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Performance Comparison of Adaptive FIR Filter using Different Algorithms

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Abstract: **In this paper an efficient method is presented to design and implement Adaptive FIR filter. The filter implementation is based on Least-Mean Square (LMS), Normalized Least- Mean Square (NLMS) and Recursive Least squares (RLS) adaptive algorithm which uses random signal as input. The prototype filters have been designed and simulated using Matlab tool by taking filter length as 32. In this paper the rate of convergence, Minimum mean-square error (MMSE), and system prediction performance is analyzed and compared for various algorithms. It can be observed from the simulated results that RLS gives better convergence rate as compared to LMS and NLMS. LMS algorithm gives better MMSE and system prediction of unknown system as compared to RLS and NLMS.**

Keywords: ADAPTIVE, DSP, LMS, NLMS, RLS.

1. INTRODUCTION

Due to programmability the demands for digital products are growing day by day. Various industries like audio, video, and wireless industry rely basically on digital technology and a great part of digital technology deals with digital signal processing. This domain in engineering has gained increasing interest. Most of the common functions performed by almost all DSP chips are FFTs, FIR filters, Interpolator, Decimator [1]. Digital signal processing (DSP) involves the digital representation of signals and the application of digital systems to analyze, modify, store, or extract information from the signals. DSP methodologies find application in consumer electronics, communications, automotive electronics, instrumentation, medical electronics, tomography and acoustic imaging, cartography, seismology, speech recognition, robotics etc [2]. Much research has been carried to develop DSP algorithms and systems for real-world applications. The rapid advancements occurred in recent years, in digital technologies, has supported the implementation of sophisticated DSP algorithm solutions for real-time applications. DSP algorithms can be developed, analyzed, and simulated using the high-level language and software tools such as C/C++ and MATLAB (matrix laboratory), Simulink. MathWorks and Xilinx. The performance of the algorithms can be verified using a low-cost, general-purpose computer. Therefore, a DSP-based system is relatively easy to design, develop, analyze, simulate, test, and maintain [1]. DSP is usually used to measure or filter continuous real-world analog signal data. Thus, the initial step is to convert the signal from an analog to a digital form, by using an analog to digital converter [1]. The digital filter performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal. An analog signal may be processed by digital filter by first being converted into digital domain and represented as a sequence of numbers, then manipulated mathematically, and then reconstructed into a new analog signal. The filters are used for two things: signal separation and signal restoration. Signal separation is required when a signal has been contaminated with interference, noise, or other signals. Signal restoration is required when a signal has been distorted in some way [3].

Digital filters can be described by two types of transfer functions: transfer functions of finite impulse response filters and those of infinite impulse response filter. The FIR filters have a few advantages over the IIR filters— FIR filter designing is easy to meet the constraint of required magnitude response and it achieves a constant group delay also. The FIR filters are stable and free from limit cycles that arise as a due to finite wordlength representation of multiplier constants and signal values [3].

2. ADAPTIVE ALGORITHMS

Discrete-time (or digital) filters are ubiquitous in today's digital signal processing application areas. Filters are used to achieve desired spectral response of a signal, to reject unwanted signals (noise or interferers), for bit rate reduction in signal transmission, etc. The reason of making filters adaptive, i.e., to alter parameters (coefficients) of a filter according to some adaptive algorithm, handles the problems that we might not in advance know, e.g., the characteristics of the desired signal, or of the unwanted signal (noise), or of a systems influence on the signal that we would like to compensate. Adaptive filters can adjust its coefficients to unknown environment, and can even track signal or system characteristics varying over time [4]. The well defined specifications are required to be available before designing for the fixed coefficient filter. But, there can be situations where the specifications are not available, or are varying with time. For such cases a digital filter with adaptive coefficients is employed, known as adaptive filters [5, 6].

Since no specifications are available beforehand, the adaptive algorithm that determines the updating of the filter coefficients. Thus, requires extra information that is usually given in the form of a signal called a desired or reference signal, whose choice is normally a complex task that depends on the type of application. Adaptive filters are considered to be the nonlinear systems. Hence, the behavioral analysis is more complicated than for fixed coefficient filters. On the other hand, the adaptive filters are self designing filters. Thus, from the designer's point of view, their design is less involved than in the case of the fixed coefficient digital filters [6]. The general set up of an adaptive filter is illustrated in Figure 1, where k denotes the iteration number, x(k) denotes the input signal, y(k) denotes the adaptive-filter output signal, and d(k) denotes the desired signal. The error signal e(k) is calculated as : d(k) – y(k).

