Acoustic Echo Cancellation

Digital Signal Analysis and Applications

Subhadeep Dash,
Computer Science and Engineering-201601021,
Indian Institute of Information Technology Sri City,
IIITS,
Sri City, India
Email_id: subhadeep.d16@iiits.in

***Abstract*—One of the major issues involved with communication is acoustic echo, which is actually a delayed version of sound reflected back to the source of sound resulting in hampered communication. The foremost thought that storms our brain when we discuss about acoustic echo cancellation is adaptive filtering. Adaptive filtering techniques are used in wide range of applications, including echo cancellation and adaptive equalization. In this paper, we focus on utilization of adaptive algorithms, most primarily LMS and NLMS, to remove the unnecessary echo and hence enhancing the communication quality.**

Keywords—Acoustic Echo Cancellation, Adaptive filter algorithms, Least Mean Square(LMS), Normalized Least Mean Square(NLMS)

I. INTRODUCTION

Acoustic echo cancellation is prevalent in today's telecommunication systems. Acoustic echo occurs when an audio signal is reverberated in a real environment resulting in intended signal interfering with the attenuated, time-delayed version of the same signal [1]. The chief necessity of the emerging sophisticated world is effective transmission of voice signals through long distances. To get rid of the echo from the signal and recover the original speech, we make use of adaptive filter algorithms.

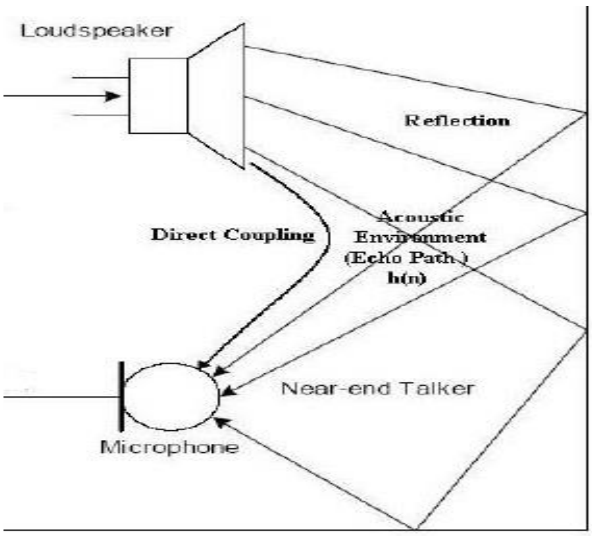

Figure 1.1: Acoustic Echo Generation

Filters are the basic components in telecommunication system and signal processing. They are widely used in noise reduction, channel equalization and audio processing. A vital block of Acoustic Echo Cancellation is Adaptive Filter [2] which reduces noise without any previous knowledge of signal or noise. Adaptive filters are dynamic filters which iteratively modify their characteristics in order to obtain the desired output. They are used in situation where speech and noise signals are random in nature.

The adaptive filter calculates the difference between the desired signal d(n) and the output of the adaptive filter y(n) and then this resultant error signal e(n) is fed back into the adaptive filter. The resultant coefficients are changed dynamically so that the optimal output of the adapter filter is equal in value to the unwanted echoed signal. In case the optimal output of the adaptive filter y(n) is equal to the desired signal d(n), the error signal e(n) becomes zero. This results in complete cancellation of the unwanted echo in the signal. Thus, the far user would not hear any of their echoed speech returned to them.

$$e[n] = d[n] – y[n] \qquad (1)$$

where e is the error signal which is equal to difference of desired signal d[n] and filtered output y[n].

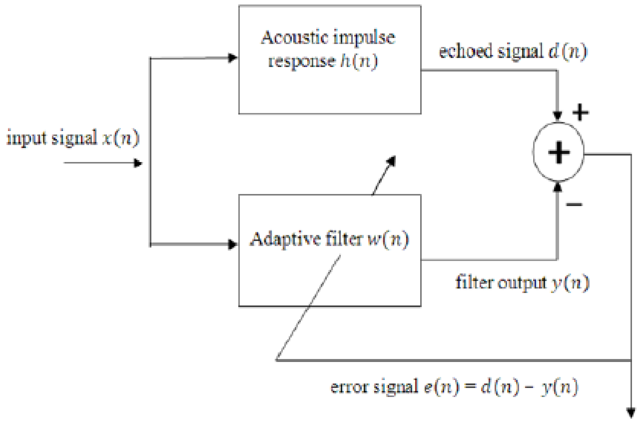

Figure 1.2: Adaptive Echo Cancellation System

Adaptive Filters consists of two parts. The functionality of the first part is to filter the echoed signal. An adaptive filter algorithm constitutes the second part whose purpose is to update the filter per iteration.

