PERFORMANCE ANALYIS OF LMS ADAPTIVE FIR FILTER AND RLS ADAPTIVE FIR FILTER FOR NOISE CANCELLATION

Jyotsna Yadav[1], Mukesh Kumar[2], Rohini Saxena[3], A. K. Jaiswal[4]

[1]M.Tech Student, SSET, SHIATS, Allahabad, India
jyotsna10jul@gmail.com
[2]Assistant Professor, SSET, SHIATS, Allahabad, India
mukesh044@gmail.com
[3]Assistant Professor, SSET, SHIATS, Allahabad, India
rohini.saxena@gmail.com
[4] Prof. & HOD, ECE, SSET, SHIATS, Allahabad, India
akjaiswal@gmail.com

ABSTRACT

Interest in adaptive filters continues to grow as they begin to find practical real-time applications in areas such as channel equalization, echo cancellation, noise cancellation and many other adaptive signal processing applications. The key to successful adaptive signal processing understands the fundamental properties of adaptive algorithms such as LMS, RLS etc. Adaptive filter is used for the cancellation of the noise component which is overlap with undesired signal in the same frequency range. This paper presents design, implementation and performance comparison of adaptive FIR filter using LMS and RMS algorithms. MATLAB Simulink environment are used for simulations.

KEYWORDS

FIR, Adaptive Filter, LMS, RLS.

1. INTRODUCTION

In the digital signal processing the major problem occurs while deigning the filter, at the receiver processing in order to transmit enormous amount of data within the filter band. Tighter filter parameters are the need of the day. This work represents the performance analysis and comparison between the LMS and RLS Adaptive FIR filter. Interest in adaptive filters continues to grow as they begin to find practical real-time applications in areas such as channel equalization, echo cancellation, noise cancellation and many other adaptive signal processing applications. On the contrary in the case of the Adaptive Filters, they are implemented where ever there is the need for the digital filter's characteristics to be variable, adapting to changing signal. Adaptive filtering consists of two basic operations; the filtering process which generates an output signal from an input data signal, and the adaptation process which adjusts the filter coefficients in a way to minimize a desired cost function. Basically, there are a large number of filter structures and algorithms that have been used in adaptive filtering applications

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Adaptive filters are an important part of signal processing. Adaptive filter is a nonlinear filter since its characteristics are dependent on the input signal and consequently the homogeneity and additivity conditions are not satisfied [7]. The key to successful adaptive signal processing understands the fundamental properties of adaptive algorithms such as LMS, RLS etc. Application of adaptive filter is the cancellation of the noise component, an undesired signal in the same frequency range.

2. ADAPTIVE FIR FILTER

Adaptive filters are an important part of signal processing. Adaptive filters adjust its transfer function according to an optimizing algorithm. Due to the complexity of the optimizing algorithms, most adaptive filters are digital filters [7]. Adaptive filters perform digital signal processing and adapt their performance based on the input signal [7]. Adaptive filter can be classified as linear and non-linear adaptive filter.

The non-linear adaptive filters have more complicated calculations, but the actual use is still the linear adaptive filter [7]. Figure 1 shows the basic block diagram of the Adaptive FIR Filter. Where X(n) is the input signal, F(n) is the variable filter response, and G(n) is the Desired signal. In this Figure the input signal are connected to the variable filter and gives a output signal, the error signal can minimize the error by separating the actual output and desired signal by adjusting the filter coefficient.