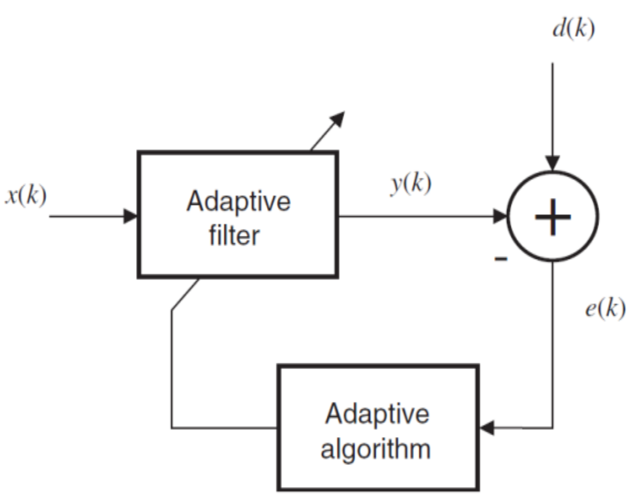

Figure 1 General Adaptive Filter Structure

The error signal is then used to form an objective function that is required by the adaptive algorithm in order to determine the appropriate updating of the filter coefficients [6, 7]. This paper presents the performance analysis of some important features and comparison between the LMS, NLMS and RLS Adaptive algorithm for FIR filter. Adaptive filters are used for real-time applications in areas such as channel equalization, system prediction, echo cancellation, noise cancellation and many more adaptive signal processing applications [8].

In contrast, the adaptive filters are implemented in the areas where there is the need for the digital filter's characteristics to be variable. Adaptive filtering performs: the filtering process which results in an output signal from an given input data signal, and the adaptation process which adapts the filter taps so that the error signal is minimised, desired cost function. There are large number of filter structures and algorithms that have been used in adaptive filtering applications. The adaptive signal processing is based on the fundamental properties of adaptive algorithms such as LMS, NLMS, and RLS etc. Most common application of adaptive filter is the cancellation of the noise component that lie in the same frequency range [6].

LMS is the frequently used algorithm in adaptive filtering. It is basically a gradient descent algorithm which means that it adjusts the adaptive filter coefficients by modifying them by an amount which is proportional to the gradient of the error surface. It can be represented in following equations [7],

$$f(u(n), e(n), \mu) = \mu e(n) u^*(n) \tag{1}$$

NLMS can be formulated by the natural formulation of the conventional LMS filter. The weight vector of an adaptive filter change in a minimal manner subject to a constrained imposed while we move from one iteration to the next [5].

$$f(u(n), e(n), \mu) = \mu e(n) \frac{u^*(n)}{\varepsilon + u^H(n) u(n)} \tag{2}$$

where u(n) is input, e(n) is error and μ is step size.

RLS algorithm performs the minimization operation for the sum of the squares of the desired signal estimation errors described at each instant. To initialize the algorithm P(n) should be made equal to δ-1 where δ is a small positive constant. [9]

$$K = \frac{Pu}{(Forget\ Factor) + u^H Pu} \tag{3}$$

where u is input, P is current inverse correlation matrix, K is Kalman gain vector, and H denotes the Hermitian transpose.

3. DESIGN SPECIFICATIONS & SIMULATION

The methodology is used for the comparison between the performance of Adaptive FIR filter using Least Mean Square, Normalized Least-Mean Square and Recursive Least Square algorithms. This work used MATLAB for the filter realization [7, 10]. In the proposed work various parameters used for the simulation are direct form Filter Structure, Filter length as 32, Filter type 2, adaptive algorithm, and FIR based design method.

In this proposed work FIR filter has been designed and simulated using MATLAB Tools by taking filter order as 32 along with adaptive filter structure to compare various algorithms. The figure 2, 3 and 4 shows the curve of Normalized MMSE trend (lies between 0 & 1). The simulated outputs for average coefficient values for different weights and Measured MMSE is shown in figure 2, 3 and 4 for various algorithms:-