Adaptive Filtering algorithms consist of algorithms of mainly two kinds of approaches. The first one being the Stochastic Gradient Approach and the other one being the Least Square Estimation Approach. The most popular adaptive filtering algorithms are Least Mean Square(LMS), Normalized Least Mean Square(NLMS) and the Recursive Least Square(RLS) algorithms. The Least Mean Square and Normalized Least Mean Square algorithms come under the Stochastic Gradient Approach whereas Recursive Least Square comes under the Least Square Estimation Approach. In this paper, we shall discuss the utilization of Least Mean Square and Normalized Least Mean Square algorithms to eliminate acoustic echo caused while transmitting sound signals.

II. ADAPTIVE FILTER ALGORITHMS

A. Least Mean Square(LMS) Algorithm

The Least Mean Square(LMS) algorithm was initially developed by Widrow and Hoff in 1959 through their studies in pattern recognition. It is now considered to be one of the most widely used algorithms in adaptive filtering. The basic idea behind LMS filter is to approach the optimum filter weights by updating the filter weights in such a way so that it converges to the optimum filter weight [3]. This adaptive filter algorithm comes under the Stochastic Gradient Descent method. It is because the filter in it is only adapted based on the error at the current time. This adaptive filter is used to mimic a desired filter by determining the filter coefficients that relate to producing the least mean squares of the error signal, which is actually the difference between the desired signal and signal obtained as output by the filter.

The algorithm starts with the assumption of small weights (mostly zero) and at each iterative step, the weights of the filter are updated by finding the gradient of the mean square error. If the Mean Squared Error (MSE) gradient is positive, the error would keep on increasing positively. If the same weight is used in case of further iterations, we will obviously need to reduce the weights. The same applies if the gradient is negative. But in case of negative gradient, we need to increase the weights, so as to stop it from decreasing negatively. Considering these situations, the basic weight update equation is:

$$w_{n+1} = w_n - \mu\Delta\varepsilon[n] \tag{2}$$

where ε represents the mean-square error, μ represents the step size and w_n, w_{n+1} represent the adaptive filter weights. The negative sign indicates that the change in weights should be in a direction opposite to that of the gradient slope.

The LMS algorithm has a complexity of 2N + 1. Of this N is for obtaining the output signal, one for the scalar multiplication of $2\mu\Delta\varepsilon[n]$ and N for the scalar by vector multiplication. This algorithm is widely used due to its low complexity nature than other algorithms. The primary disadvantage of this algorithm is that it suffers from slow and data dependent convergence behavior.

B. Normalized Least Mean Square(NLMS) Algorithm

The main drawback of the “pure” LMS algorithm is that it has a finite step size(μ). This leads to the fact that it is sensitive to the scaling of its input. This makes it very difficult to choose a suitable step size that could guarantee the stability of the algorithm. The Normalized Least Mean Square(NLMS) Algorithm is a variant of LMS algorithm that solves this problem by normalizing with the power of the input [5]. The step size is updated at each iteration.

The NLMS has a computational complexity of 3N+1. The extra N when compared to LMS is due to the update of step size or convergence factor μ in every iteration. Though it has higher computational complexity then LMS, it is more stable when compared to LMS.

C. Recursive Least Square(RLS) Algorithm

The Recursive Least Square(RLS) adaptive filter is an algorithm which recursively determines the filter coefficients that minimize a weighted linear least square cost function relating to the input signal. The RLS algorithm is known for its high stability, excellent performance when compared to Least Mean Squares algorithms. Their excellent performance when working in time varying environments can only be achieved at the cost of an increased computational complexity.

Though the RLS algorithm is highly stable, it has a complexity of $4N^2$ which makes it almost unfeasible for real-time applications.

Since we are discussing on implementation of real-time echo cancellation, we prefer to skip this algorithm in our paper utilizing the other two popular algorithms mainly LMS and NLMS to cancel echo in our input signals. The table below draws the differences between all the three algorithms discussed so far [7].

