

ADAPTIVE ALGORITHMS FOR ACOUSTIC ECHO CANCELLATION IN SPEECH PROCESSING

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ABSTRACT

Adaptive filtering constitutes one of the core technologies in digital signal processing and finds numerous application areas in science as well as in industry. Adaptive filtering techniques are used in a wide range of applications, including echo cancellation, adaptive equalization, adaptive noise cancellation, and adaptive beamforming. Acoustic echo cancellation is a common occurrence in today's telecommunication systems. The signal interference caused by acoustic echo is distracting to users and causes a reduction in the quality of the communication. This paper focuses on the use of LMS and NLMS algorithms to reduce this unwanted echo, thus increasing communication quality.

Keywords: Adaptive filters, Adaptive algorithms, Acoustic echo cancellation.

1. INTRODUCTION

Acoustic echo occurs when an audio signal is reverberated in a real environment, resulting in the original intended signal plus attenuated, time-delayed images of the signal [1]. This paper will focus on the occurrence of acoustic echo in telecommunication systems.

Adaptive filters are dynamic filters which iteratively alter their characteristics in order to achieve an optimal desired output. An adaptive filter algorithmically alters its parameters in order to minimise a function of the difference between the desired output $d(n)$ and its actual output $y(n)$. This function is known as the cost function of the adaptive algorithm. Figure 1 shows a block diagram of the adaptive echo cancellation system. Here the filter $H(n)$ represents the impulse response of the acoustic environment, $W(n)$ represents the adaptive filter used to cancel the echo signal. The adaptive filter aims to equate its output $y(n)$ to the desired output $d(n)$ (the signal reverberated within the acoustic environment). At each iteration the error signal, $e(n) = d(n) - y(n)$, is fed back into the filter, where the filter characteristics are altered accordingly.

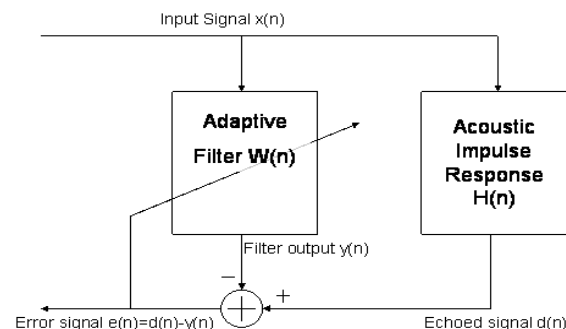


Figure 1. Adaptive echo Cancellation system

The aim of an adaptive filter is to calculate the difference between the desired signal and the adaptive filter output, $e(n)$. This error signal is fed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize a function of this difference, known as the cost function. In the case of acoustic echo cancellation, the optimal output of the adaptive filter is equal in value to the unwanted echoed signal. When the adaptive filter output is equal to desired signal the error signal goes to zero. In this situation the echoed signal would be completely cancelled and the far user would not hear any of their original speech returned to them.

2. LEAST MEAN SQUARE (LMS) ALGORITHM

The Least Mean Square (LMS) algorithm was first developed by Widrow and Hoff in 1959 through their studies of pattern recognition. From there it has become one of the most widely used algorithms in adaptive filtering. The

LMS algorithm is a type of adaptive filter known as stochastic gradient-based algorithms as it utilises the gradient vector of the filter tap weights to converge on the optimal wiener solution [2-4]. It is well known and widely used due to its computational simplicity. It is this simplicity that has made it the benchmark against which all other adaptive filtering algorithms are judged.

With each iteration of the LMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu e(n)\mathbf{x}(n) \quad (1)$$

Here $\mathbf{x}(n)$ is the input vector of time delayed input values, $\mathbf{x}(n) = [x(n) \ x(n-1) \ x(n-2) \ \dots \ x(n-N+1)]^T$. The vector $\mathbf{w}(n) = [w_0(n) \ w_1(n) \ w_2(n) \ \dots \ w_{N-1}(n)]^T$ represents the coefficients of the adaptive FIR filter tap weight vector at time n . 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 [5-8].