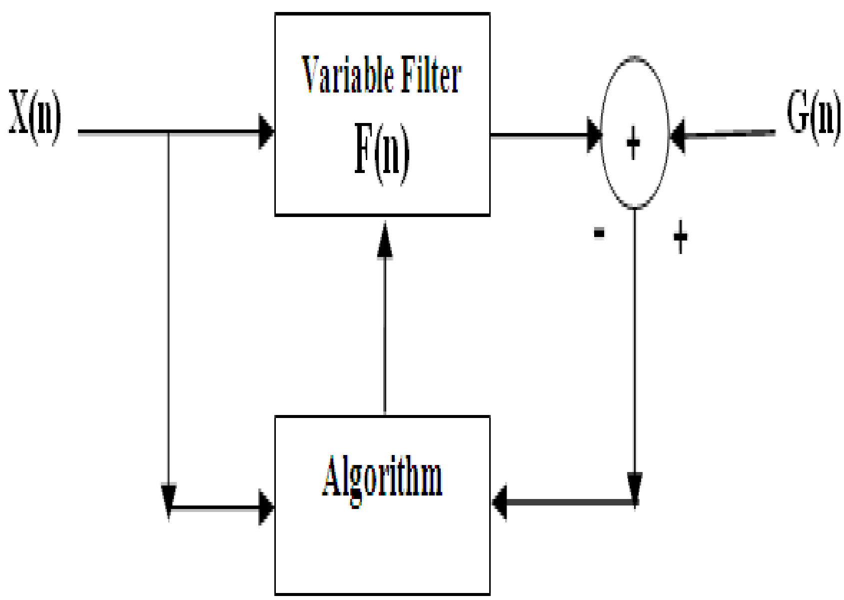

Figure 1: Block diagram of the Adaptive FIR Filter

3. ADAPTIVE ALGORITHM

3.1. LMS Algorithm

One of the most used algorithm for adaptive filtering is the LMS algorithm. LMS adjusts the adaptive filter taps and modifying them by an amount proportional to the instantaneous estimate of the gradient of the error surface [6]. It neither requires correlation function calculation nor matrix inversions, which makes it simple and easier when compared to other algorithms. Minimization of mean square error is achieved due to the iterative procedure incorporated in it to make successive corrections in the direction of negative of the gradient vector it is represented in following equations [2, 6].

$$\mathrm{Y\,(n)\ =\ F\,(n).\,U\,(n)} \qquad \text{(i)}$$

$$\mathrm{E\,(n)\ =\ G\,(n) - Y\,(n)} \qquad \text{(ii)}$$

$$\mathrm{F\,(n\ +\ 1)\ =\ F\,(n)\ +\ \mu.\,U\,(n).\,E\,(n)} \qquad \text{(iii)}$$

Where, Y(n) = filter output, X(n) = input signal, E(n) = error signal, G(n) = other observed signal

3.2. RLS Algorithm

At each instant, RLS algorithm performs an exact minimization of the sum of the squares of the desired signal estimation errors [3, 4, 6]. The computations begin with known initial conditions and based on the information's contained in the new data samples, RLS algorithm updates the old estimates. These are its equations: To initialize the algorithm **P** (*n*) (inverse correlation matrix) should be made equal to δ^{-1} where δ (regularization factor) is a small positive constant [2, 6].

$$\mathrm{Y\,(n)\ =\ F\,(n).\,U\,(n)} \qquad \text{(i)}$$

$$\alpha\,\mathrm{(n)\ = G(n) - F(n)\,u(n)} \qquad \text{(ii)}$$

$$\pi\,\mathrm{(n)\ =\ P(n-1)\,u(n)} \qquad \text{(iii)}$$

$$\mathrm{k(n)\ =} \lambda\ +\ \pi\,\mathrm{(n)\ \ u(n)} \qquad \text{(iv)}$$

$$\mathrm{K\,(n)} = \frac{\mathrm{P(n-1)\,u(n)}}{\mathrm{k(n)}} \qquad \text{(v)}$$

$$\mathrm{F(n)\ =\ F(n-1)\ +\ K(n)}\,\alpha\,\mathrm{(n)} \qquad \text{(vi)}$$

$$\mathrm{P1(n-1)\ =\ K(n).}\,\pi\,\mathrm{(n)} \qquad \text{(vii)}$$

$$\mathrm{P(n)\ =\ \{\,P(n-1)\ -\ P1(n-1)\,\}\,/\,}\lambda \qquad \text{(viii)}$$

Where, F(n) = filter coefficients, K(n) = gain vector, λ = forgetting factor, P(n) = inverse correlation matrix of the input signal α (n), π (n) = positive constant

4. SIMULATION METHODOLOGY

The methodology is used for the comparison between the performance of Adaptive FIR filter using Least Mean Square (LMS) and Recursive Least square (RLS) algorithms. This work used MATLAB FDA tool and Simulink environment for the filter realization. We have used a Various Parameters for the simulation are Filter Structure (direct form), Filter length (32), Filter type (type 2), adaptive algorithm (LMS, RLS), Simulation time (30, 50, 80), Design method (Kaiser Window β=0.5), and frequency specifications (Normalized 0-1).

Figure-2 and Figure-3 shows the MATLAB implementation of adaptive FIR Filter using LMS and RLS algorithm. Which have four output; input signal, input siganl with noise signal, filter output and error signal produed at scope 1 and weight output (Wts) are produced at the vector scope. LMS and RLS Filters can be used to reduce the noise by seprating the error between the actual output and desired signal mixed.

