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COMPARISON OF LMS ALGORITHM WITH VARYING STEP SIZE FOR NOISE CANCELLATION

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Abstract: Digital noise reduction technique, to enhance the quality of speech is a vast area of research now-a-days. This investigation focuses on the noise reduction methods employing the technique of adaptive filtering, using a popular LMS algorithm. The simplicity and relative effectiveness of this noise reduction algorithm has resulted in explosive growth in its use for a variety of speech communications applications. In this paper a comparative study by varying the parameters of LMS algorithm is done. Implementation and analysis of the filters are done by taking different step sizes on same orders of the filters.

Index Terms – **Noise cancellation, LMS algorithm, MATLAB, Adaptive filtering, step size.**

I. INTRODUCTION

Noise enters speech communications systems in many ways. Traditional wire-line telephone calls or calls made from public telephone booths, cellular or wireless telephones serve as an example. It is common for such communications to be degraded by noise of varied origin. Acoustical characteristics of the environmental noises are generally supposed to be quiet, Therefore, it is very uncommon for heating, ventilation equipment, traffic, crowds or air-conditioning systems to contribute substantial levels of noise to the environment and thus to the speech communications systems [9].

A variety of approaches have been proposed to reduce noise for the purpose of speech enhancement. As a result, the communication signal has to be cleaned up with digital signal processing (DSP) tools before storing, analysing or transmitting. This cleaning process is often referred to as noise reduction. The term noise cancellation has got phenomenal attention in recent years. The noise cancellation schemes [8] provide an improved quality of speech signal that helps in achieving a better performance. One of the objectives of this paper is to present in a common context, an overview of the latest noise reduction algorithm with varying parameters.

Many different applications have been invented and introduced in the communication field [1]. One of the most basic parameters which is needed in a communication is the transmitting signal. Noise, which is an unwanted disturbance, adds itself to the speech signal. 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.

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.

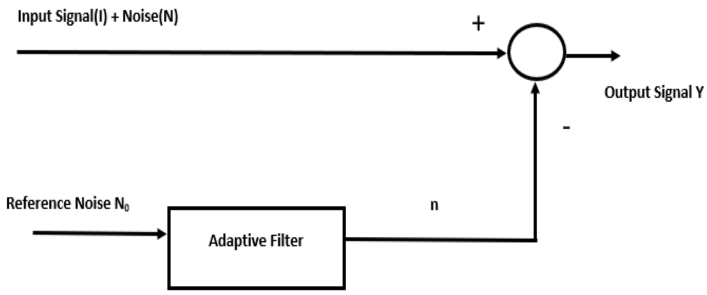

Figure 1 Basic block diagram of an ANC filter

Upon filtering using the adaptive techniques, an estimation of noise signal 'n' is obtained from the inputted reference noise 'N_0'. 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 generally depicted as d (n); desired signal, as seen in the Fig.1 [2].

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

An adaptive filter consists of two discrete parts: a digital filter with adjustable coefficients, and an adaptive algorithm which is used to adjust or modify the coefficients of the filter.

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 to cancel the noise and will be varying and comparing their parameters in de-noising the signals [3].

[1] Shubra et al., has stated through her research that though RLS provides better results but, LMS is a better option to go for as it has simpler calculations and stability. Proakis, has mentioned the different noise filtering methods available in the signal processing [2]. Mendiratta et al., has discussed in detail about Adaptive noise cancelling for audio signals using Least mean square (LMS) algorithm with successful results [3]. Panda et al., had done a thorough analysis on the real time noise cancellation using adaptive filters and has efficiently filtered out the noise from the speech signal by applying LMS technique [4].

II. Adaptive Noise Cancellation Algorithms

Adaptive noise cancellation algorithms have a diverse range of applications ranging from hearing aids to the large systems used in vehicles such as cars or airplanes. Mainly, Least Mean Square (LMS) algorithm, Normalized Least Mean Square (NLMS) algorithm and Recursive Least Square (RLS) algorithm are used for speech enhancement.

An adaptive filter is a time-varying filter, since their parameters are continually changing in order to meet certain performance requirements. Refer to Figure 2, where x(n) denotes the input signal, n is the noise added to the signal, n_0 is the reference noise, d(n) is the desired signal, e(n) is the error signal which is defined as the difference between the desired d(n) and filter output y(n). This error signal is used as a feedback to the adaptation algorithm in order to determine the proper updating of the coefficients of the filter [5].

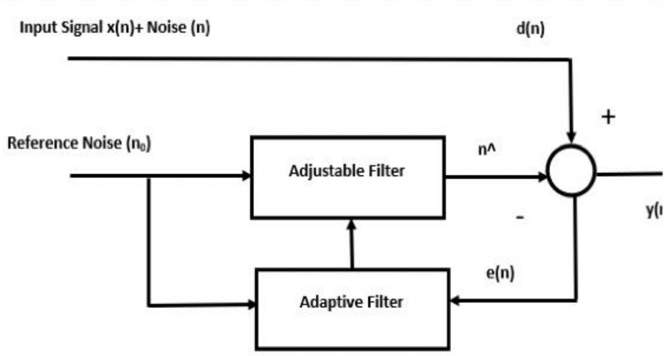

Figure 2 Block Diagram of an Adaptive Noise Cancellation System

As shown in Fig.2, an adaptive noise canceller (ANC) has two inputs-

i. Primary input
ii. Reference input

The primary input receives a signal x(n) that is degraded by the presence of noise (n). The reference noise n_0 is passed through a filter to produce an output nˆ that is a close replica of the primary input noise. This noise estimate is then subtracted from the corrupted signal to produce an estimate of the output signal y(n) [6,11].

