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A Novel Approach for the Adaptive Filtering in Echo Cancellation

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ABSTRACT

Adaptive filter constitutes one of the core technologies in adaptive filter signal processing and finds numerous application areas in science as well as an industry. Adaptive filtering techniques are used in wide ranges of applications, including the echo cancellation, adaptive equalization, adaptive noise/echo cancellation, and filter adaptive beam forming.Acoustic echo cancellation is a common occurrence in today's telecommunication systems. The present work focuses on the conventional adaptive algorithms and modified LMS algorithm to reduce the unwanted echo, thus in the improving communication quality. Thus the number of multiplications required is very large because of which the RLS algorithm is too costly to important. Simulation results are presented to support the analysis and to compare the performance of the modified algorithm with the other conventional adaptive algorithms. They show that LMS, NLMS and RLS algorithms. An attempt has been made to examine the adaptive filtering techniques as they apply to acoustic echo cancellation, to simulate these of adaptive filtering algorithms using MATLAB and to compare the performance of these adaptive filtering algorithms as they are applied to the acoustic echo cancellation application.

Keywords: Adaptive filter, LMS algorithm, NLMS algorithm, RLS algorithm, Matlab.

1. INTRODUCTION

The development of echo reduction began in the late 1950s, and continues today as represented integrated landline and wireless cellular networks put additional requirement on the performance of echo cancellers. Echo is the repetition of a waveform due to reflection points where the characteristics of the medium through which the wave propagates changes. Echo is usefully purposes employed in radar and sonar for detection and exploration .In telecommunication, echo/voice canceller is an important part can degrade the quality of service, and echo cancellation of communication systems [1].

Echo is a delayed and distorted version of an original sound or electrical signal which is reflected back to the source. The reflected wave arrives after a very short time of direct and indirect echo/voice sound, it is considered as a spectral distortion or reverberation. However, when the reflected wave a few tens of milliseconds arrives after the direct and indirect sound, it is heard as a distinct echo. In data communication, echo can incur a big data transmit error. In applications like hands- free telecommunications, the echo, in extreme conditions, can make the conversations, the echo in extreme conditions, can make the conversation impossible. The echo has been a big issue in communication networks.

Acoustic echo cancellation is important for audio teleconferencing when simultaneous communication (or full duplex transmission) of speech is necessary. In acoustic and hybrid echo/voice cancellation, a measured DSP microphone signal d(n) contains two signals:-

- The near – end speech signal v(n)
- The far – end echoed speech signals d(n)

The goal is to remove the far – end echoed speech signal from the microphone signal so that only the near – end speech signal is transmitted.

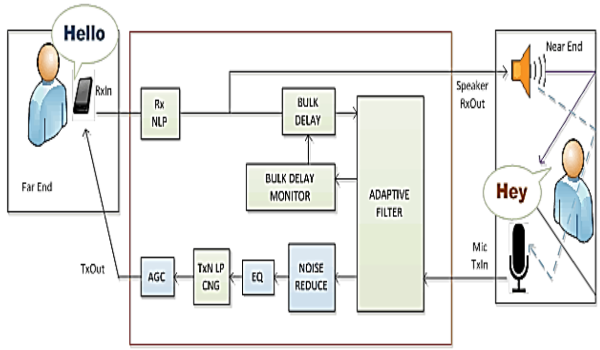


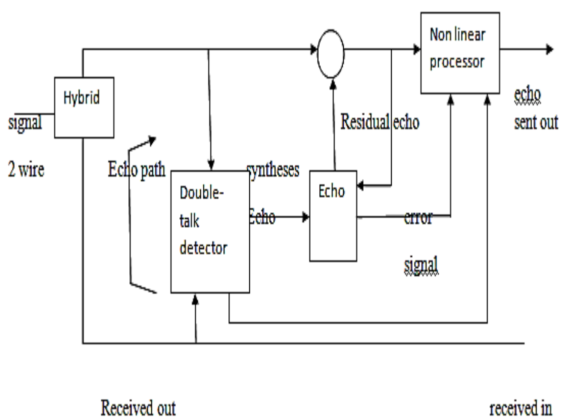
Fig.1 Echo cancellation

II. THEORETICAL REVIEW

Echo cancellation is one of the most widely used digital signal processing devices in the world because each telephone call requires a pair of echo cancellers. Basically, a transversal filter, which is adaptively modeling the echo path impulse responses, generates an estimate of the echo, with an echo estimate is created at the right time to cancel the actual echo. The common problems faced by echo cancellation are the convergence time and the degree of cancellation [5]. Convergence time is the time taken to reach an acceptable level of steady state residual echo. Cancellation is the amounts of the echo cancelled, measured in Echo Return Loss Enhancement (ERLE).Echo cancellation consists of three important blocks:

1. **Non-linear processor**, to remove unconcealed residual echoes.
2. **Adaptive Echo canceller,** to adapt the echo path impulse responses and synthesize replica echoes.
3. **Double talk Detector**, to detect double talk and consequently stop the echo canceller adaptation.