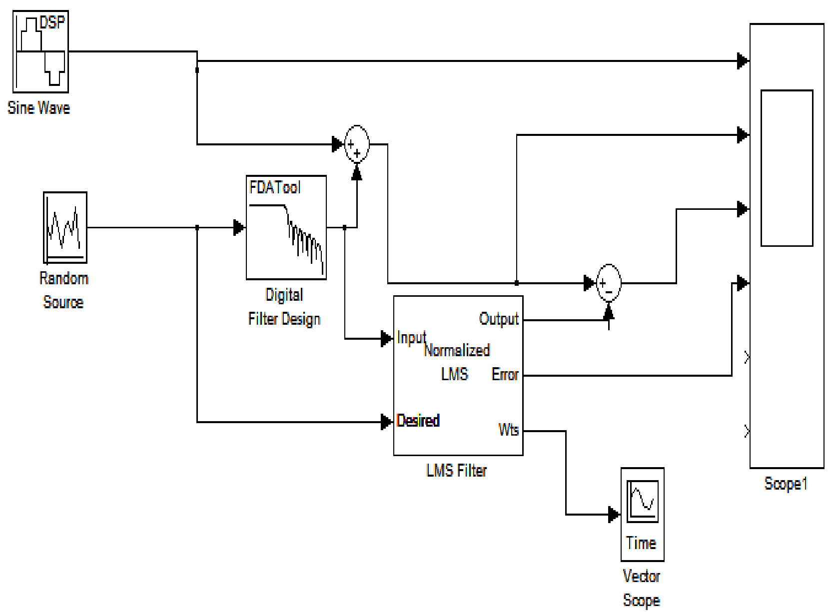

Figure 2: Implemented Adaptive FIR filter using LMS algorithm

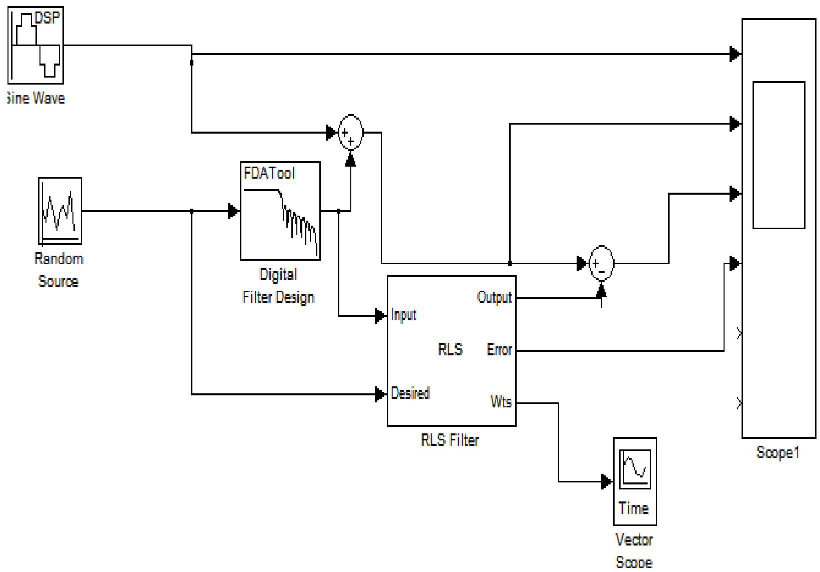

Figure 3: Implemented Adaptive FIR filter using RLS algorithm

5. SIMULATION RESULTS

Figure 4, 5 and 6 shows the scope output of Adaptive FIR filter using LMS algorithm for simulation time t=30, 50, 80 second respectively. It is observed that as the simulation time increases the error or noise from the signal is effectively removed using LMS filtering. More the simulation time more the reduction of noises from the signal.

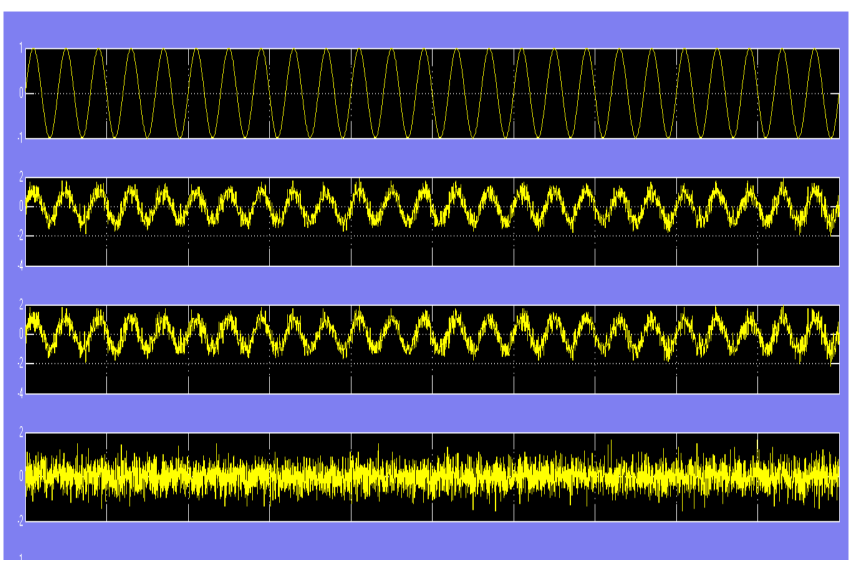

Figure 4: Scope output of Adaptive FIR filter using LMS algorithm for simulation time 30 seconds

