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A Novel LMS Algorithm Applied to Adaptive Noise Cancellation with Varying Parameters

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Abstract: Adaptive filters have become active area of research in the field of communication system. This paper explores the novel concept of adaptive noise cancellation (ANC) using least-mean-square (LMS) adaptive filters. The model of the LMS-ANC is designed and simulated in MATLAB environment. The proposed algorithm utilizes adaptive filters to evaluate gradients accurately which results in good adaptation, stability and performance. The objective of this investigation is to provide solution in order to improve the performance of noise canceller in terms of filter parameters. The results are obtained with the help of adaptive algorithm with variable step size and filter order in order to deliver high convergence speed and stability of the error signal.
Keywords: Adaptive Noise cancellation, LMS algorithm, MATLAB, Filter order, Step size.

I. INTRODUCTION

Noise is an unwanted sound, which has a sound pressure level that exceeds the human hearing threshold. In the long term of exposure, noise can be bad for health, one of which is causing hearing damage [1]. The noise can originate from a single source or various sound sources in the environment. In the current scenario of modern technology, we are facing a necessity of noise removal in signal processing. This cleaning process is often referred to as noise reduction [2]. Various approaches are used for the same. The two mainly used methods to degrade the noise, are Passive noise reduction/ Passive noise control (PNC) and Active noise reduction/ Active noise control (ANC) [13].

The passive noise reduction method reduces the noise by using sound absorption technique with the help of physical materials such as using earplugs, ear-protector, sound insulation walls etc. Meanwhile, the active noise reduction or Active noise control (ANC) method was proposed in the early 20th century which reduces the noise by generating a secondary sound source, which has a phase opposite to unwanted noise, so that by using Young's sound wave interference principle, destructive interference is generated [3] and we get a clear signal. For the basic concept of Young's sound wave interference principle, refer figure 1.

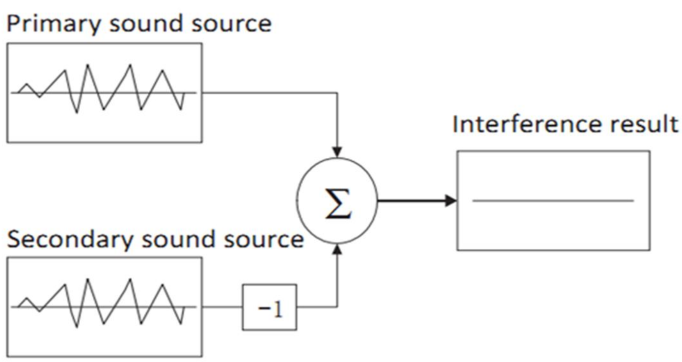

Figure 1. Schematic diagram of the Young’s Interference Principle sound wave control [3].

The active noise control (ANC) method has several advantages over the passive noise reduction method. First, the control system parameters can be designed in accordance with the target noise to be reduced. Second, it has a better control effect which make it capable to attenuate the low frequencies noise [4] and third, this method is more flexible to use, implement low cost and is easier to install [5].

The noise cancellation techniques discussed above [6] provides an improved quality of speech signal that helps in achieving a better performance. One of the main objectives of this paper is to present in a common context, an overview of the latest noise reduction algorithm with varying parameters.

There can be different types of noises present in the surroundings, like additive white Gaussian noise, shot noise, random noise, thermal noise etc. These noise signals add themselves onto the speech signal and create problems by corrupting the original signal. There are number of methods available to remove this interference from the original signal. The method implied here to de-noise the original signal is Adaptive Noise Cancellation. Adaptive Noise Cancellation or Active Noise Control (ANC) is a method in which a reference noise, which is inputted, is adaptively subtracted from the original noise signal [13].

The advantage of ANC lies in the fact that the noise cancellation can be done to such an extent that no other digital signal processing tool can attain. Suppose an input signal 'I', which has 'N' noise already added to it, is transmitted and received, Refer to Figure 1.