The echo cancellation block diagram is shown in fig 2. The adaptive echo canceller is the engine of echo cancellation, with its performance being mainly determined by the adaptive algorithm. The Normalized least mean square (NLMS) algorithm is the most popular algorithm implemented in echo cancellation. It is a well-studied algorithm, simple to implements and it is guarantees convergence. The non-linear processor is used to remove residual echoes by injecting comfort noise [13]. A low level of comfort noise is usually added during the silence periods as an audio-psychological comfort effect for the listener. The double-talk detector is used to sense the near-end and far end signal levels.



Lifetime

The main role of adaptive filtering is the development of a filter capable of adjusting to the statistics of the signal. Usually, to the adaptive filter algorithm takes the form of a FIR filter, with an adaptive algorithm this is modifies the values of its coefficients.

The main configurations of an adaptive filter are the adaptive cancellation of noise and the adaptive cancellation of echo.
A communication system is a system that allows two or more persons, in communication or perhaps, one person and one machine, to interact with speech over an artificial channel, such as pictures, video, documents, or any other type of media,

but we will only discuss the audio component. The most commonly used system in this category is of course the standard telephone system also called the public switched telephone network (PSTN).

III ALGORITHMS

These assumptions are often for the ease of mathematical analysis, but do not take into account of the broader problems of signals with non-Gaussian statistics. In the digital communication systems, efficient bandwidth utilization is economically important to maximizing profits, while the same time maintaining performance and reliability. More importantly, the adaptive filter solution has to be relatively simple, which often leads to the use of the conventional Least Mean Square (LMS) algorithm. However, the performance of the LMS algorithm is often sub-optimal and the convergence rate is small. This, therefore, provides the motivation to explore and study variable step size LMS adaptive algorithms for various applications [18].

The Wiener Filter

These are a class of linear optimum discrete time filters known collectively as Wiener filters. Wiener filters are a special class of transversal Finite Impulse Response (FIR) filters that build upon the Mean Square Error (MSE) cost function to arrive at an optimal filter tap weight vectors, which reducing the MSE signals to a minimum[7]. Theory for a Wiener filter is formulated for general case of complex valued time series with filter specified in terms of its impulse response because baseband signal appears in complex from under many practical situations.

Mean Square Error Criterion

Fig 3 illustrates a linear filter with the aim of estimating the desired signal d(n) from input x(n). Assume that d(n) and x(n) are samples of infinite length, random processes. In 'optimum filter design', signal and noise are viewed as stochastic processes. The filter is based on minimization of the mean square value of the difference between the actual filter output and some desired output, as shown in fig.3

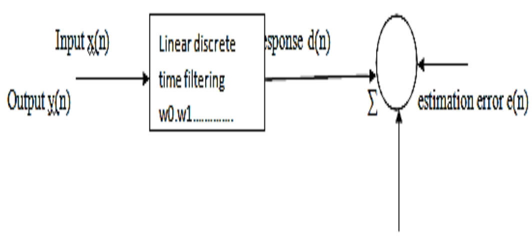


Fig:3 Wiener Filtering Scheme

The requirement is to make the estimation error as small as possible in some statistical sense by controlling impulse response coefficients.

$$w0,w1,\ldots\ldots\ldots\ldots w_{N-1}.$$

Two basic restrictions are:

1.The adaptive filter is linear, which is making to mathematical analysis easy to the handlings.