Table 1.1: Performance Comparison of Adaptive Algorithms

S. No.	Algorithms	MSE	Complexity	Stability
1.	LMS	$1.5*10^{-2}$	2N+1	Less Stable
2.	NLMS	$9.0*10^{-3}$	3N+1	Stable
3.	RLS	$6.2*10^{-3}$	$4N^2$	High Stable

III. IMPLEMENTATION OF LEAST MEAN SQUARE ALGORITHMS

A. Least Mean Square(LMS) Algorithm

The implementation of Least Mean Square Algorithm involves three stages for each iteration [6].

1. The output from the adaptive filter is obtained as stated in (3) i.e. by dot product of filter tap weights with input signal.

$$y[n] = w^T[n].x[n] \tag{3}$$

where $y[n]$ is the obtained output signal, $w^T[n]$ is the transpose of the filter tap weights vector and $x[n]$ is the input signal on which LMS is to be applied.

2. The value of the error signal (e) is calculated as discussed earlier.

$$e[n] = d[n] - y[n] \quad (4)$$

where e is the error signal and d is the desired signal.

3. The tap weights of the filter are updated in preparation of the next iteration.

$$w_{n+1} = w_n + 2*\mu*e[n]*x[n] \quad (5)$$

where w_n, w_{n+1} are the filter tap weights.

B. *Normalized Least Mean Square(NLMS) Algorithm*

The implementation of Normalized Least Mean Square Algorithm involves four stages for each iteration [7].

1. The output from the adaptive filter is obtained as stated in (6) i.e. by dot product of filter tap weights with input signal.

$$y[n] = w^T[n].x[n] \quad (6)$$

where y[n] is the obtained output signal, $w^T[n]$ is the transpose of the filter tap weights vector and x[n] is the input signal on which LMS is to be applied.

2. The value of the error signal (e) is calculated as discussed earlier.

$$e[n] = d[n] - y[n] \quad (7)$$

where e is the error signal and d is the desired signal.

3. As per the original algorithm μ should be the reciprocal of the dot product of the transpose of the input signal with itself.

$$\mu_{n+1} = 1/(x^T[n]*x[n]) \quad (8)$$

But this equation seems to dissatisfy when x[n] = 0, then μ tends to take a value of infinity which is not practically possible. So, we modify the above equation as follows.

$$\mu_{n+1} = 1/(x^T[n]*x[n] + \alpha) \quad (9)$$

where α is some considerably small value.

4. The tap weights of the filter are updated in preparation of the next iteration.

$$w_{n+1} = w_n + \mu*e[n]*x[n] \quad (10)$$

where w_n, w_{n+1} are the filter tap weights.

IV. Results

A. *Least Mean Square Algorithm*

After the LMS Algorithm is successfully implemented in MATLAB, the results can be depicted through the following figures.

The first graph shows the signal we have input, i.e. it is the desired signal, the clear signal which we intend to obtain after the application of adaptive filters on the echoed signal.

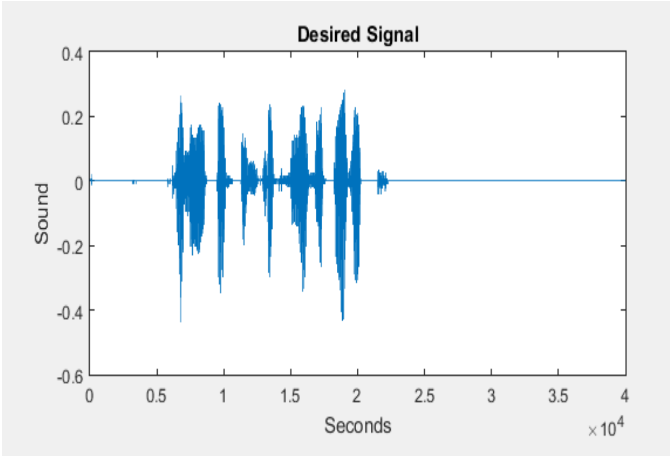

Figure 4.1: Desired Signal for LMS

The second plot shows the echoed signal which we have created manually interfering the signal with the delayed version of the same signal having lesser frequency.