2.1. Implementation of the LMS Algorithm

Each iteration of the LMS algorithm requires 3 distinct steps in this order:

1. The output of the FIR filter, $y(n)$ is calculated using equation 2.

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n) \quad (2)$$

2. The value of the error estimation is calculated using equation 3.

$$e(n) = d(n) - y(n) \quad (3)$$

3. The tap weights of the FIR vector are updated in preparation for the next iteration, by equation 4.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu e(n)\mathbf{x}(n) \quad (4)$$

The main reason for the LMS algorithms popularity in adaptive filtering is its computational simplicity, making it easier to implement than all other commonly used adaptive algorithms. For each iteration the LMS algorithm requires $2N$ additions and $2N+1$ multiplications (N for calculating the output, $y(n)$, one for $2\mu e(n)$ and an additional N for the scalar by vector multiplication[9]).

3. NORMALISED LEAST MEAN SQUARE (NLMS) ALGORITHM

One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. Even if we assume the only signal to be input to the adaptive echo cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance[10-12].

The normalised least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by calculating maximum step size value. Step size value is calculated by using the following formula.

$$\text{Step size} = 1 / \text{dot product}(\text{input vector}, \text{input vector})$$

This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector $\mathbf{x}(n)$. This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix, \mathbf{R} [13-15].

$$\begin{aligned} \text{tr}[\mathbf{R}] &= \sum_{i=0}^{N-1} E[x^2(n-i)] \\ &= E\left[\sum_{i=0}^{N-1} x^2(n-i)\right] \end{aligned} \quad (5)$$

The recursion formula for the NLMS algorithm is stated in equation 6.

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{1}{\mathbf{x}^T(n)\mathbf{x}(n)} e(n)\mathbf{x}(n) \quad (6)$$

3.1. Implementation of the NLMS algorithm

The NLMS algorithm has been implemented in Matlab. As the step size parameter is chosen based on the current input values, the NLMS algorithm shows far greater stability with unknown signals. This combined with good

convergence speed and relative computational simplicity make the NLMS algorithm ideal for the real time adaptive echo cancellation system [16].

As the NLMS is an extension of the standard LMS algorithm, the NLMS algorithms practical implementation is very similar to that of the LMS algorithm. Each iteration of the NLMS algorithm requires these steps in the following order.

1. The output of the adaptive filter is calculated.

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n) \quad (7)$$

2. An error signal is calculated as the difference between the desired signal and the filter output.

$$e(n) = d(n) - y(n) \quad (8)$$

3. The step size value for the input vector is calculated.

$$\mu(n) = \frac{1}{\mathbf{x}^T(n)\mathbf{x}(n)} \quad (9)$$

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

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \mu(n)e(n)\mathbf{x}(n) \quad (10)$$

Each iteration of the NLMS algorithm requires $3N+1$ multiplications, this is only N more than the standard LMS algorithm. This is an acceptable increase considering the gains in stability and echo attenuation achieved.

4. RESULTS OF LMS ALGORITHM

The LMS algorithm was simulated using Matlab. Figure 2 shows the input speech signal which is collected from the computer system through microphone. Figure 3 shows the desired echo signal derived from the input signal. Figure 4 shows the adaptive filter output which will reduce the echo signal from the input signal. Figure 5 shows the mean square error signal calculated from the filter output signal. Figure 6 shows the attenuation which is derived from the division of echo signal to the error signal.

The adaptive filter is a 1025th order FIR filter. The step size was set to 0.02. The MSE shows that as the algorithm progresses the average value of the cost function decreases.