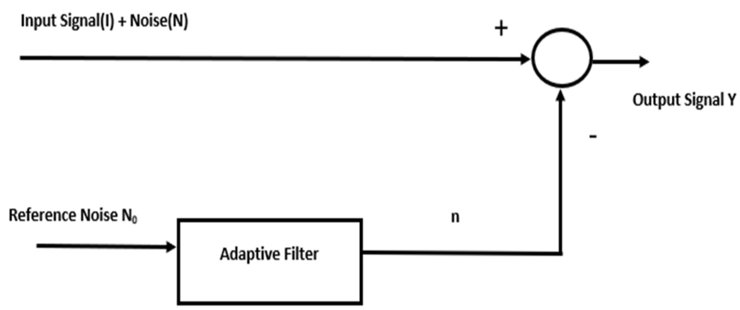

Figure 2: Basic block diagram of an ANC filter

Reference noise 'N0' is used to create an estimation of noise signal 'n', using the adaptive techniques, this estimated signal is subtracted from the input to get the final de-noised signal. The final output after filtering is I+(N-n) which is the Output signal 'Y', as seen in the Fig.1.

$$I+(N-n) = Y$$

There are various kinds of adaptive algorithms present, like Least Mean Square (LMS) Algorithm, Normalized Least Mean Square (NLMS) Algorithm and Recursive Least Square (RLS) Algorithm, amongst which we would be employing the Least Mean Square (LMS) algorithm in this paper to cancel the noise and will be varying and comparing their parameters in de-noising the signals [7].

Many researchers [2],[9],[7],[8] has stated through their research that out of different noise filtering methods available in the signal processing, LMS is a better option to go for as it has shown successful results with simpler calculations and stability

II. LEAST MEAN SQUARE

LMS algorithm was developed by Widrow and Hoff in 1959 as an adaptive filter which can be used for noise reduction. Fig. 3 shows a diagram of the LMS algorithm. The parameters d(n) and x(n) are the inputs of the algorithm in the form of column vector. In this d(n) is the desired signal which is the noise corrupted signal and x(n) is the noise signal. The parameter W(n) is the column weight vector of the filter at nth time, which is used in the algorithm to update the subsequent column weight vector

The error e(n) is defined as the difference between the noises corrupted signal and the weighted-noise signal as shown the equation below,

$$e(n)= d(n)-y(n)$$

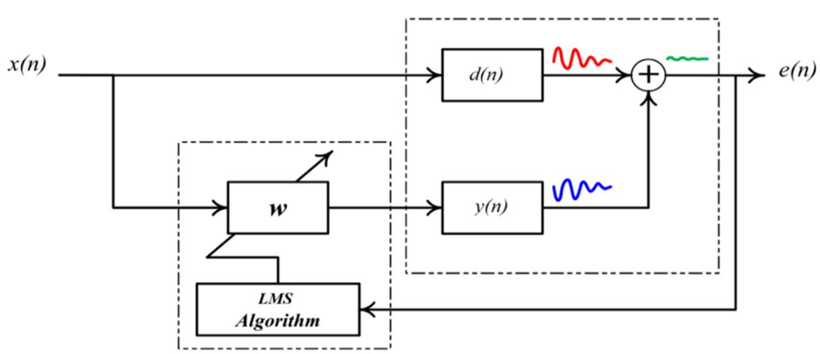

Figure 3. Diagram Block of LMS Algorithm [10]

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Among all the existing adaptive algorithms, the most successful adaptive algorithm is the LMS algorithm. It is particularly attractive for its low-cost real-time implementations, low computational complexity and good stability [10].

In this paper, we propose a noise reduction method based on the Least Mean Square (LMS) adaptive algorithm of audio signals. LMS algorithm is an algorithm in which the desired signal is restored by passing the noisy speech signal through a filter whose coefficients are estimated by minimizing the mean square error (MSE) between the clean signals. The weights of the filter are updated in a manner that they converge to the actual filter weights. This type of adaptive filter has a simple design, high convergence rate and easy computations [11]. When we talk about convergence rate, we talk about the process of minimizing the error signal. LMS algorithm is easy to solve mathematically as it does not involve any complex operations.

The Algorithm can be determined using the equation given below:

$$w(n+1) = w(n) + 2\mu * e(n) * x(n)$$

where, $x(n)$ is the input vector of time delayed input values, $w(n)$ represents the coefficients of the adaptive FIR filter tap weight vector at time n, $w(n+1)$ is the filter coefficient for the next iteration, $e(n)$ is the error value and μ is known as the step size which is introduced here to control the step width of the iteration and thus the stability and convergence or divergence rate of the algorithm.