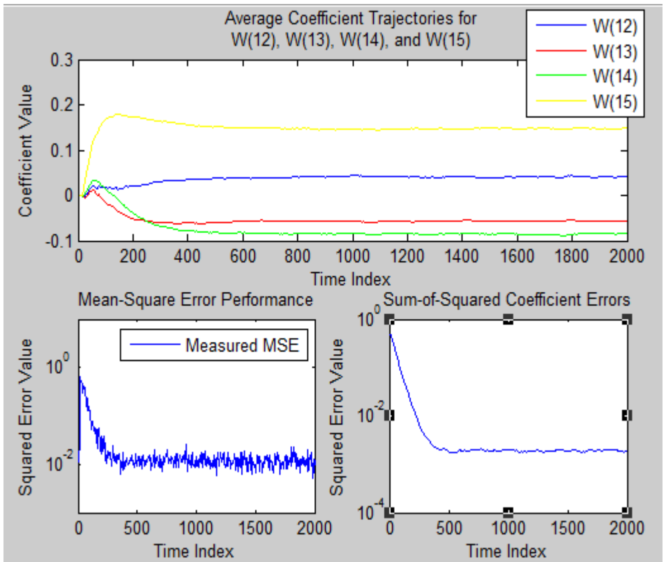

Figure 2 MMSE of LMS Algorithm

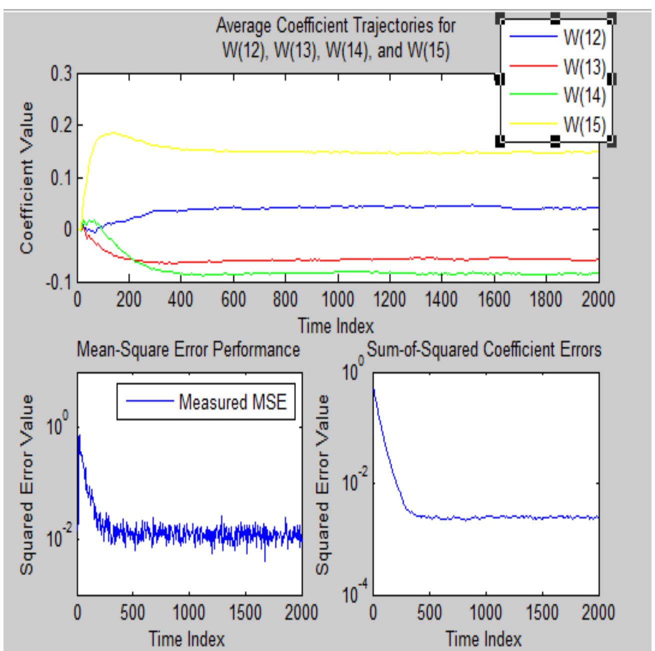

Figure 3 MMSE of NLMS Algorithm

The curve plotted between the MMSE and the number of iterations is known as the 'learning curve'. The learning curve denotes the natural mode of the algorithm. The value of MMSE for the LMS algorithm is least because the LMS algorithm gives the exact estimation of the system response but numbers of iterations required are more. The transient behaviour of the coefficients with varying filter weights is also shown in the results.

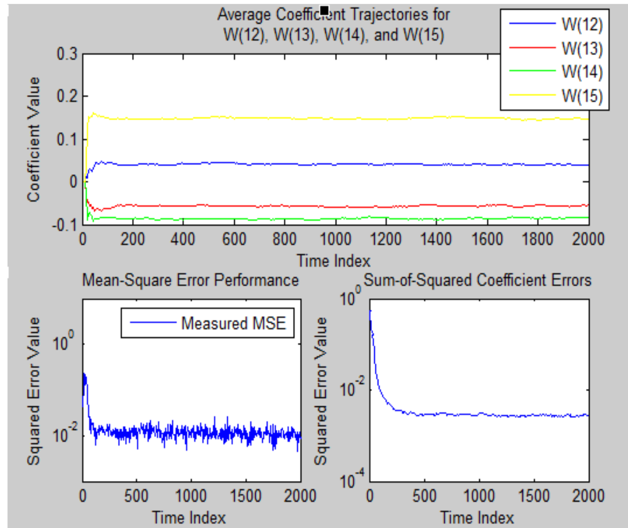

Figure 4 MMSE of RLS Algorithm

The adaptive noise canceller arrangement consist of an adaptive filter that gives reference sensor output to produce an estimate of the noise, which is then subtracted from the primary sensor output. The simulated results for Noise cancellation by LMS, NLMS, and RLS algorithm are as follows:-