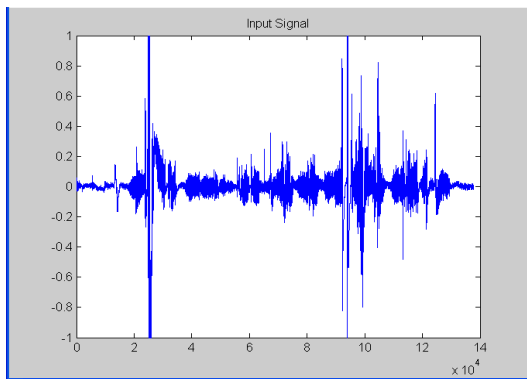


Figure 2. Input Signal

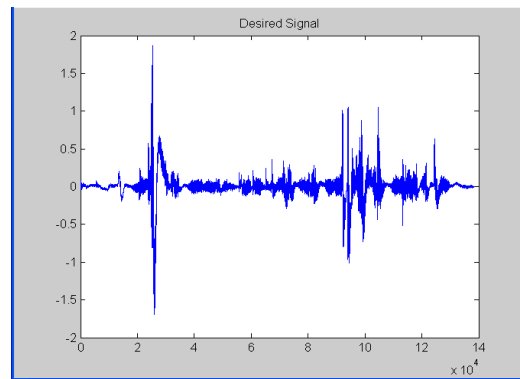


Figure 3. Desired Signal

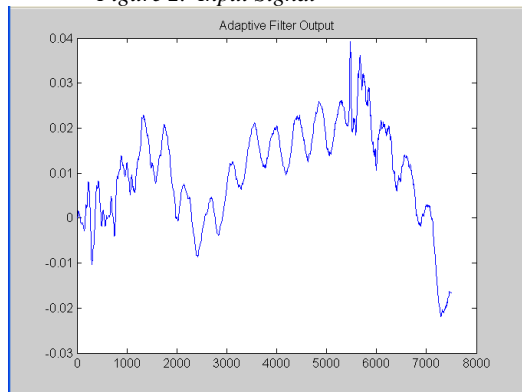


Figure 4. Adaptive Filter Output

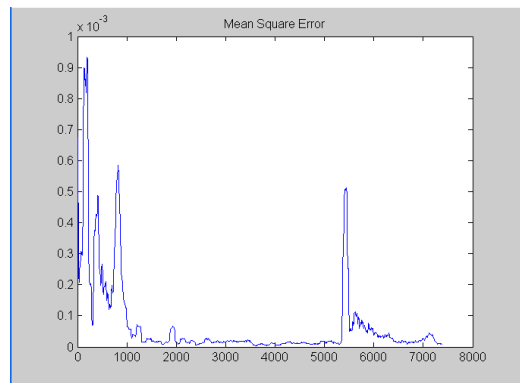


Figure 5. Mean Square Error

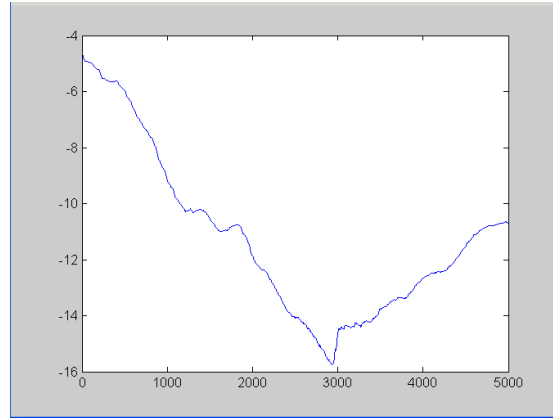


Figure 6. Attenuation

5. RESULTS OF NLMS ALGORITHM

The NLMS algorithm was simulated using Matlab. Figure 7 shows the input signal. Figure 8 shows the desired signal. Figure 9 shows the adaptive filter output. Figure 10 shows the mean square error. Figure 11 shows the attenuation.

The adaptive filter is a 1025th order FIR filter. The step size was set to 0.1.