2. The filter is an FIR (There are adaptive filter symmetrical and odd ordered) filter.

Algorithms are discussed below:-

1. LMS (Least Mean Square) Algorithm
2. NLMS (Normalized LMS) Algorithm
3. RLS(Recursive Least Square) Algorithm

LMS Algorithm

The LMS (Least Mean Square) to the adaptive filter based algorithms. It is the most widely used adaptive filter algorithm for numerous applications, especially channel equalization and echo cancellation. The simplicity and robustness of the LMS updating equation are the most important features and leads to many successful developments of other gradient descent-based algorithms. The LMS (Least Mean Square)Algorithm is introduced as a way to recursively adjust the parameters of w(n) of a linear filter with the goal of minimizing the error between a given desired signal and the output of the linear filter. LMS is one of the many related algorithms which are appropriate for the task and whole family of algorithms have been developed which can address a variety of problem settings, signal of computational restrictions and minimization criteria [9]. This is derivation of the LMS algorithm as an instantaneous approximation to the steepest descent minimization of a cost function, which results in simple recursive scheme.

Normalized LMS (NLMS) Algorithm

Determining the upper bound step size is a problem for the variable step size algorithm if the input signal to the adaptive filter is non-stationary. The fastest convergence is achieved with the choice of step size as follows:

$$\mu_{max} = 1/ (\lambda_{max} + \lambda_{min})$$

However, experimental results have shown that the maximum step size in equation (4.1) does not always produce the stable and fast convergence, according to Kong (2/3)μ_{max} is a rule of thumb for LMS algorithm. To increase the convergences speed of NormalizedLMS (NLMS) algorithm is normalized by the input signal power.

The NLMS algorithm is always the favorable choice because of fast convergence speed. For non-stationary input the correction term applied to the estimated tap weight vector w(n) at the n-th iteration is normalized with respect to squared Euclidean norm of the tap input x(n) at the (n-1)th iteration,

$$w(n+1)=w(n)+\frac{\alpha}{\|x(n)\|^2}e(n)x(n)$$

Where $\| \ \|^2$ = Euclidean Norm

When the echo cancellation algorithm is input vector x(n) is small, instability may occur since we are trying to perform numerical division by small value of the Euclidean Norm.

However, this can be easily overcome by appending a positive constant to the denominator in the above equation such that

$$w(n+1)=w(n)+\frac{\alpha}{c+\|x(n)\|^2}e(n)x(n)$$

Where $c + \|x(n)\|^2$ is the normalization factor. With this, more than robust and reliable an implementation of the NLMS algorithm is obtained.

RLS Algorithm

The major advantage of the LMS algorithm lies in its low computational complexity. However, this is simplicity is slow convergence, especially when the Eigen values of the auto correlation matrix have a large spread, that that is when λ_{max} / λ_{min} >>1. The LMS algorithm has only a single adjustable parameter for controlling the convergence rate, namely, the step size parameter. Since this is limited for purposes of stability to be less that the upper bound, the modes corresponding to the Eigen values converge slowly. To obtain faster convergence, it is necessary to devise more complex algorithms, which involve additional parameters.

The equations for the implementation of the RLS algorithm are given below:-

- The filter output is the calculated using in the filter tap weights from the previous iteration and the current input vector.

$$y'_{n-1}(n) = w'^{T}(n-1)\ x(n)$$

- The gain vector is calculated using the following equation:-

$$k(n) = \frac{\lambda^{-1}\beta_{\lambda}^{'-1}(n-1)x(n)}{1+\lambda^{-1}x^{T}(n)\beta_{\lambda}^{'-1}(n-1)x(n)}$$

- The estimation error is calculated as

$$e'_{n-1}(n)= d(n) - y'_{n-1}(n)$$

- The filter tap weights can be calculated as

$$w'(n) = w'(n-1) + k(n)\ e'_{n-1}(n)$$

  This is using the Recursive Least Squares (RLS) algorithm to subtract noise from an input signal. The RLS adaptive filter algorithms uses the reference signal on the input port and the desired signal on the desired port to automatically match the filter response in the Noise Filter block. The filtered noise signal should be completely subtracted from the “signal + noise” adaptive signal, and the “Error Signal” should be contain only the original signal.

IV.SIMULATION RESULTS

We performed extensive simulations in MATLAB® in order to evaluate performance of LMS, NLMS and RLS algorithms. In start we have considered a signal which is the output of a system excited by a known signal and the system is in turn modeled so that its output resembles the original signal in an optimal way. It is used original signal MATLAB code. Fig.4 shows the original signal and fig.6 shows the echo signal recieved.

