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Comparative Study of Different Algorithms for the Design of Adaptive Filter for Noise Cancellation

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ABSTRACT
The main goal of this paper is to study and to compare the performance of different adaptive filter algorithms for noise cancellation. Adaptive noise cancellation method is used for estimating a speech signal which is corrupted by an additive noise. The reference input containing noise is adaptively filtered and subtracted from the primary input signal to obtain the de-noised signal. The desired signal which is corrupted by an additive noise can be recovered by an adaptive noise canceller using Least Mean Square (LMS) algorithm, Data Sign algorithm, Leaky LMS algorithm and constrained LMS algorithm. A performance comparison of these algorithms based on Signal to Noise Ratio(SNR) is carried out using MATLAB.
Keywords-Adaptive Filter, Adaptive algorithms, MATLAB, Noise cancellation System, SNR

I INTRODUCTION

Noise is disturbance unwanted signal during communication. Noise can occur because of many factors like interference, delay, and overlapping. Noise problems in the environment are obtained due to the enormous growth of technology that has led to noisy engines, heavy machinery and other noise sources. Noise cancellation system employed for variety of practical applications such as the cancelling of various forms of periodic interference in electrocardiography, the cancelling of periodic interference in speech signals. Adaptive filtering has been extensively used in many practical applications. Important results have been obtained, for instance, in noise and interference cancelling for biomedical applications [1]. In the process of digital signal processing for noise or time varying signals, Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) fixed coefficient filters cannot achieve optimal filtering. So, we must design adaptive filters, to provide the changes of signal and noise signal. Adaptive filter technology shows better performance as compared to conventional methods.

Section 2 gives an overview of adaptive filters. The brief description of noise cancellation system is made in section 2. The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise signal while leaving the signal unchanged. This is an adaptive process, which means it does not require a priori knowledge of signal or noise characteristics. Adaptive Noise Cancellation (ANC) completely attenuates the low frequency noise for which passive methods are ineffective. Section 4 introduces about adaptive algorithms such as Least Mean Square (LMS), Data Sign LMS, Leaky LMS, and Constrained LMS. Section 5 shows the results, observations and comparison of these algorithms on the basis of SNR. Section 6 concludes the main research work.

II ADAPTIVE FILTER

It is a filter that self-adjusts its transfer function according to the best algorithm operated by an error signal. Because of the complicated of these algorithms, most adaptive filters are digital filters. Adaptive filters are required for some applications because some parameters of the desired processing action are not known in progress [2] [3]. The adaptive filter uses feedback in the form of an error signal to filter its transfer function to associate the changing parameters. The adaptive process involves the use of a cost function which is a criterion for foremost performance of the filter, to deliver an algorithm, which determines how to alter filter transfer function to minimize the cost on the next iteration. Fig.1 shows the block diagram of adaptive filter [4].

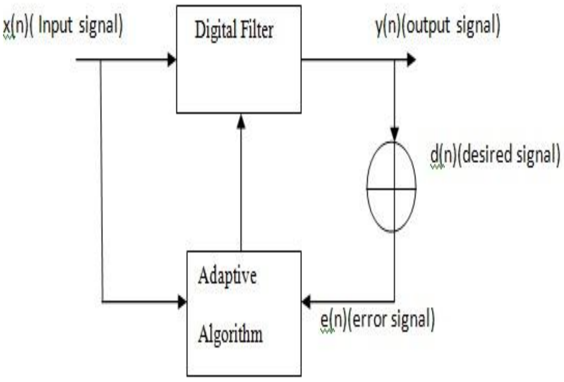

Fig 1: Block diagram of adaptive filter

III Noise Cancellation System

The general configuration of Noise Cancellation System [5] is shown in Fig.2. It has two inputs, the corrupted signal $d(n)$, which represents the desired signal $s(n)$ corrupted by an undesired noise $x_I(n)$, and the reference signal $x(n)$, which is the unwanted noise to be filtered out of the system. The goal of Noise Cancellation system is to reduce the noise signal, and to obtain the uncorrupted de-noised signal. In order to achieve this, a reference of the noise signal is required which is called as reference signal $x(n)$. However, the reference signal is usually not the same signal as the noise portion of the primary amplitude, phase or time. So, the reference signal cannot be simply subtract from the primary input signal to obtain the desired portion at the output. In general, noise that affects the speech signal can be modeled as White noise or Colored noise.