III. LEAST MEAN SQUARE

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 and low computational complexity [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 FIR 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 [12]. 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 as that of the RLS algorithm; as a result, the calculations are unproblematic.

The working of the LMS filter is mainly divided into two parts,

i. Filtering Process, in which the output of the filter and the error signal is determined by comparing the output signal to the desired signal
ii. Adaptive Process, in which the weights of the filter are adjusted based on the obtained error signal [7].

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.

In general, the step-size should be large in the early adaptation, and have its value progressively reduced as the algorithm approaches steady-state. Therefore, μ is determined to be optimal by test experiments using MATLAB coding with pre-defined conditions.

After the calculation of the output of the filter, the error signal is calculated after which the weights are readjusted and the output is calculated repeatedly till the error signal is minimized.

IV. 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 distorting the original signal.

Algorithms like RLS, LMS, NLMS can be incorporated to free the signal from the added noises. To conduct this, a procedure needs to be followed. Each algorithm has its own mathematical calculations that have to be followed. The general procedure has been elucidated below.

To de-noise the distorted signal, the following steps have been followed:

Step 1: Generate a particular frequency desired signal using MATLAB.

Step 2: Add a random noise to the generated signal and plot their respective graphs.

Step 3: Choose between LMS, NLMS and RLS adaptive algorithm for active noise cancellation of the corrupted Speech signal.

Step 4: Choose the variables which are associated with the adaptive noise cancellation filters such as filter length and step size.

Step 5: Apply the algorithm and plot the denoised signal obtained as the output of the adaptive noise cancellation filter.

Step 6: Repeat the same procedure by varying the parameters such as filter length and step size.

Step 7: Compare the obtained results

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

V. 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, while the filter order is kept same.

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 was set to M = 20. The parameter μ(Step-size) is varied. Outputs are obtained by varying step size i.e. μ = 0.001 to μ = 0.01. System reaches steady state faster when the step size is larger.

Below shown are the test results, keeping the filter order constant and varying the step size. Fig 3 shows the results of the de-noised signal when the value of μ is 0.001whereas, Fig 4 shows the results of the de-noised signal when the value of μ is 0.01. Also, the frequency of the signal is chosen to be 220 Hz in both the cases.

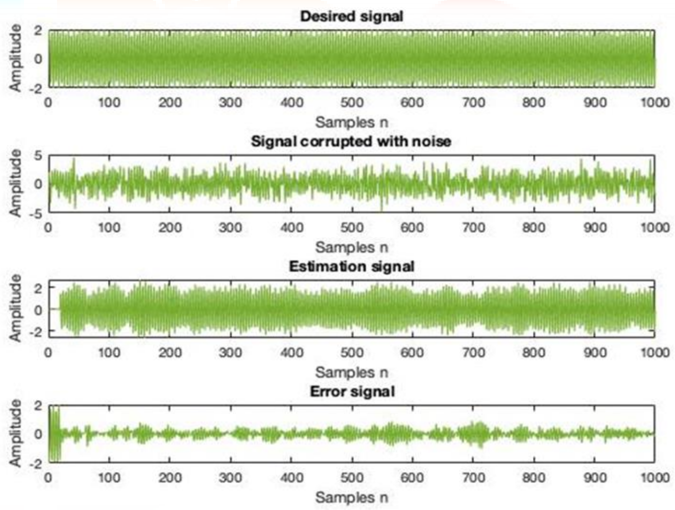

Figure 3 Results obtained with μ = 0.001

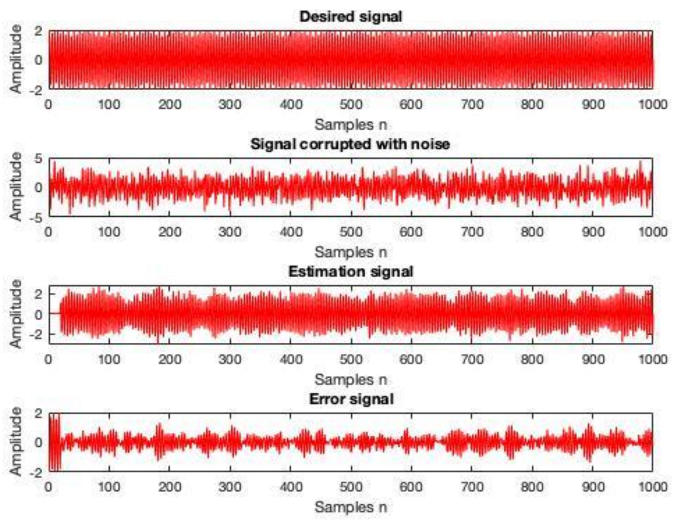

Figure 4: Results obtained with μ = 0.01

After test experiments with gradual reduction of step size parameter, 0.001 is the best choice since it is small enough but efficient to meet the design requirements to get the optimal convergence.

VI. Conclusion

When the information signal travels in the free environment, it gets corrupted by the noise present in it. Removing this noise emerges 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. The very first conclusion which arises from the above analysis for the most popular LMS algorithm presented in this paper is that the LMS algorithm is the best fit for all the applications. Different applications present different challenges for the adaptation of the algorithm, thus requiring different step-size update. In this paper, the speech enhancement approach is primarily based on the adaptive techniques. It may be visible from the experimental outcomes that the proposed approach correctly reduces the noise from the selected signal. This approach makes the speech signal clean and audible. Results are also shown in the previous section after variation in the step size parameter which is implemented using MATLAB environment.

VII. Acknowledgment

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