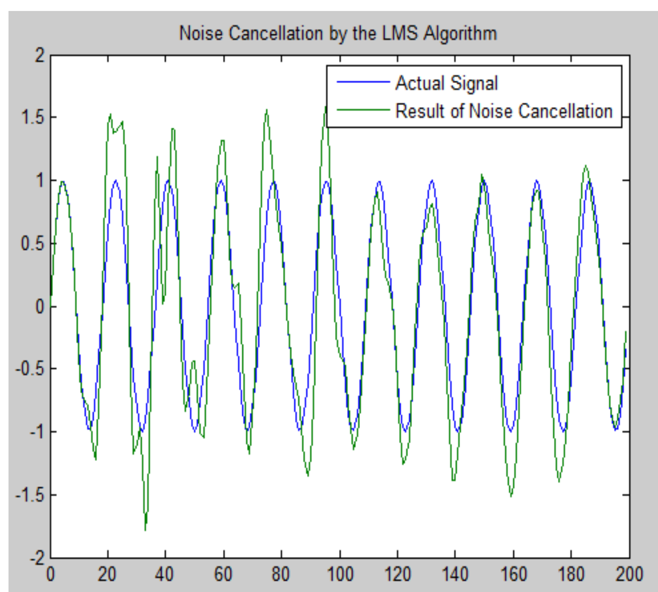

Figure 5 LMS based Noise Cancellation

The overall output of the noise canceller is used to control the adjustment applied to the tap weights in the adaptive FIR filter. The adaptive canceller tends to minimise the MMSE value of the overall output, thereby causing the output to be the best estimate of the desired signal. The plots have been obtained for various adaptive algorithms. As it can be observed from the figure 5-7, that LMS based noise cancellation gives better results than other algorithms compared in this paper.

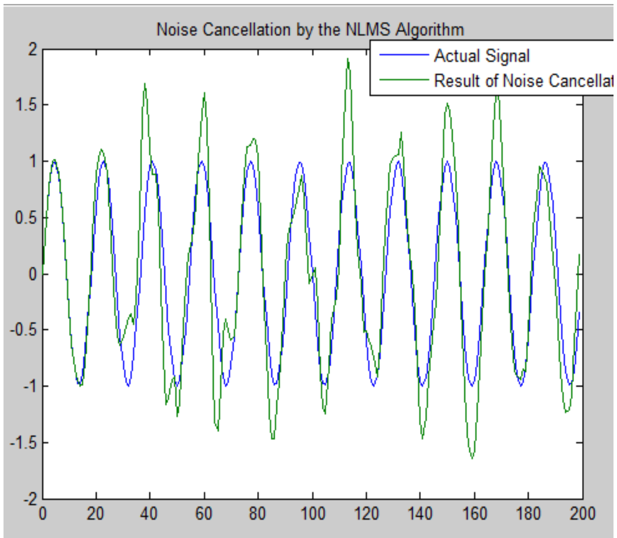

Figure 6 NLMS based Noise Cancellation

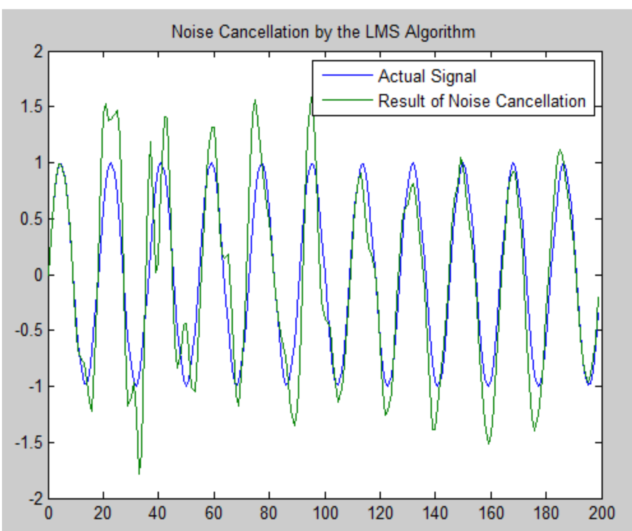

Figure 7 RLS based Noise Cancellation

System identification is the process of specifying the unknown model in terms of available experimental evidence. The error has to be optimized for better system prediction. The estimated value should lie near to the actual value for exact system prediction. This paper shows the system coefficients identification of the FIR filter using various algorithms. The simulated results for System identification for LMS, NLMS, and RLS are as follows:-

