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A Simulation & Performance Analysis of ANC using Adaptive Filters

Deepika , Isha Chauhan
Research Scholar GITAM, Jhajjar, Asst. Professor, GITAM, Jhajjar

*Abstract - One of the main objectives within the adaptive signal processing is the noise suppression, i.e. the detection of information bearing signal in noise. Adaptive filter application ANC (Adaptive Noise Cancellation) using adaptive Filter algorithm is an important basis for signal processing. The device automatically filters out the components of the signal which are not related to the orignal signal means uncorrelated signal in time and passes the similar signal i.e. correlated portions of the signal. Since the output as the orignal signal of the device is determined by the input signal statistics and the properties of the filter here use is automatically adjust to variations in the input signal using adaptive filter algorithms. NLMS (Normalised Least Mean Square) adaptive filtering is used as a further modification in LMS (Least Mean Square algorithm to increase the performance of the system. Signal compared with the reference signal and finds noise free signal i.e. noise in the primary input can be essentially eliminated without signal distortion. Kalman Filter is used for both stationary signal and non stationary signal. This uses step gradient method and provides a recursive solution. RLS (Recursive Least Square) is used for recursive solution. Here Comparison of all three adaptive filters algorithms which find the minimum signal to noise ratio. Computer simulations for all cases are carried out using Mat lab software and experimental results are presented that illustrate the*step *usefulness of Simulation & Performance Analysis of ANC using Adaptive Filters.*

Index Terms **-Adaptive Filter, ANC, Kalman Filter, LM, NLMS, RLS.**

I. INTRODUCTION

Adaptive filter used for the non stationary signal. Adaptive filter adjust its coefficient according to change of the signal. It used in many application including system identification, equalization, system modeling image processing and speed processing, communication signal processing, echo cancellation, noise cancellation and many other applications[1]. For noise cancellation in communication we use adaptive noise cancellation. In ANC technique adaptive filter used as a feedback network [3], [10]. It is used for reducing the additive noise which may arise from different sources and it update filter coefficients according to change of the input signal [2]. ANC using adaptive filter algorithm and compare the signal output to the desired signal and then filter out the noise signal by estimating the coefficients according to change of the signal. In ANC noise signal is calculated by the estimation of difference between output and the desired signal. Different algorithm can be use for updating the coefficient of adaptive filter. The performance of filter can be analyses by many factors like mean square error, convergence rate, filter length, stability, complexity etc. The signal having minimum error means high signal to noise ratio having better performance.

II. ADAPTIVE FILTER ALGORITHMS AND THEIR COMPARISON

Adaptive filter algorithms use to update the filter coefficient. The adaptive filter has input x (n), output y (n). The output *y* is (n) generated as a linear combination of the delayed samples of the input sequence *x (n)* according to the equation [8]:-

$$y(n) = \sum_{i=0}^{N-1} w_i(n)x(n-1) \qquad (1)$$

LMS (Least Mean Square)

Most popular adaptive filter algorithm is LMS [3]. The LMS algorithm is extremely simple so that it minimizes the instantaneous square error instead of mean square error [6].

Implementation of LMS algorithm

Each itération of the LMS algorithme have following steps [9] :

- Initialy, set each weight Wn (i), where i= 0,1…. N-1,to an arbitary fixed value such as 0.
- Compute the filtre output

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

(2)

- Compute the error estimate
  e(n)=d(n)-y(n)

(3)

- Update the next filter weights
  w(n+1)=w(n)+2µe(n)x(n)

(4)

Normalised Least Mean Square Algorithm

LMS has drawback as the learning parameter µ increase convergence rate increases but error is more. . So there is a compromise between these two in LMS. The NLMS algorithm [Albert and Gardner] can be considered as a special case of the LMS recursion which takes into account the variation in the signal level at the filter output.The NLMS updates the cofficients of adaptive filtre by using the following équation [5]-[6] :-

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$$\hat{w}(n+1) = \hat{w}(n) + \mu.e(n)\frac{u(n)}{\|\hat{u}(n)\|^2} \quad (5)$$

LMS equation can change as:

$$\hat{w}(n+1) = \hat{w}(n) + \mu(n).e(n).\hat{u}(n) \quad (6)$$

Here $\hat{\mu(n)} = \frac{\mu}{\|\hat{u}(n)\|^2}$ (7)

In the previous equation, the NLMS algorithm becomes the same as the standard LMS algorithm except that the NLMS algorithm has a time-varying step size μ(n). This step size can improve the convergence speed of the adaptive filter.