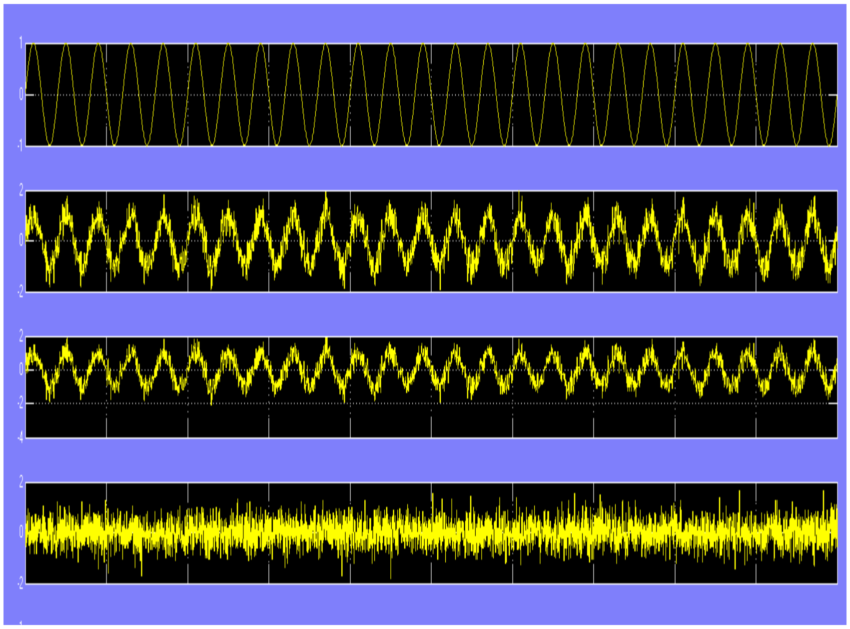

Figure 5: Scope output of Adaptive FIR filter using LMS algorithm for simulation time 50 seconds

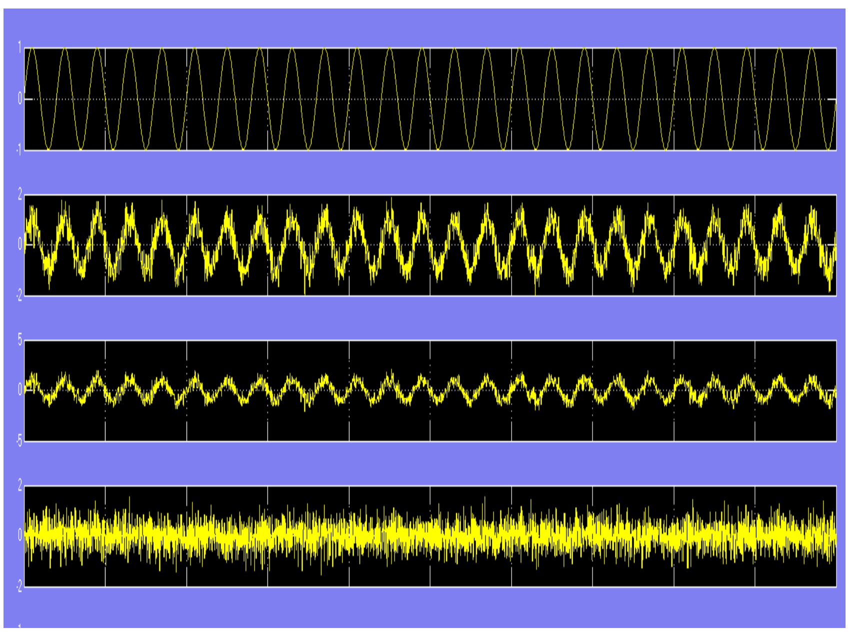

Figure 6: Scope output of Adaptive FIR filter using LMS algorithm for simulation time 80 seconds

Figure 7, 8, 9 shows vector time scope output of Adaptive FIR filter using LMS algorithm for simulation time 15, 30 and 80 second respectively, it is observed that as the simulation time increases the error or noise from the signal is effectively removed. As the simulation time increases the noises in the signal are reduced to almost zero.