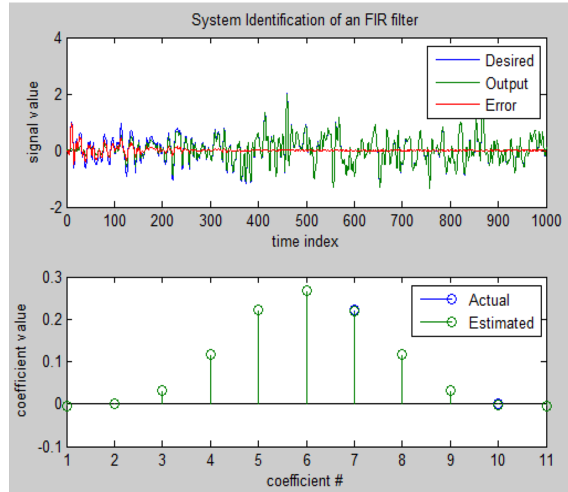

Figure 8 LMS based System Identification

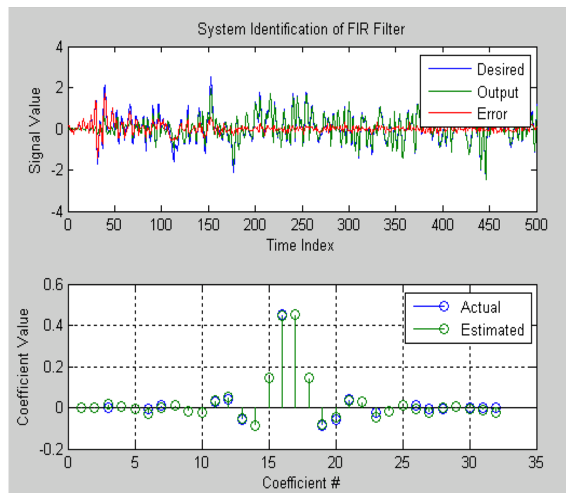

Figure 9 NLMS based System Identification

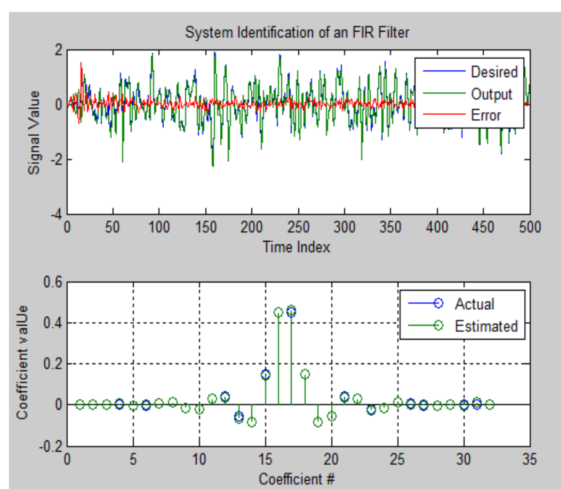

Figure 10 RLS based System Identification

4. Result Discussions

From the figure 2-4, the trend of MMSE can be withdrawn. The figure 11 shows the increasing value of MMSE as we move from LMS to RLS adaptive algorithm. It can be analysed from the graph that the MMSE is minimum for LMS algorithm as compared to NLMS and RLS algorithms. If Normalised MMSE is 0, the optimum filter works perfectly which means that there is complete agreement between the estimated output and the desired output and on the other hand, if Normalised MMSE is 1, this corresponds to the worst possible situation.

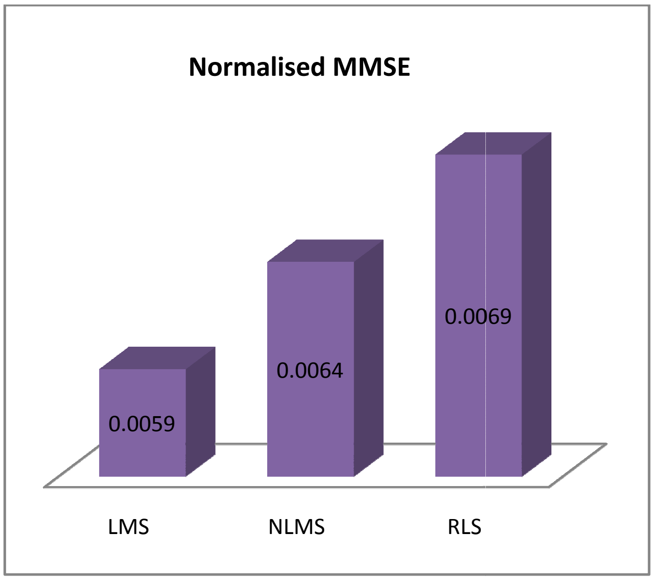

Figure 11 MMSE Comparison for various algorithms

It can be analysed from figure 5-7, the LMS algorithm gives the best results in noise cancellation i.e. it adapts to the desired input signal very accurately and the system coefficients are predicted near to the actual coefficients. But the RLS and NLMS gives poor response when it is applied to the system prediction and noise cancellation. As the rate of convergence is very fast of RLS, the adapted coefficients do not fall with the desired input signal.