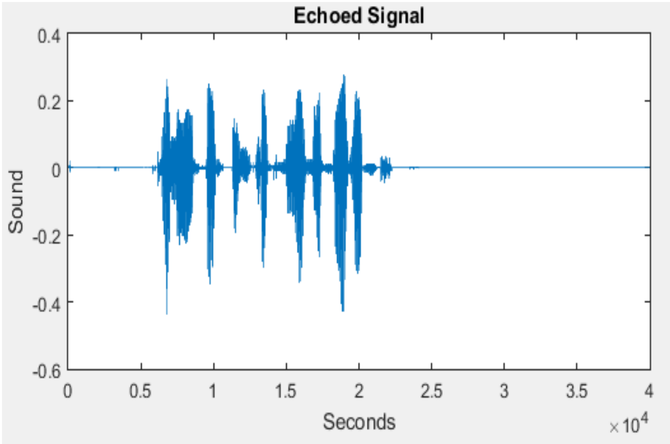

Figure 4.2: Echoed Signal as Input for LMS

The third plot depicts the signal obtained by passing the echoed signal through the adaptive filters. This is our signal obtained after the cancellation of echo.

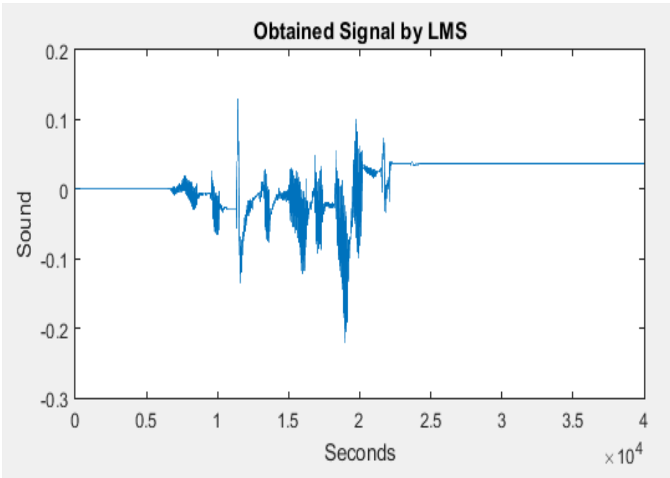

Figure 4.3: Output Signal as obtained by LMS

The fourth plot shows the deviation of the obtained output signal from the desired signal.

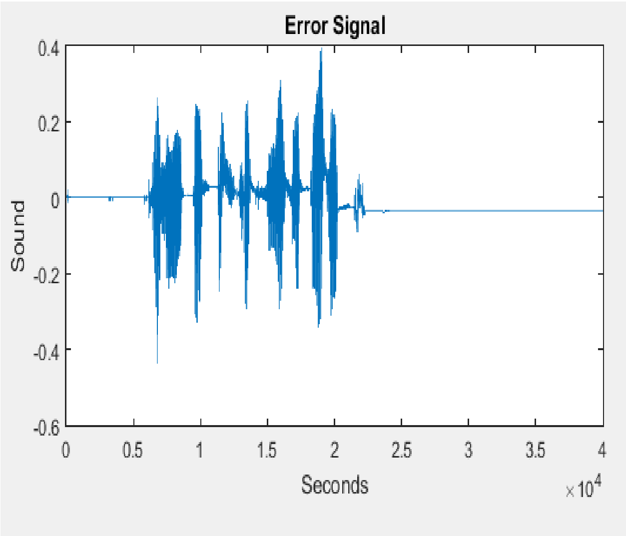

Figure 4.4: Error Signal in LMS

B. Normalized Least Mean Square(NLMS) algorithm

After the NLMS Algorithm is successfully implemented in MATLAB, the results can be depicted through the following figures.

The first graph shows the signal we have input, i.e. it is the desired signal, the clear signal which we intend to obtain after the application of adaptive filters on the echoed signal.

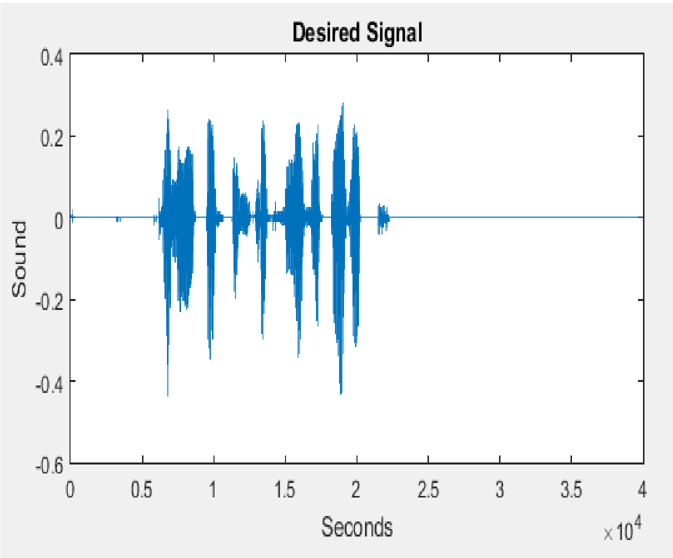

Figure 4.5: Desired Signal for NLMS

The second plot shows the echoed signal which we have created manually interfering the signal with the delayed version of the same signal having lesser frequency.