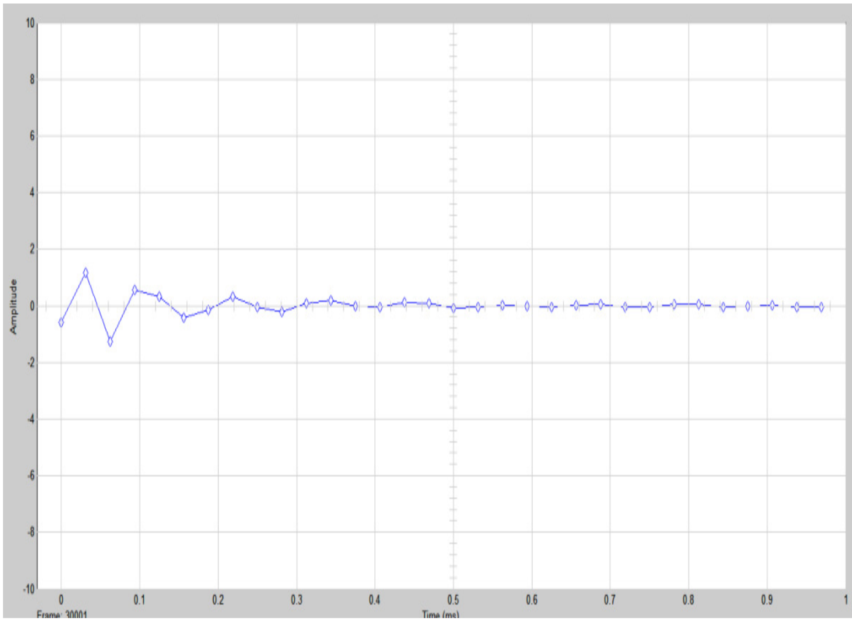

Figure 7: Coefficients Adjustment in the Adaptive FIR Filter using LMS algorithm for simulation time 30 seconds

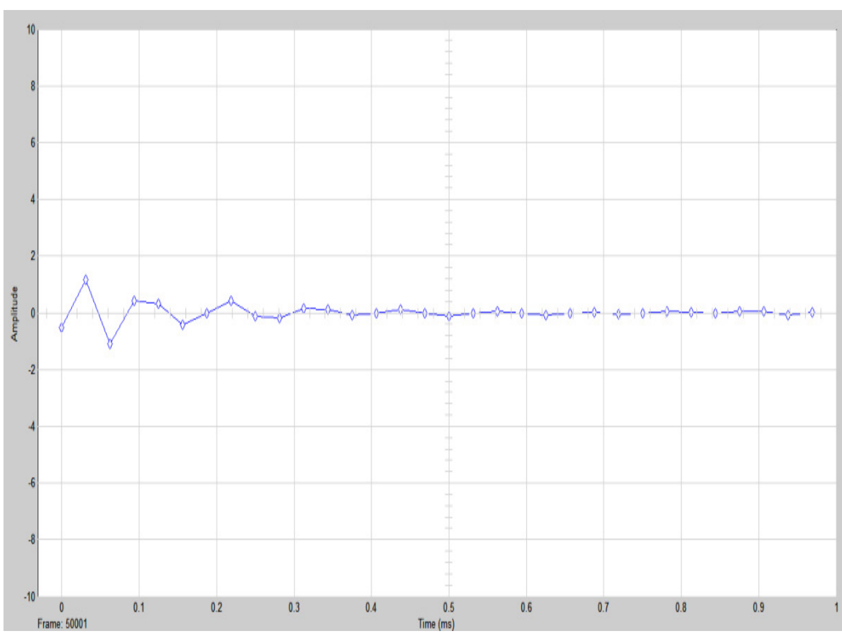

Figure 8: Coefficients Adjustment in the Adaptive FIR Filter using LMS algorithm for simulation time 50 seconds

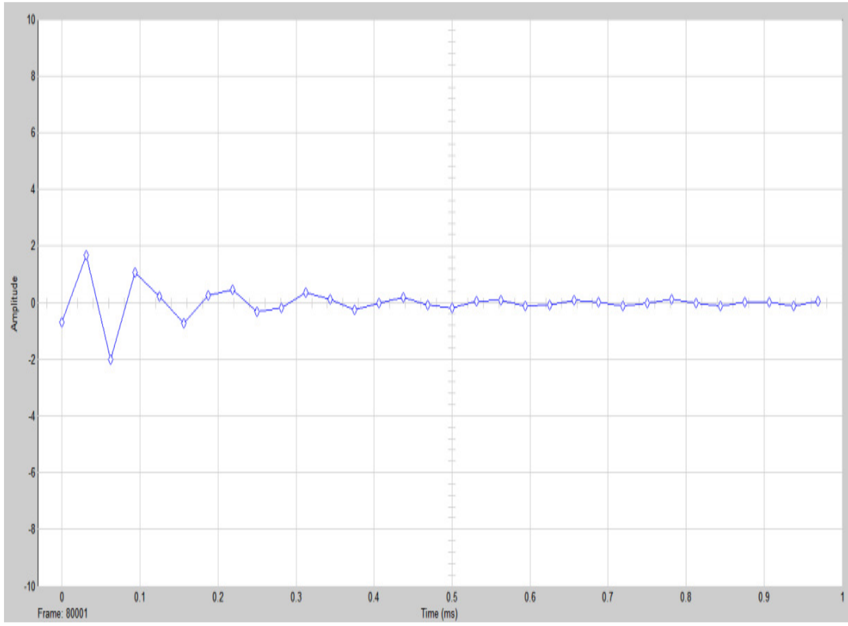

Figure 9: Coefficients Adjustment in the Adaptive FIR Filter using LMS algorithm for simulation time 80 seconds

Figure 10, 11, 12 shows scope output of Adaptive FIR filter using RLS algorithm for simulation time t=30, 50, and 80 second respectively. It is observed that as the simulation time increases the error or noise from the signal is removed but not effectively as that of in the case of LMS filtering.