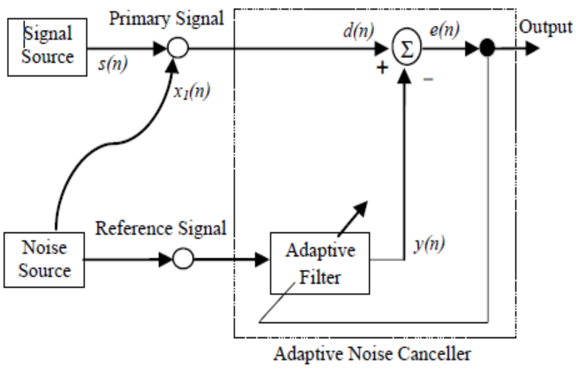

Fig 2: Adaptive noise cancellation system

Where $s(n)$-source signal, $d(n)$ corrupted signal, $x_I(n)$-noise signal, $x(n)$-noise reference input, $y(n)$-output of adaptive filter, $e(n)$-system output signal.

Adaptive Noise Cancellation system utilize two signals, One signal is used to measure the speech with noise signal while the other signal is used to measure the noise signal alone. This technique adaptively adjusts a set of filter coefficients so as to remove the noise from the corrupted signal. This technique requires that the noise component in the corrupted signal and the noise in the reference signal have high coherence. Unfortunately this is a limiting factor, as the microphones need to be space apart in order to prevent the speech being included in the noise reference and thus it being removed. In summary, to realize an adaptive noise cancellation system we use two inputs and an adaptive filter. One input is the signal corrupted by noise which can be expressed as:

$$d(n) = s(n) + x_I(n) \quad (1)$$

The other input contains noise related in some way to that in the main input but does not contain anything related to the signal and it is known as noise reference input signal which can expressed as $x(n)$. The noise reference input pass through an adaptive filter and output $y(n)$ is produced as close a replica as possible of $x_I(n)$.The filter readjusts its filter coefficients itself continuously to minimize the error between $x_I(n)$ and $y(n)$ during this process. Then the output $y(n)$ is subtracted from the corrupted signal to produce the system output. It is represented as:

$$e(n) = s(n) + x_I(n) - y(n) \quad (2)$$

This is the de-noised signal.

IV Algorithms Of Adaptive Filters

The LMS adaptive filter family is very attractive for implementation of low-cost real-time systems due to its low computational intricacy and robustness [6] [7]. One of the most popular adaptive algorithms available in the literature is the stochastic gradient algorithm also called LMS [2] [3].

4.1 LMS Algorithm

This is extensively used for different applications such as channel equalization, echo cancellation and noise cancellation. The equation below is LMS algorithm for updating the tap weights of the adaptive filter for each iteration.

$$w(n+1) = w(n) + \mu e(n) x(n) \quad (3)$$

Where x(n) is the input vector of time delayed input values and w(n) is the weight vector at the time n. μ is the step size parameter. This algorithm is used due to its computational simplicity. It requires 2N+1 multiplications and additions but it has a fixed step size for each iteration.

4.2 Data Sign LMS algorithm

In a high speed communication the time is critical, thus faster adaptation processes is needed

$$\operatorname{sgn}(a) = \begin{cases} 1 & a > 0 \\ 0 & a = 0 \\ -1 & a < 0 \end{cases} \quad (4)$$

For data Sign algorithm [7] weight update coefficients equation is:

$$w(n+1) = w(n) + 2\mu e(n) sign(x(n)) \quad (5)$$

By introducing the signum function and setting μ a value of power of two, the hardware implementation is highly simplified. It improves the convergence behavior, requires less computational complexity and also provides good result but throughput is slower than LMS Algorithm.

4.3 Leaky LMS Algorithm

It introduces a leakage coefficient into LMS algorithm so it becomes as:

$$w(n+1) = (1 - 2\mu\gamma)w(n) + 2\mu e(n)x(n) \quad (6)$$

Where $0 < \gamma << 1$ .The effect of introducing the leakage coefficient γ is to force any undamped modes to become zero and to force the filter coefficients to zero if either $e(n)$ or $x(n)$ is zero.