A LMS filter is built around the transversal structure. LMS algorithm gives accurate prediction of system coefficients as can be seen from figure. 8-10, when the step size is kept small, but the number of iterations required is more i.e. the rate of convergence of the LMS algorithm is slow as can be seen due to large matrix calculations. The performance of LMS degrade as we increase the step size i.e. misadjustment increases. Figure 12 shows the trend of rate of convergence for various algorithms. As can be seen from the graph, The RLS algorithm has the fastest rate of convergence as compared to the NLMS and LMS.

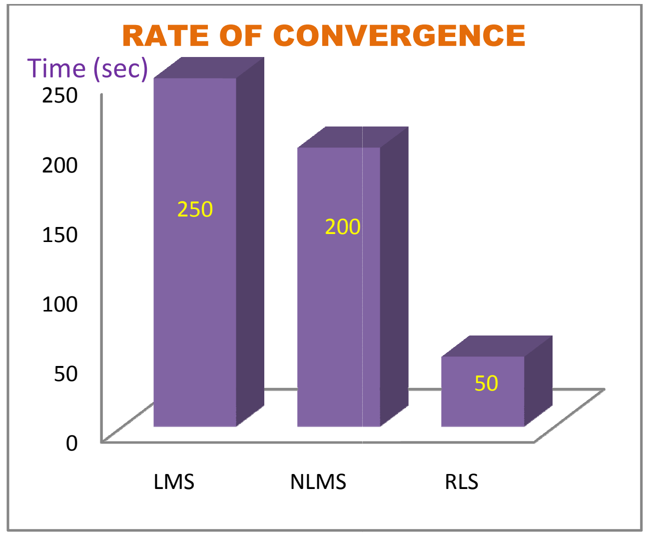

Figure 12 Rate of Convergence Comparison

4. CONCLUSIONS

This paper presents the performance comparison of adaptive FIR filter using different algorithms, namely, LMS, NLMS, and RLS. These algorithms are compared on the basis of rate of convergence, MMSE, and system coefficient prediction. Thus from the above results it can be concluded that the LMS algorithm gives least normalised MMSE (0.0059) value than that of NLMS and RLS. System prediction performance is best for the LMS based design. The rate of convergence is fastest for the RLS based filter due to ease of computation and recursive nature of the algorithm. Hence the RLS based filters cannot be used for efficient system prediction and noise cancellation applications such as seismology, ECG, EEG, etc where accuracy is required LMS algorithm can be implemented.

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REFERENCES

[1] John G. Proakis, Manolakis, “Digital Signal Processing”, Fourth edition, Pearson education, pp. 2-6, 2007.

[2] B. A. Shenoi, “Introduction to Digital Signal Processing And Filter Design”, John Wiley & Sons Inc, 2nd Edition, Vol. 3, pp. 23-26, 2006.

[3] S. K. Mitra, ‘Digital Signal Processing’, A Computer based approach, Mc-Graw Hill, 3rd Edition, pp. 220-222, 739-740, 2006.

[4] Simon Haykin, “Adaptive filter theory”, Fourth edition, Prentice Hall Inc., pp. 231-238, 271, 320-328, 385, 2009.

[5] E. C. Ifeachor, and B. W. Jervis, “Digital Signal Processing: A Practical Approach”, Prentice hall, Third Edition, pp. 342-440, 2009.

[6] Jyotsna Yadav, Mukesh Kumar, Rohini Saxena, A. K. Jaiswal, “Performance Analyis of LMS Adaptive Fir Filter And RLS Adaptive Fir Filter For Noise Cancellation”, *Signal & Image Processing : An International Journal (SIPIJ)*, Vol.4, No.3, pp. 45-56, June, 2013.

[7] K. R. Borisagar, G. R. kulkarni “Simulation and Comparative Analysis of LMS and RLS Algorithms”, *Global Journal of Researches in Engineering,* Vol.10 Issue 5, pp. 44-46, October, 2010.

[8] S Haykin, A.H. Sayed, J.R. Zeidler, P Yee, P.C. Wei, "Adaptive tracking of linear time variant systems by extended RLS algorithms", *IEEE Transactions on Signal Processing, Volume: 45, No. 5*, pp. 1118 -1128, May, 1997.

[9] Mathworks, “Users Guide Filter Design Toolbox-4”, March-2007.

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