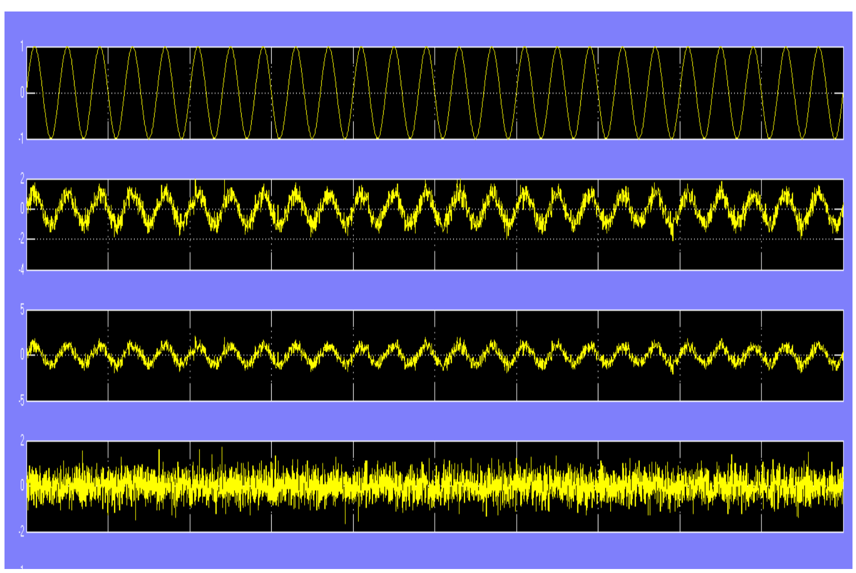

Figure 10: Scope output of Adaptive FIR filter using RLS algorithm for simulation time 30 seconds

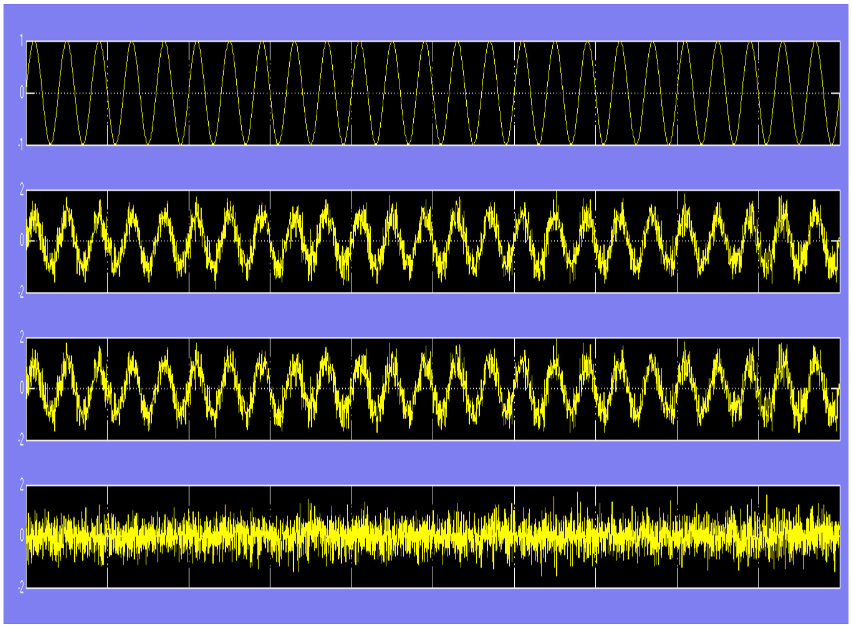

Figure 11: Scope output of Adaptive FIR filter using RLS algorithm for simulation time 50 seconds