It is well known that the performance of LMS- based algorithms depends directly on the choice of the step-size parameter. Larger step-sizes speed up the convergence rate at the expense of a larger steady-state adaptation. Smaller step-sizes tend to improve steady-state performance at the cost of a slower adaptation. Therefore, μ is determined to be optimal by test experiments using MATLAB environment [14].

III. IMPLEMENTATION AND ANALYSIS

Implementation of adaptive noise cancellation using LMS filter requires an input signal. Noise is added to this signal. This added noise creates problems by deforming the original signal.

Algorithms like RLS, LMS, NLMS can be integrated to free the signal from the added noises. To conduct this, a procedure needs to be followed in MATLAB. Each algorithm has its own mathematical calculations and methods that's need to be followed. To de-noise the signal, the following steps can be followed that has been explained below in the form of a Flow chart. Refer Figure no 4.

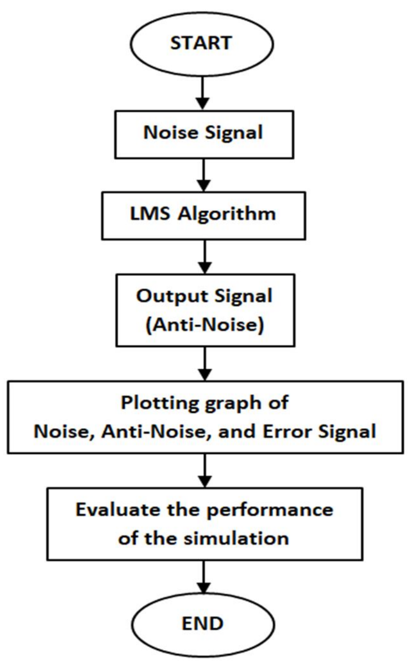

Figure 4. Flowchart of LMS adaptive filter algorithm implementation [12]

Thus, in this way, we can implement the adaptive noise cancellation algorithm in MATLAB and get the desired results.

The same procedure can be repeated with varying parameters like the step size, μ and filter order, M to compare the simulation results which are discussed in next section.

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IV. SIMULATIONS, RESULTS AND DISCUSSIONS

The adaptive filter algorithm which is discussed in this paper is the Least Mean Square (LMS) algorithm. This has been implemented and simulated using the MATLAB environment. The comparison between the graphs is done on the basis of step size, μ, and the filter order M.

In this section, we evaluate the performance of LMS algorithms in noise cancellation setup. Two signals were added and subsequently fed into the simulation of LMS adaptive filter. The order of the filter M and The Step-size , μ is varied. Outputs are obtained for particular values of step size and filter order i.e., $\mu = 0.003$, M=15 and $\mu = 0.03$, M= 25.

Below shown are the simulation results, for particular values of step size and filter order as mentioned above. Also, the frequency of the signal is chosen to be 60 Hz in both the cases. In the below figure, the simulation results of a series of steps for the LMS algorithm are shown. The desired signal is represented in black line and the corrupted signal is represented in green line.

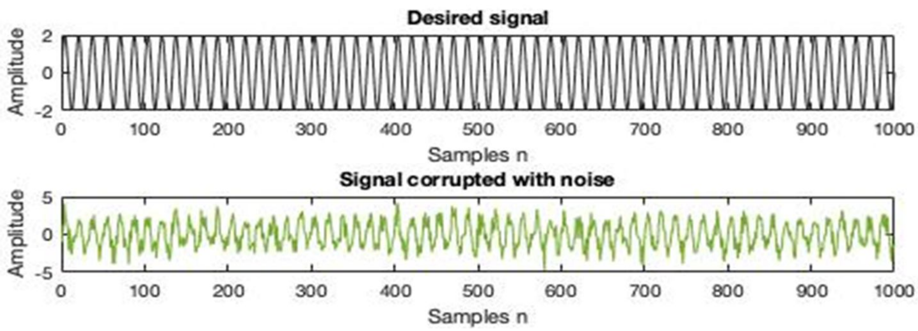

Figure 5: LMS Algorithm simulation

Below Fig. 6 indicates LMS results for step-size $\mu = 0.03$ and M = 25, with their respective error signal e(n) which is shown in blue line and the estimated signal is shown in red line.

