complex form of the adaptive least mean square (LMS) algorithm. At each iteration or time update, this algorithm requires knowledge of the most recent values u(n), d(n) w(n). The LMS algorithm is a member of the family of stochastic gradient algorithms.

In particular, when the LMS algorithm operates on stochastic inputs, the allowed set of directions along which we "step" from one iteration to the next is quite random and therefore cannot be thought of as consisting of true gradient directions.

The parameter μ is known as the step size parameter and is a small positive constant. This step size parameter controls the

influence of the updating factor. Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long; if μ is too large the adaptive filter becomes unstable and its output diverges.

B. Normalized Least Mean Square Algorithm

In the standard LMS algorithm, when the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. This difficulty is solved by NLMS (Normalized Least Mean Square) algorithm. The correction applied to the weight vector w(n) at iteration n+1 is "normalized" with respect to the squared Euclidian norm of the input vector x(n) at iteration n.

The NLMS algorithm can be viewed as a time-varying step-size algorithm, calculating the convergence factor μ as in Eq. (6)

$$(n) \quad \frac{}{c \quad \| x(n)\|^2} \qquad (6)$$

where α is the NLMS adaption constant, which optimize the convergence rate of the algorithm and should satisfy the condition 0<α<2, and c is the constant term for normalization, which is always less than 1.

The filter weights using NLMS algorithm are updated by the Eq .

$$w(n \quad 1) \quad u(n) \quad \frac{}{c \quad \| x(n) \|^2} e(n)x(n) \qquad (7)$$

IV.ADAPTIVE NOISE CANCELLATION

Adaptive noise cancellation (ANC) is performed by subtracting noise from a received signal, and an operation controlled in an adaptive manner is done during the adaptation process to get an improved signal-to-noise ratio. Noise subtraction from a received signal could generate disastrous results by causing an increase in the average power of the output noise. However when filtering and subtraction are controlled by an adaptive process, it is possible to achieve a superior system performance compared to direct filtering of the received signal. Fig.2 shows adaptive noise cancelling system.

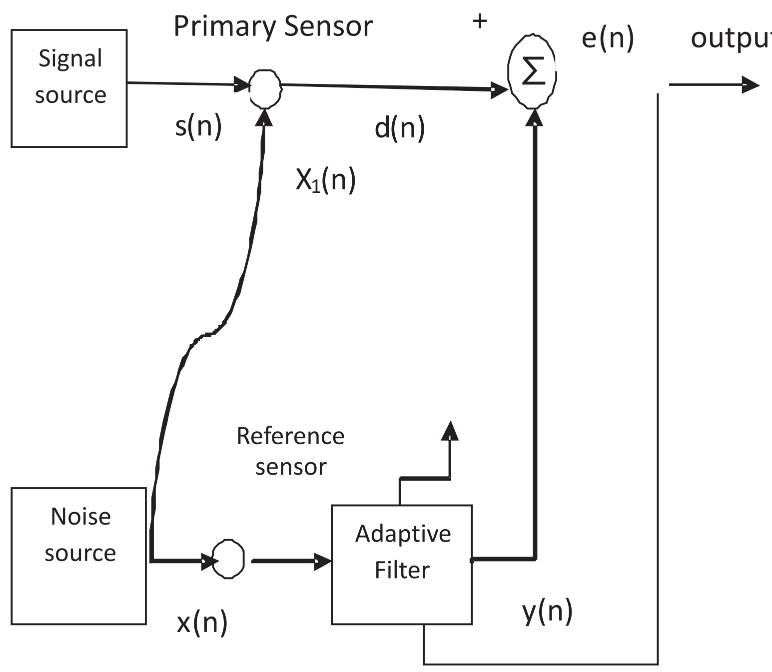

Figure 2. Adaptive Noise Cancellation system

The ANC system composed of two separate inputs, a primary input i.e. source signal s(n) and a reference input i.e. noise input x(n). The primary signal is corrupted by a noise x1(n) which is highly correlated with noise signal x(n). The desired signal d(n) results from addition of primary signal s(n) and correlated noise signal x1(n). The reference signal x(n) is fed into adaptive filter and its output y(n) is subtracted from desired signal d(n). The output of the summer block is then fed back to adaptive filter to update filter coefficients. The above process is run recursively to obtain the noise free signal which is supposed to be the same or very similar to primary signal s(n).

V.SIMULATION RESULTS

The simulation is done in MATLAB Simulink, using the model shown in figure 3.The adaptive noise canceller was implemented in MATLAB for NLMS algorithms.