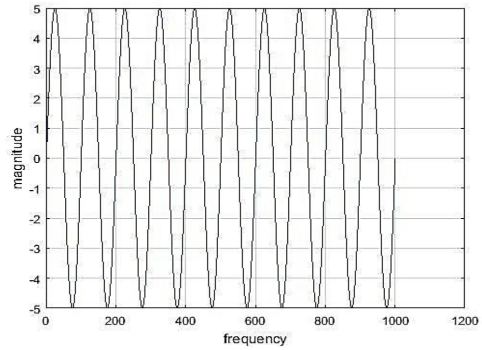


Fig.4: Original signal

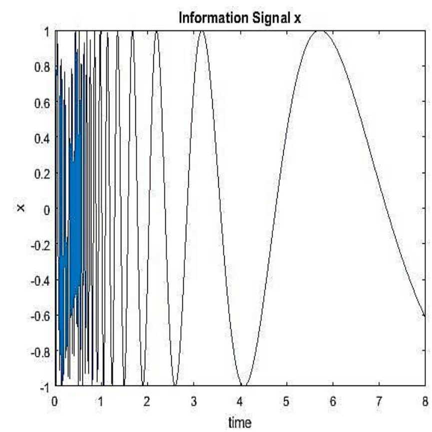


Fig. 5: Echo signal

There are use an adaptive filter to extract a desired signal from a noise-corrupted signal by filtering out the noise. The desired signal (the output from the process) is a sinusoid with 1000 samples.

```
  n = (1:1000)';
s = sin(0.055*pi*n);
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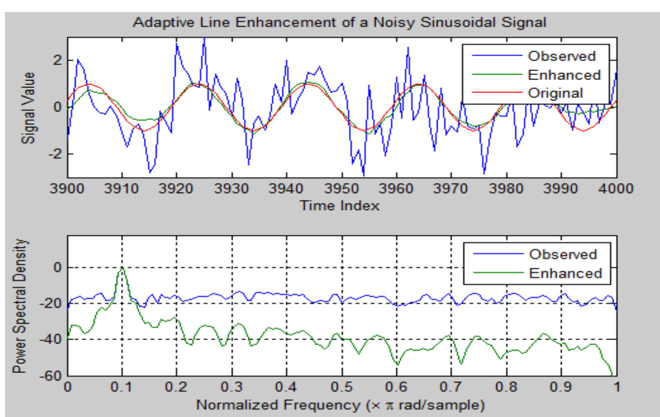
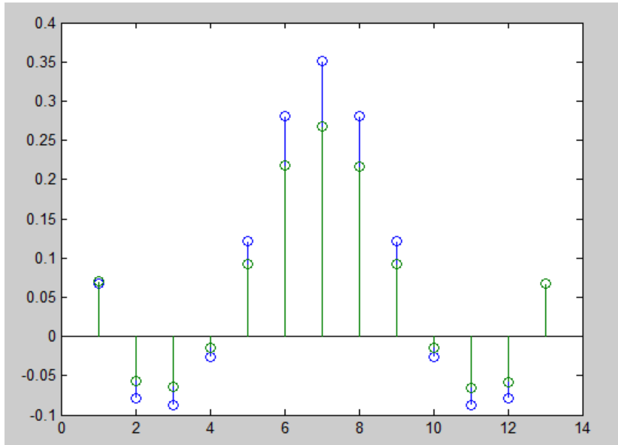


Fig.6: Comparison of the original signal and echo signals.

The following figures show the desired signal, adaptive filter output signal, estimation error, MSE and Echo return loss enhancement plots for the LMS algorithm with vocal input. The results have been obtained for 7500 iterations, 25000 iterations and with a filter length of 1000. The MSE shows that as the algorithm progresses the average value of the cost function decreases, this corresponds to the LMS filter impulse response converging to the actual impulse response, more accurately in emulating the adaptive desired signal and thus more effectively canceling the echoed signal. The results have been shown only for no of iterations= 400 and filter order 36.



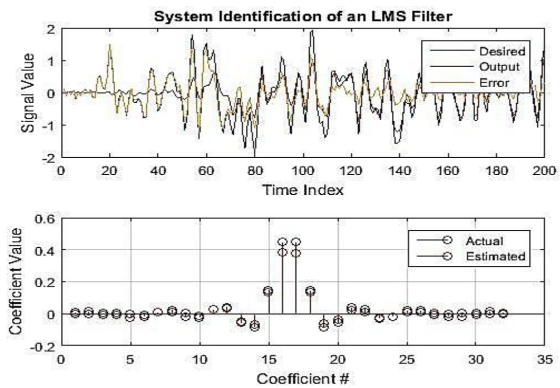


Fig.8 Desired echo signal with impulse response

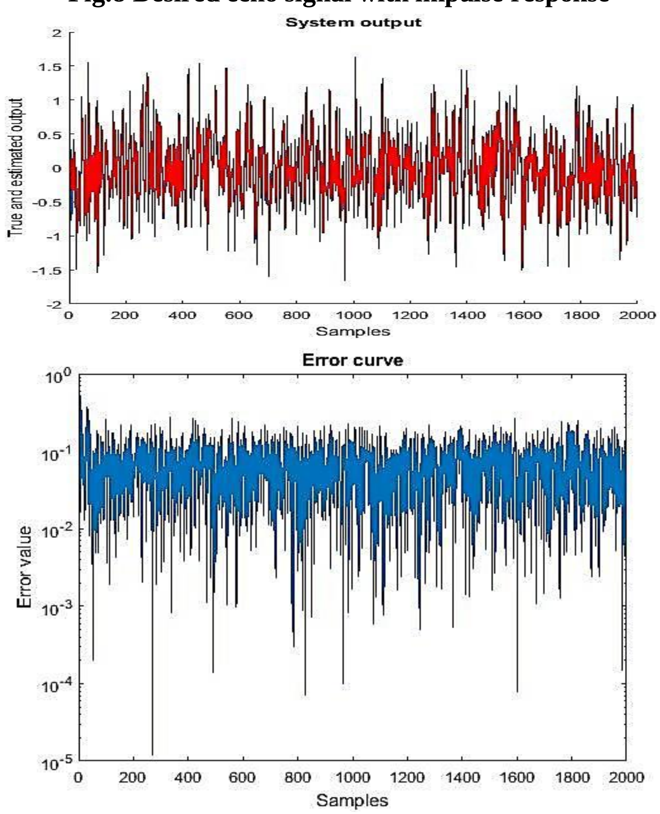


Fig.9 Simulation Results for Echo Cancellation using LMS algorithm