4.4 Linearly constrained LMS Algorithm

In LMS algorithm, no constrain was imposed on the solution of minimizing the MSE. However, in some applications there might be some mandatory constraints that must be taken into consideration in solving optimization problems. The problem of minimizing the average output power of a filter while the frequency response must remain constant at specific frequencies. In this we discuss the filtering problem of minimizing the MSE subject to a general constraint. This algorithm has following two steps:

Step 1: $w'(n) = w(n) + 2\mu e(n)x(n)$ (7)

Step 2: $w(n+1) = w'(n) + \eta(n)$ (8)

using the Lagrange multiplier method that gives where

$$\eta(n) = \frac{a - c^T w'(n)}{c^T c} c \quad (9)$$

To obtain final form:

$$w(n+1) = w'(n) + \frac{a - c^T w'(n)}{c^T c} c \quad (10)$$

Where c is constant vector and a is constraint constant.

V RESULTS AND OBSERVATIONS

SNR is defined as the power of the desired signal divided by the noise power. It is measured in Decibel(dB).It is a measure used in science and engineering that compares the level of a desired signal to the level of background noise.

$$SNR_{dB} = 10\log_{10}\left(\frac{P_{signal}}{p_{noise}}\right) \quad (11)$$

.The simulation results show that LMS, Data Sign LMS, Leaky LMS, Linearly Constrained LMS algorithms are used to cancel the noise and provides good results. Convergence of the adaptive for the choices of step size parameter μ which is constant is very sensitive. The order of the filter was set to M=40. Original input signal having sampling frequency 500Hz is corrupted by adding white Gaussian noise. No. of Samples is 4000. No of iterations are same as no. of samples. The value of parameter μ varies for all algorithms for good result. Frequency response of de-noised signal should be same as the original signal. This can be achieved by using different types of algorithms. Fig. 3 shows the original signal, corrupted signal and reference input signal. Fig. 4 shows the frequency response of original signal and corrupted signal. The parameter μ is set to 0.001.Fig.5 shows the frequency response of de-noised signal by using LMS algorithm. Fig.6 shows the frequency response of de-noised signal by using Data Sign LMS. Fig.7 shows the frequency response of de-noised signal by using leaky LMS. Fig.8 shows the frequency response of de-noised signal by using linearly Constrained LMS. Table 1 show that the results of linearly Constrained LMS are better than LMS, Sign LMS and Leaky LMS.

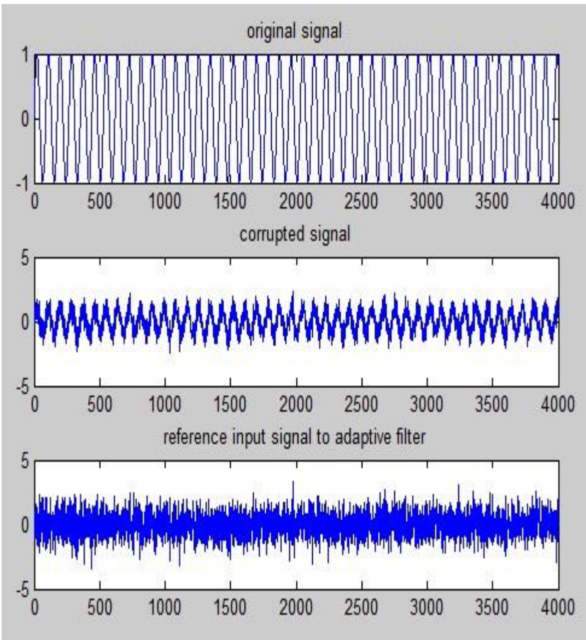

Fig 3: Original signal, corrupted signal and reference input signal

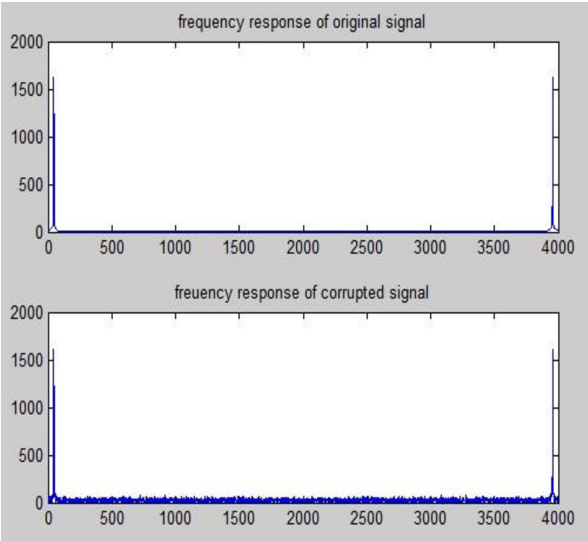

Fig 4: Frequency response of original signal and corrupted signal.