III. RECURSIVE LEAST MEAN SQUARE ALGORITHM

In adaptive filter RLS is an algorithm which recursively finds the filter coefficients that minimize a weighted linear least squares cost function relating to the input signals. This is in contrast to other algorithms such as the LMS that aim to reduce the mean square error. In the derivation of the RLS, the input signals are considered deterministic, while for the LMS and similar algorithm they are considered stochastic. The RLS algorithm performs at each instant an exact minimization of the sum of the squares of the desired signal estimation errors [2]. These are its equations: To initialize the algorithm **P** (*n*) should be made equal to δ-1 where δ is small positive constant [5], [7].

$$y(n) = \hat{w}^H(n).u(n) \quad (8)$$

$$e(n) = d(n) - y(n) \quad (9)$$

Updates the filter coefficients by using the following equation:

$$\hat{w}(n+1) = \hat{w}(n) + e(n).\hat{k}(n) \quad (10)$$

Where $\hat{w}(n)$ is the filter coefficients vector and $\hat{k}(n)$ is the gain vector $\hat{k}(n)$ is:

$$\hat{k}(n) = \frac{P(n)u(n)}{\lambda + u^H(n)P(n)u(n)} \quad (11)$$

Where λ is the forgetting factor and P (n) is the inverse correlation matrix of the input signal. The standard RLS algorithm uses following equation to update the inverse correlation matrix.

$$P(n+1) = \frac{1}{\lambda}(P(n) - \hat{k}(n)u^H(n).P(n)) \quad (12)$$

Kalman Filter

The Kalman filter is an optimal linear minimum variance estimator. It can provide real-time estimates of the states of a system from noisy measurements. The Kalman filter is a recursive algorithm composed of two parts: Measurement Update Equations and Time Update Equations [4].

- The process equation that defines the evolution of the state with time.
- The measurement equation that defines the observable in terms of the state.

The Kalman filter rooted in the state-space formulation of linear dynamical systems, and provides a recursive solution to the linear optimal filtering problem. It applies to stationary as well as non stationary environments. The solution is recursive in that each updated estimate of the state is computed from the previous estimate and the new input data, so only the previous estimate requires storage. In addition to eliminating the need for storing the entire past observed data, the Kalman filter is computationally more efficient than computing the estimate directly from the entire past observed data at each step of the filtering process.

IV. RESULTS

This describes the simulation results of adaptive filter of noise cancellation which are based on different algorithms and are provided different inputs. These adaptive filters are prepared using different input signals to show that the signal is desired using adaptive filter algorithm i.e. its weights are adjusted itself according to the input signal. The Signal to noise ratio is same for the input signals of different nature and different magnitudes. The models are prepared using different algorithms to make a comparative study of different algorithms. These algorithms are compared by comparing the outputs of different filters. The outputs of different filters being compared to make the comparative study are: noise signal present at the output and signal to noise ratio. Smaller the magnitude of noise signal and smaller the time interval during which signal is present, better the algorithm is. Similarly, higher the signal to noise ratio better the algorithm is.

- **Input Signal: -** Random signal generated which changes forming non stationery signal.

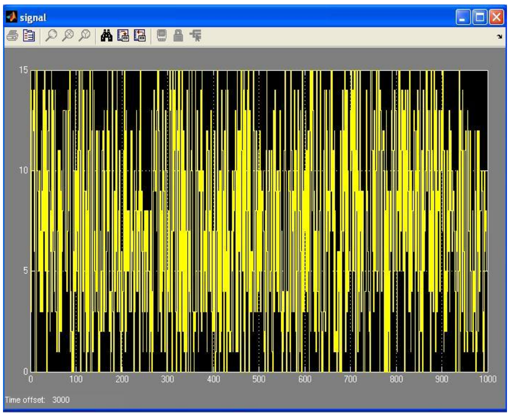

Fig. 1 Input Signals (Random)

- **LMS Results:-**Here NLMs Algorithms is used .NLMS is a special case of LMS .In NLMS step size is time varying.