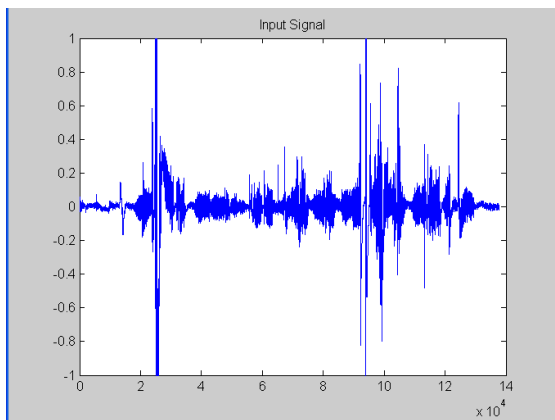


Figure 7. Input Signal

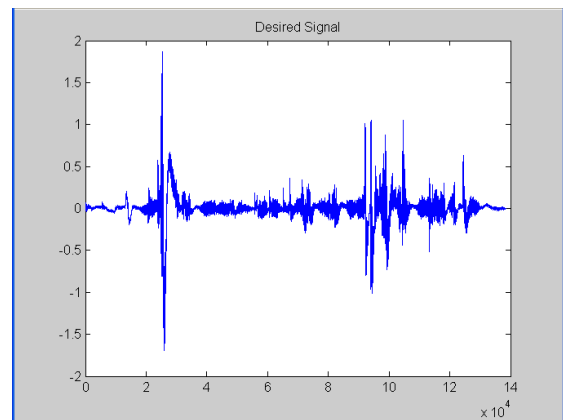


Figure 8. Desired Signal

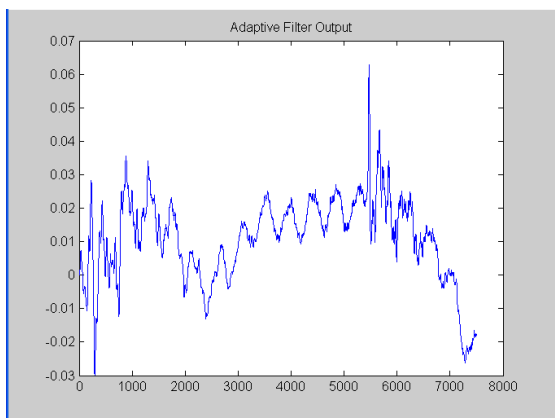


Figure 9. Adaptive Filter Output

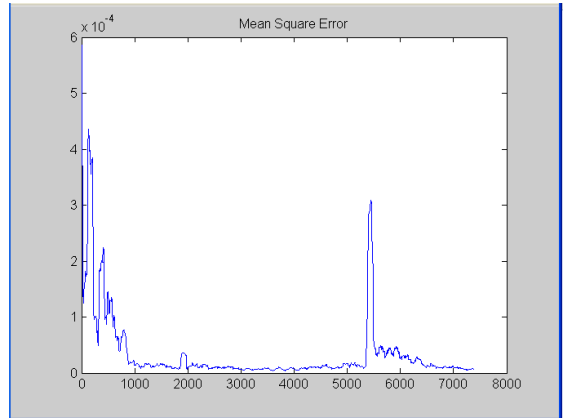


Figure 10. Mean Square Error

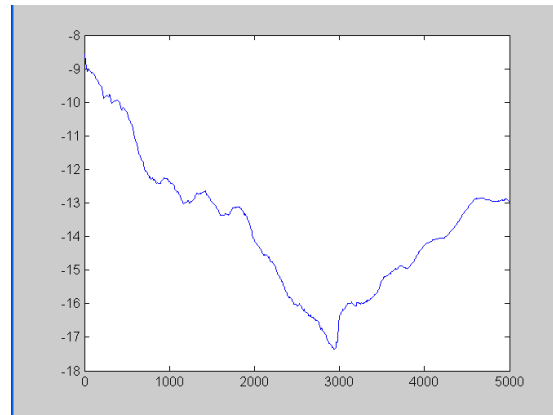


Figure 11. Attenuation

NLMS algorithm is having the advantage over the LMS algorithm incase of Mean square error and Average attenuation and its summary of the performance is presented in Table 1.

Table 1. Summary of adaptive algorithms performance

ALGORITHM	ITERATIONS	FILTER ORDER	MEAN SQUARE ERROR	AVERAGE ATTENUATION	COMPUTATIONS
LMS	7500	1025	0.001	-11.2435	2N+1
NLMS	7500	1025	0.0004	-13.6812	3N+1

6. CONCLUSIONS

Because of its simplicity, the LMS algorithm is the most popular adaptive algorithm. However, the LMS algorithm suffers from slow and data dependent convergence behavior.

The NLMS algorithm, an equally simple, but more robust variant of the LMS algorithm, exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good properties the NLMS has been largely used in real-time applications.

7. REFERENCES

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