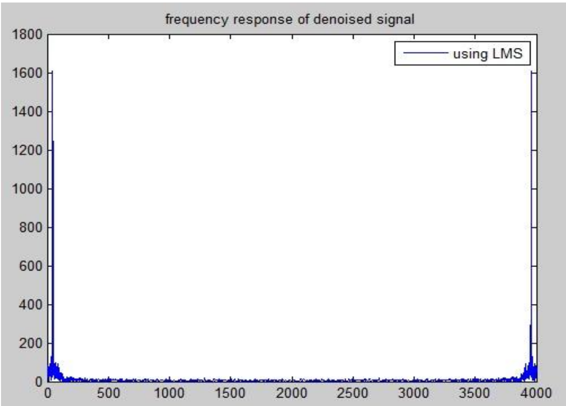

Fig 5: Frequency response of de-noised signal by using LMS algorithm

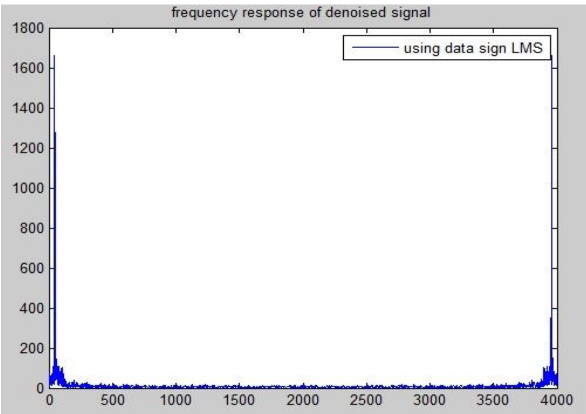

Fig 6: Frequency response of de-noised signal by using data Sign LMS.

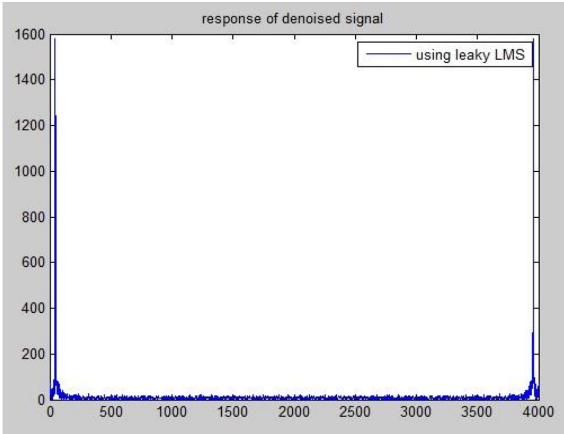

Fig 7: Frequency response of de-noised signal by using leaky LMS

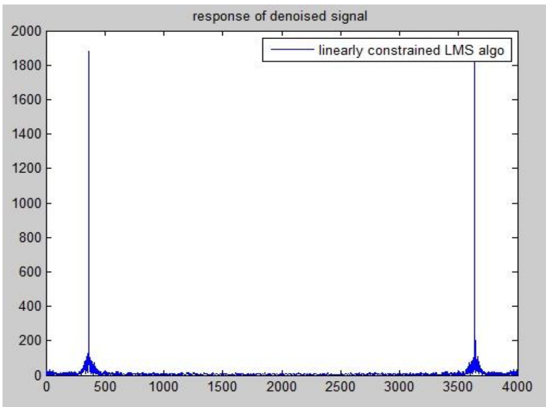

Fig 8: Frequency response of de-noised signal by using linearly constrained LMS

Table 1: Performance Comparison

Algorithms	SNR before(dB)	SNR after (dB)
LMS	-1.8887	10.34
Data-Sign LMS	-1.9676	10.478
Leaky LMS	-2.1013	11.5701
Linearly Constrained LMS	-1.9631	11.7829

VI CONCLUSION

From the above discussion, it has been concluded that four different types of adaptive algorithms are used for noise cancellation and for improving SNR after adaptive filtering. Linearly Constrained LMS algorithm provides high SNR as compared to LMS, Data Sign LMS, and Leaky LMS algorithms. The future work will be directed to determine SNR using Error Data Normalized steady state (EDNSS) algorithm.

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