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The results for the Normalized LMS algorithm are given below. The results obtained are better compared to the LMS, variable Step size LMS and Gradient adaptive step size algorithms. The estimation error and the mean square error are very small and the value of average ERLE is large=22.1603Db.This is the best algorithm for the practical implementation of the echo canceller. The results have been shown only for no of iterations= 10000Plot for the desired signal for NLMS algorithm and output and error signal.

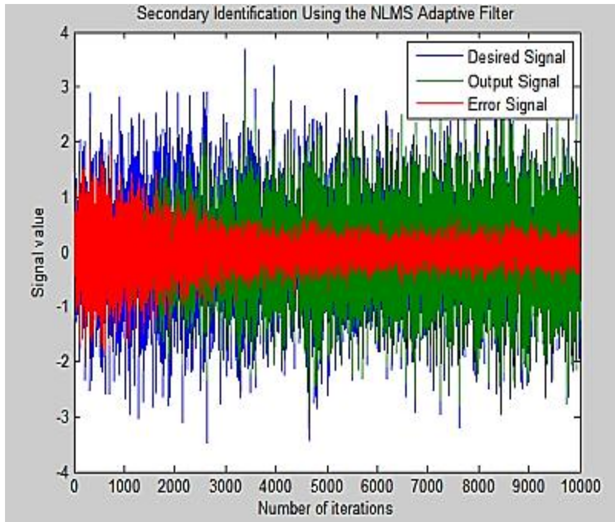

Fig.10Simulation Results for Echo Cancellation using NLMS Algorithm (Number of Iterations = 10000 And Filter Order 7500)

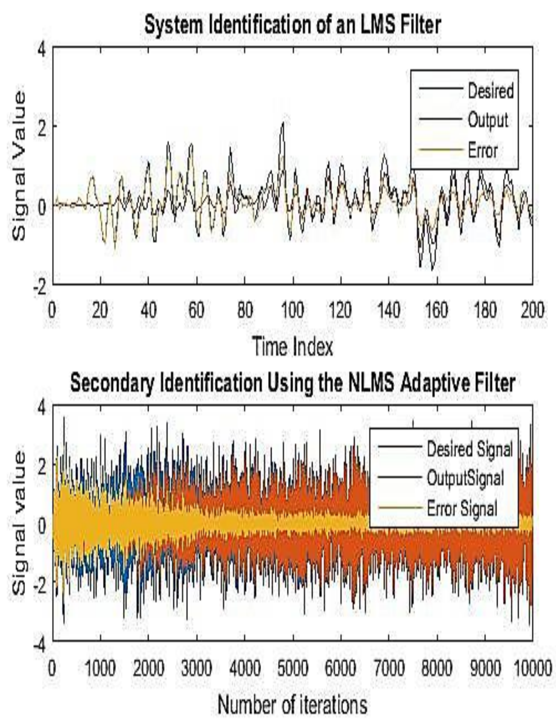

Fig.11 Comparison of the ERLE plots for the LMS and NLMS Algorithm for 400 and 800 iteration.

The results for the RLS algorithm are given below. From the plots it can be shown that the results obtained are the best for the RLS algorithms. This is estimation error is very small, even smaller than the NLMS algorithm and the average ERLE is 40.2825dB, which is much higher than the LMS and NLMS algorithm. Though the RLS algorithm gives much better results compared to other algorithms still it is not used, as each iteration requires $4N^2$ multiplications.Thus the number of multiplications required is very large because of which the RLS algorithm is too costly to implement.