For the purpose of noise cancellation in signal corrupted by random source, the NLMS algorithm was simulated and tested using MATLAB as shown in fig 3.In the simulation the reference input signal x(n) was a random source of Gaussian noise in MATLAB, and source signal s(n) was a clean amplified sinusoidal signal as shown in fig 4, the desired signal d(n) obtained by adding a delayed version of x(n) into a clean signal s(n),d(n)=s(n)+x1(n) as shown in fig 5.Fig 6 shows the NLMS filtered output signal.The simulation of the NLMS algorithm was carried out with the following specifications:

Filter order N=32, step size $\mu = 0.1$ and leakage factor=1

The step size μ control the performance of the algorithm, if μ is too large the convergence speed is fast but filtering is not proper, if μ is too small the filter gives slow response, hence the selection of proper value of step size for specific application is prominent to get good results.

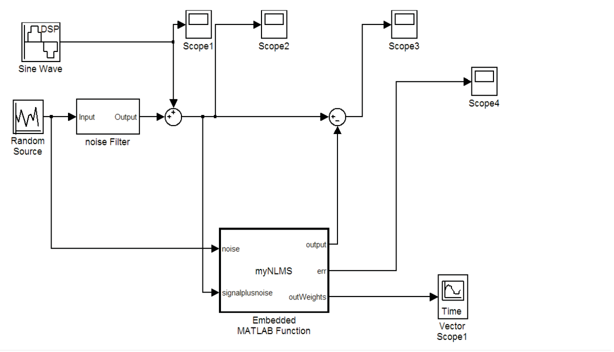

Fig 3 Simulink Model for ANC System

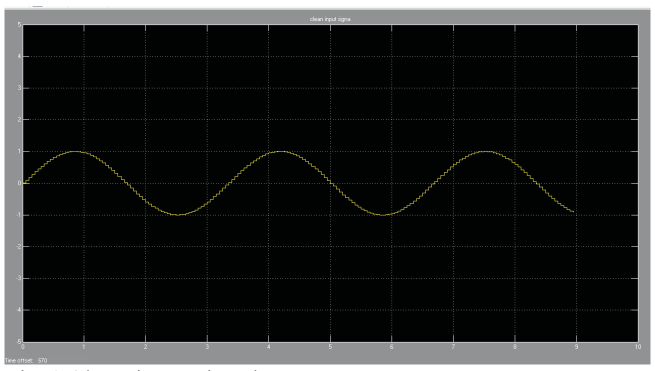

Fig 4 Clean input signal

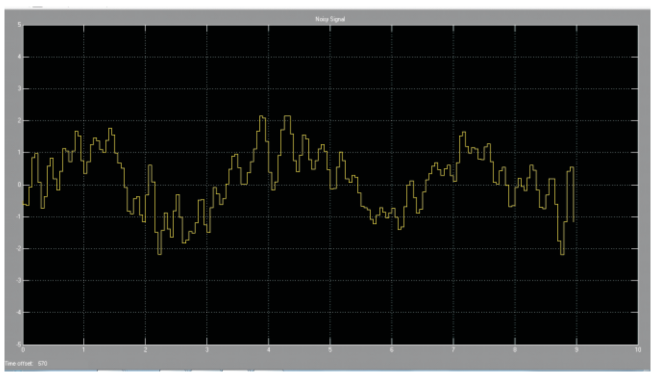

Fig 5.Noisy Signal

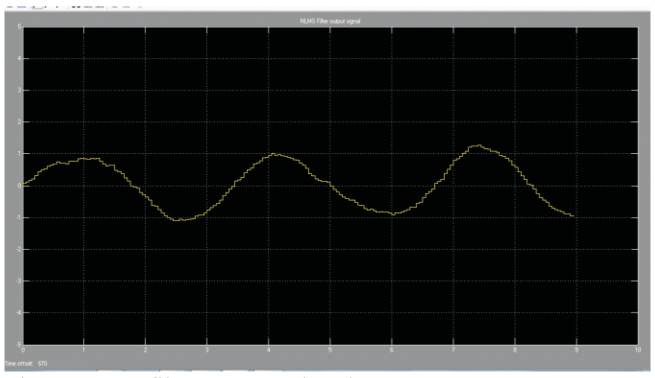

Fig 6 NLMS filter Output Signal

VI.CONCLUSION

The main objective of this paper was to implement an adaptive noise canceller for de-noising signal . This paper provides background information on FIR adaptive filter structure and adaptive algorithms, LMS and NLMS algorithms. Based on these discussions a transversal FIR adaptive filter with NLMS updating algorithm designed for this paper. The NLMS algorithm changes the step-size according to the energy of input signals hence it is suitable for both stationary as well as non-stationary environment and its performance lies between LMS and RLS. Hence it provides a trade-off in convergence speed and computational complexity. The implementation of algorithms was successfully achieved, with results that have a really good response.

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