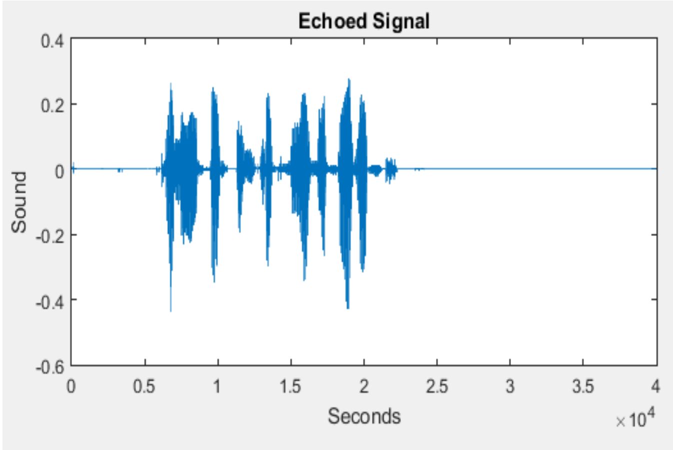

Figure 4.6: Echoed signal given as input in case of NLMS

The third plot depicts the signal obtained by passing the echoed signal through the adaptive filters. This is our signal obtained after the cancellation of echo.

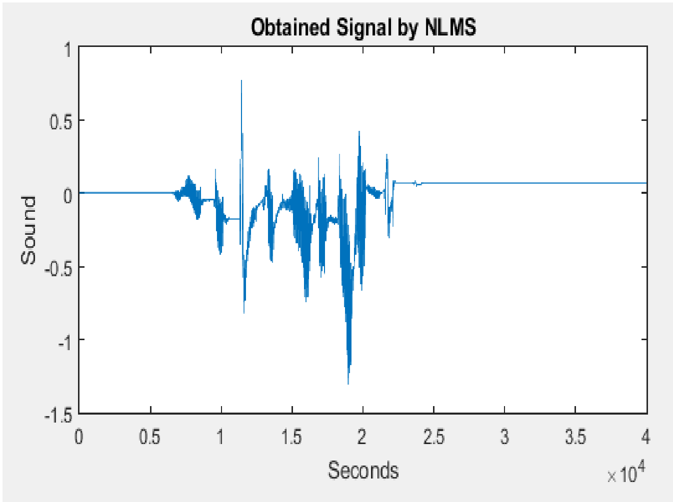

Figure 4.7: Obtained Signal as obtained by NLMS

The fourth plot shows the deviation of the obtained signal from the desired signal.

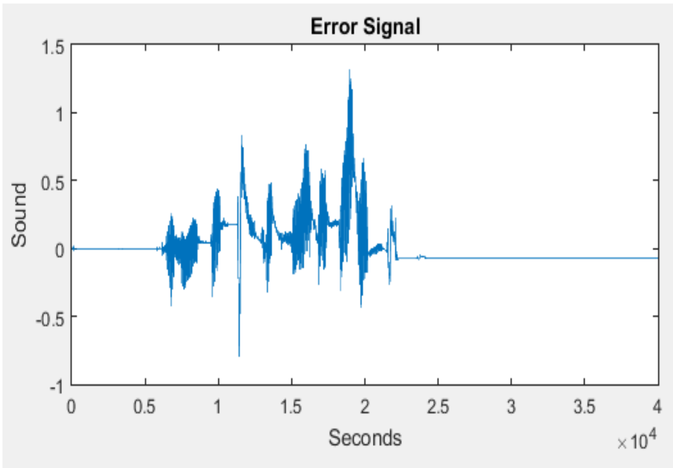

Figure 4.8: Error Signal as obtained by NLMS

V. CONCLUSION

We observe that echo is effectively cancelled by utilization of the adaptive filter algorithms namely LMS and NLMS in O(n) time. Both have lower computational complexity and have successfully cancelled echo in linear time.

A. *List of Figures and Tables*

a) List of Figures

b) List of Tables

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