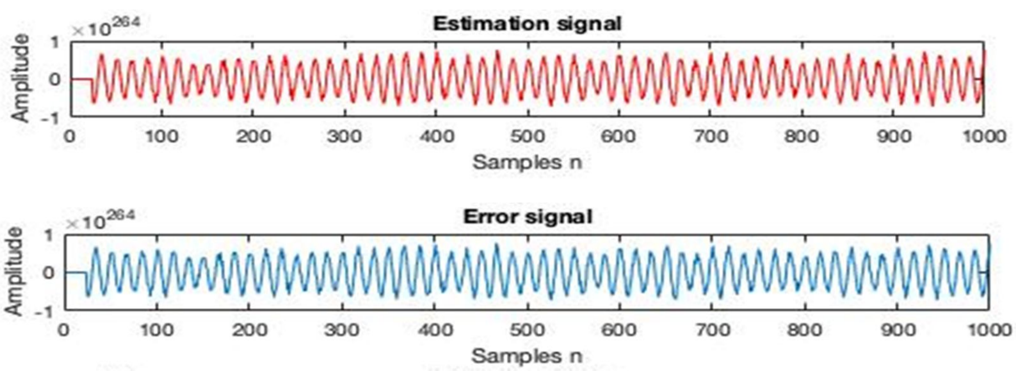

Figure 6: Result of LMS Algorithm with $\mu = 0.03$ and M = 25 and its error signal

Below Fig. 7 indicates LMS results for step-size $\mu = 0.003$ and M = 15, with their respective error signal e(n) which is shown in blue line and the estimated signal is shown in red line.

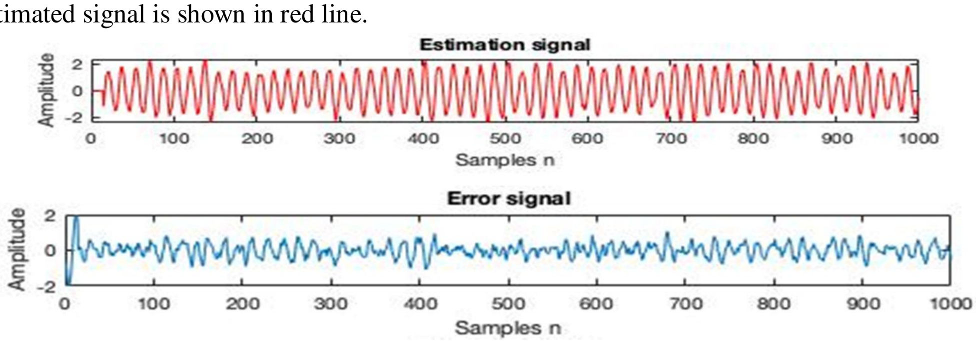

Figure 7: Result of LMS Algorithm with $\mu = 0.003$ and M = 15 and its error signal

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Figure 5,6 and 5,7 shows the comparison between the noise corrupted speech signal and the LMS adaptive algorithm filtered output (on the basis of varying step size and filter order). It is clear that the implementation of the LMS adaptive filter algorithm on the noise corrupted speech signal reduces the noise signal significantly. The amplitude and the noise of the signal are reduced. This filtered output is then compared with the original clean speech signal (desired signal). This comparison shows that the implementation of the LMS adaptive filter algorithm is able to attenuate the noise to a good extent.
In the simulation results, the best performance of the LMS algorithm is obtained when the LMS order value is 15 and the step-size value is 0.003. Therefore, 0.003 is the best choice as it is small enough but efficient to meet the design requirements to get the optimal convergence.

V. CONCLUSION

When the information signal travels in the free environment, it gets degraded by the noise present in it. Removing this noise appears out to be one of the most important concern for everyone. There are many conventional techniques to suppress the noise present in the information signal. In this paper, the simulation model of noise cancellation using LMS adaptive filter algorithm has been presented. The simulation model shows that the LMS adaptive filter algorithm is capable to update the column weight vector adaptively in approximating the noise signal to be cancelled. The implementation of LMS adaptive filter algorithm on the noise corrupted speech signal has proven that the adaptive filter can effectively remove the noise signal at lower frequencies range. Different applications present different challenges for the adaptation of the algorithm, thus requiring different step-size and filter order update. It may be visible from the experimental outcomes that the proposed approach correctly reduces the noise from the selected signal. Results are shown in the previous section after varying the parameters which is implemented and simulated using MATLAB environment.

V. ACKNOWLEDGMENT

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