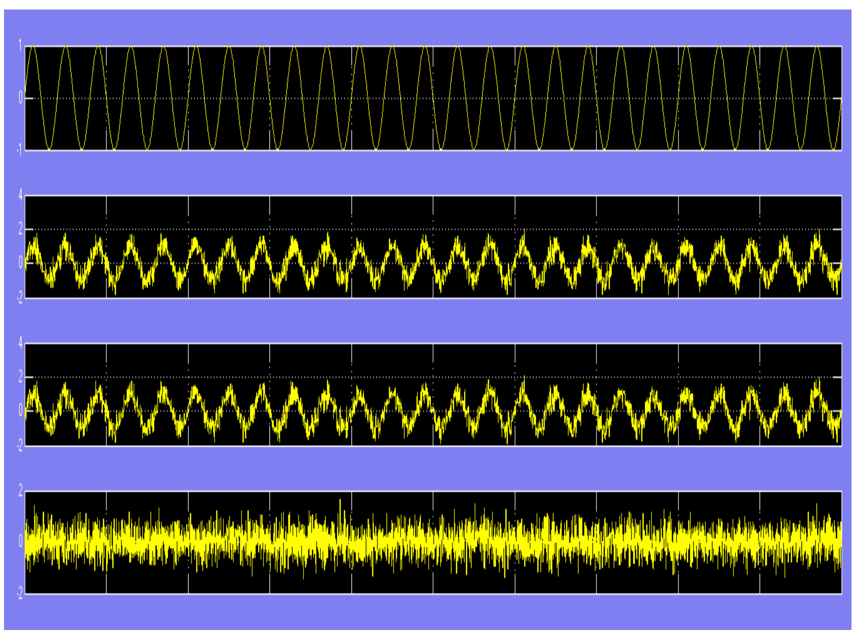

Figure 12: Scope output of Adaptive FIR filter using RLS algorithm for simulation time 80 seconds

Figure 13, 14, 15 shows time scope output of Adaptive FIR filter using RLS algorithm for simulation time t=30, 50, 80 second respectively. It is observed that as the simulation time increases the error or noise from the signal is removed but not effectively. As the simulation time increases we found that the noises in the signal are not reduced to zero and still some noises are presents in the signal.

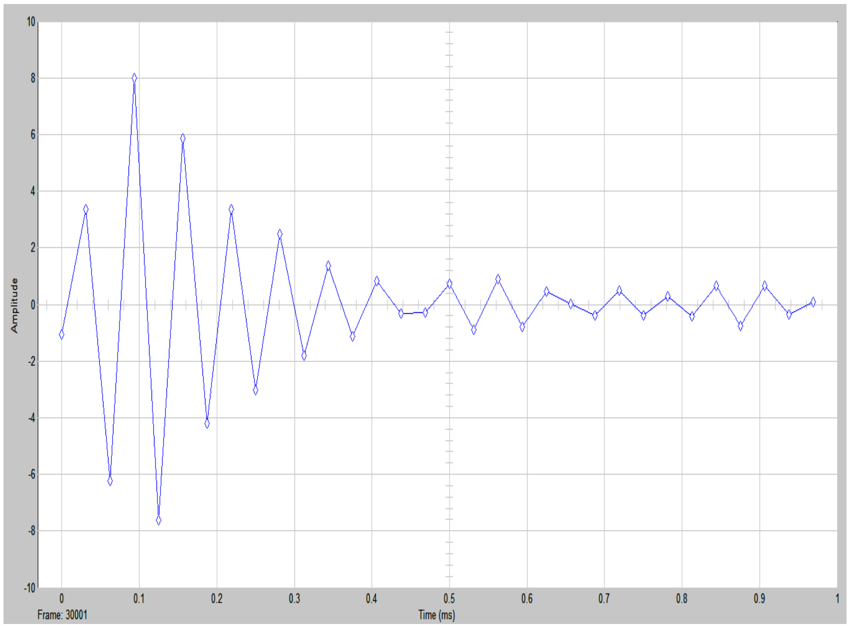

Figure 13: Coefficients Adjustment in the Adaptive FIR Filter using RLS algorithm for simulation time 30 seconds