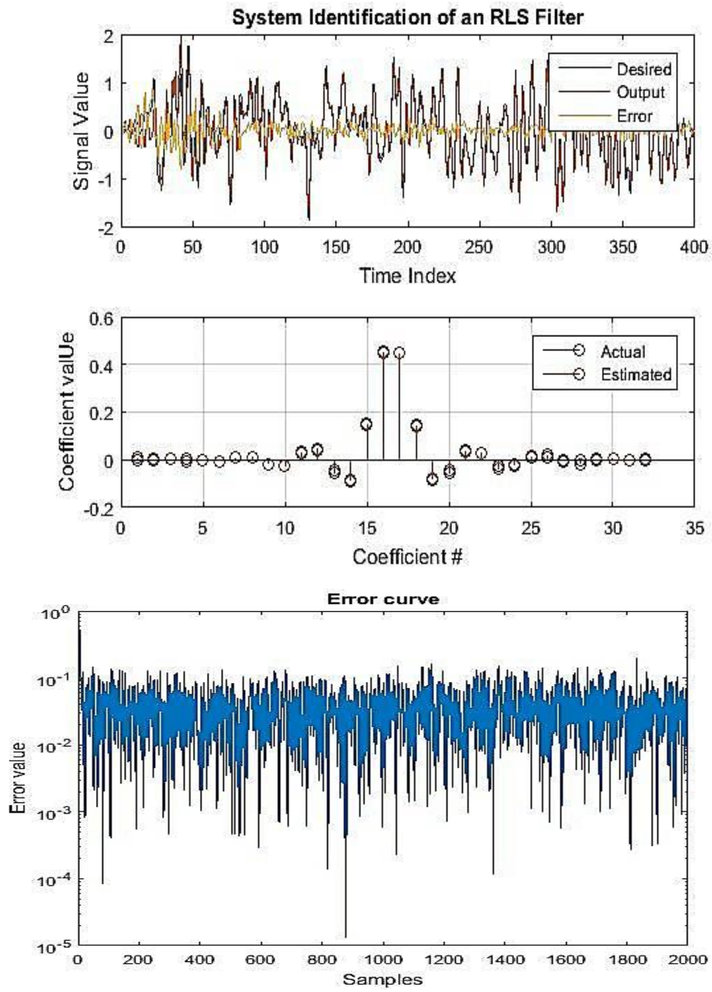

Fig.12 Plot for the desired signal, output and estimation error using RLS algorithms (Number of Iterations = 400 and filter order 200)

Table 1: Comparison of performance of various algorithms used for Echo cancellation (number of iterations = 7500 and filter order =1000)

ALGORITHM	ITERATIONS	FILTER ORDER	MSE	AVG ERLE (in dB)	COMPUTATIONS
LMS	7500	1000	0.025	10.22	2N+1
NLMS	7500	1000	0.007	16.42	3N+1
RLS	7500	1000	0.003	29.64	$4N^2$

CONCLUSION

The tables1summarize the various results obtained and compare the performance of various results obtained and compare the performance of various algorithms used for Echo Cancellation. The values of average e ERLE obtained from the plots for various adaptive algorithms shows that the ERLE is maximum for RLS algorithm. The estimation error and the mean square error are several orders smaller for RLS algorithm. The convergence rate of NLMS algorithm is greater than the LMS algorithm and the RLS algorithm gives much better results compared to the other algorithms, still it is not used, as iteration requires $4N^2$ multiplications. For the echo cancellation systems the FIR filter order is usually in the thousands. Thus the number of multiplications required is very large because of which the RLS algorithm is too costly to the implement. In practice the LMS based algorithms, although poorer performances are preferred.

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