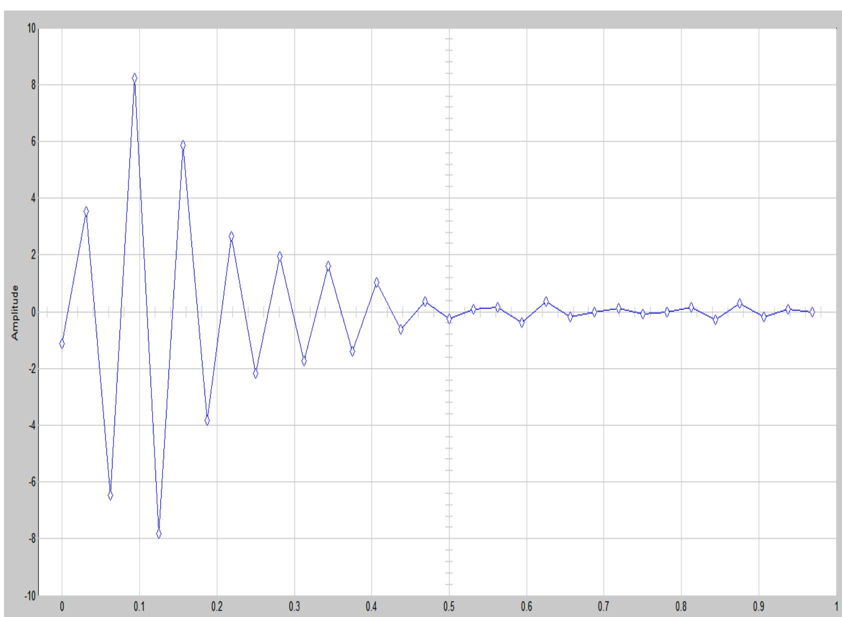

Figure 14: Coefficients Adjustment in the Adaptive FIR Filter using RLS algorithm for simulation time 50 seconds

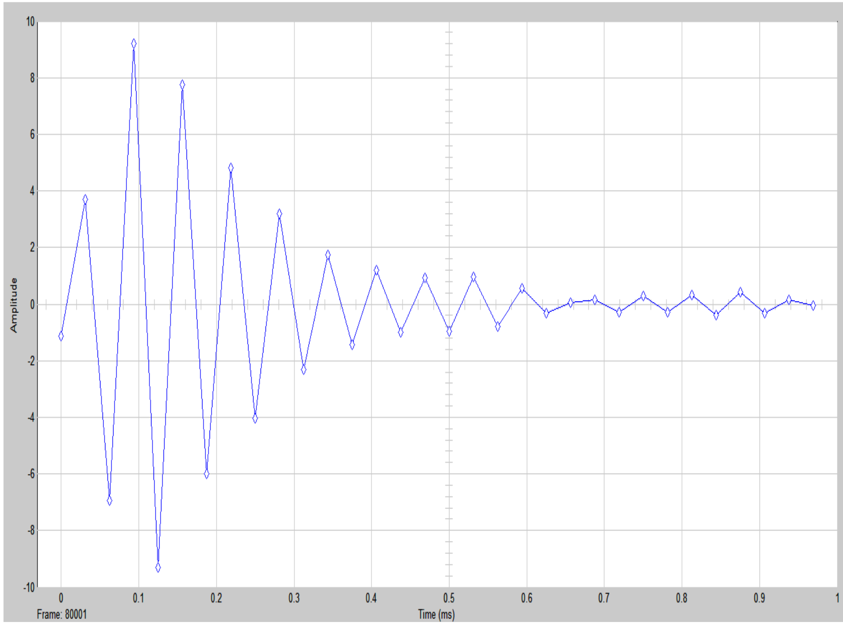

Figure 15: Coefficients Adjustment in the Adaptive FIR Filter using RLS algorithm for simulation time 80 seconds

6. CONCLUSION

The implementation and simulation of Adaptive FIR filter using LMS and RLS algorithm have been done using MATLAB Simulink environment and their responses have been studied and compared in various waveforms as given in the simulation results. After comparing these**,** it shows that as simulation time increases (t=30, 50, 80 sec) in case of LMS filtering the noise is almost reduced from the signal whereas in case of RLS filtering the noise is not completely removed with simulation time. The Adaptive FIR filter using LMS algorithm shows relatively good filtering result, having short filter length, simple structure and simple operation, and it is easy to realize hardware. On the other hand the drawback of LMS algorithm is that the convergence rate is slower. Simulation results show that filter performance is better to the least mean squares algorithm. RLS filtering required large storage capacity and the task of noise reduction is relatively difficult with large hardware.

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Author’s profile

Prof. Arvind Kumar Jaiswal is working as Head of Department in the Department of Electronics & Communication Engineering in SHIATS, Allahabad, India. He received his M.Sc (Tech). Degree in Electronics & Radio Engineering in the year 1967 from University of Allahabad, India. His research is focused on optical Wireless Networks, Advanced Communication, Antenna and Wave propagation, optical fiber communication and control system.

Mukesh Kumar is working as Asst. Prof. in the Department of Electronics &Communication Engineering SHIATS, Allahabad. He received his M.Tech. Degree in Advanced Communication System Engineering from SHIATS, Allahabad, India in 2010. His research is focused on Wireless Sensor Network and Computer Networks Microwave Engineering, and Digital signal processing as well as Optical fiber communication.

Rohini Saxena is working as Asst. Prof. in the Department of Electronics & Communication Engineering in SHIATS, Allahabad. She received her M.Tech. Degree in Advanced Communication System Engineering from SHIATS, Allahabad in 2009. Her research is focused on Digital signal processing, Microwave Engineering and Wireless Sensor Network.

Jyotsna Yadav is studying as M.tech Scholar in the Department of Electronics & Communication Engineering in SHIATS, Allahabad, India. Her studying branch is Signal processing. Her research is focused on Digital signal processing and wireless networks.