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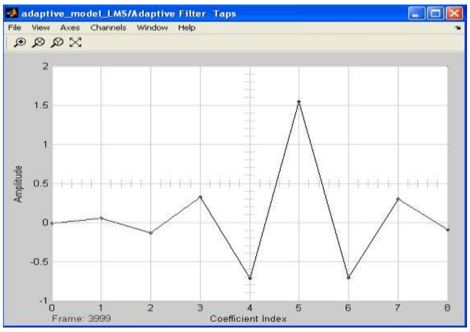


Fig. 2 Adaptive Filter Taps of NLMS

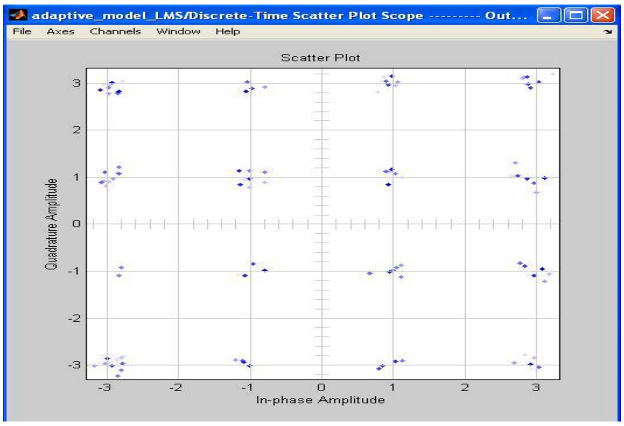


Fig. 3 Discrete time scatter plot of NLMS

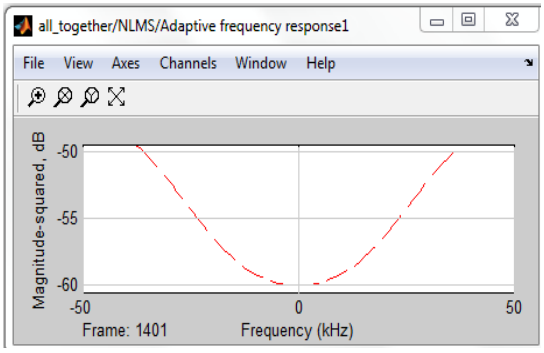
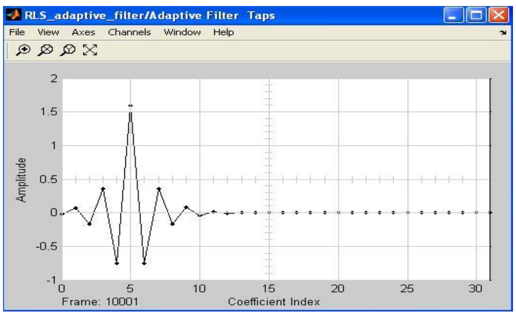


Fig. 4 Adaptive frequency response of NLMS



RLS (Recursive least Mean Square) Results

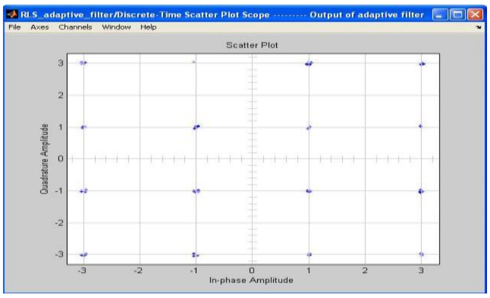


Fig. 6 Discrete time scatter plot of RLS

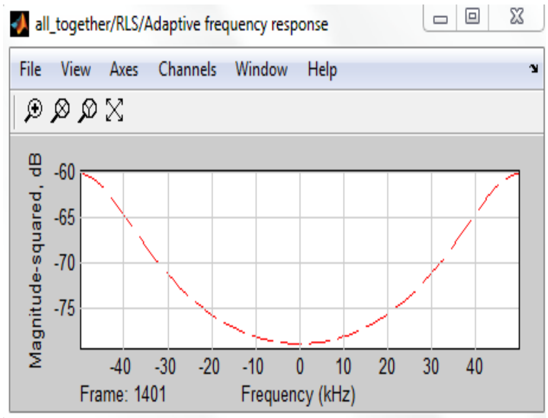


Fig. 7 Frequency Responses of RLS

KALMAN Filter Result:

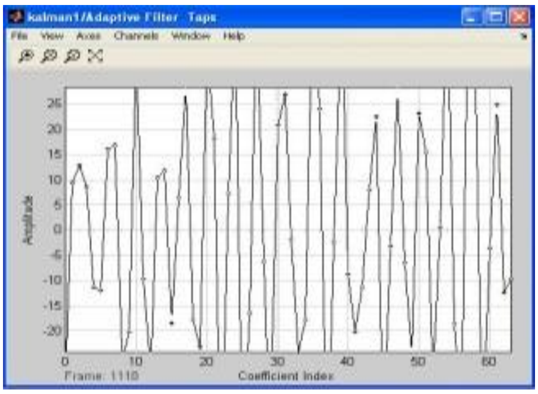


Fig. 8 Adaptive Filter Taps Kalman Filter

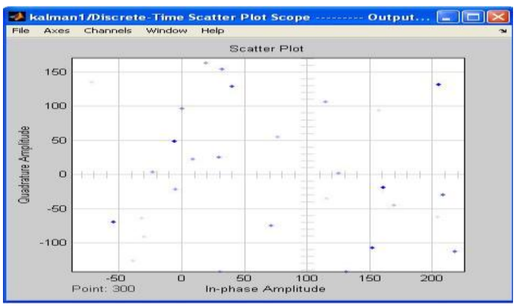


Fig. 9 Discrete time scatter plot of Kalman Filter

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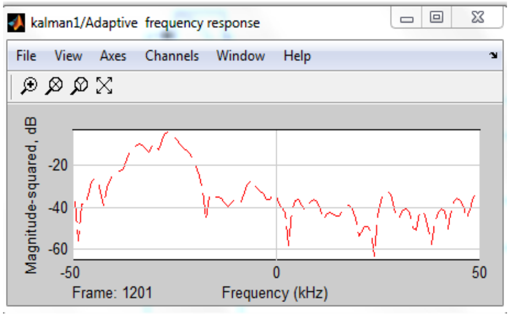

Fig. 10 Frequency response of Kalman

- **Error Signal**

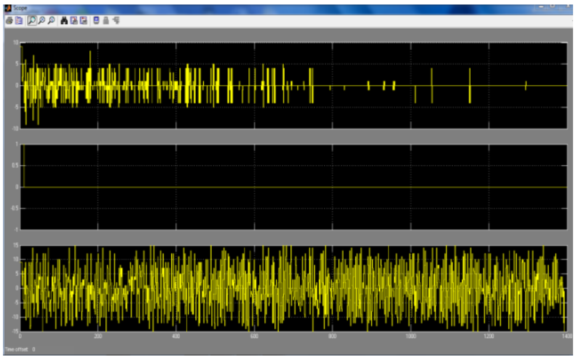

Fig.11 Error Signal by NLMS, RLS, KALMAN

It is observed that all three noise signals are exactly identical. Hence, the noise signal depends on algorithm chosen and parameters of filter. If the filters used in two or three models are same then equalizer performance (noise in output) is also same. As the three adaptive filters are having channels with different parameters so, it is independent of the channel parameters. This also proves that the filter adapts itself with varying channel parameters. This adaptive nature is provided by the adaptive weight adjusting algorithm. The comparative analysis of different algorithms is done by comparing the simulation results of different adaptive filters which are fed with same input signals but are based on adaptive filters with different algorithms. The input signal for all these models is taken as uniform noise generator.

Observations

From the observations, it is concluded that:-

1. RLS algorithm is the best among the three algorithms as it gives noise-free signal for largest time period (90%) and moderate signal for the remaining 10% time.
2. LMS algorithm is good as it gives noise-free signal for large time period (80%), poor signal for 10% of time and very strong signal for the remaining 10% time.
3. Kalman algorithm is the worst as it gives noise-free signal for large time period (80%), poor signal for 10% of time and less moderate signal for the remaining 10% time.

V. CONCLUSION

Here the estimation of the adaptive filter using MATLAB/SIMULINK environment. The behavior of the NLMS, RLS, KALMAN adaptive filters were studied for interference cancellation of signals. Attempts was made to find out the effects of the filter length and the step size parameters of these algorithms. LMS algorithm uses small adaptation coefficient μ ,so the analysis leads to conclude that if the LMS is chosen for interference cancellation of signals, a larger step size chosen, and the filter length should be kept low. The NLMS algorithm, the step size should be smaller and filter length should also be low. But in case a larger step size is chosen, then the filter length should be increased for better performance. Moreover NLMS helps to achieve fast convergence rate, rapid tracking, and low misalignment. But NLMS has problem of stability. Similarly Kalman filter have problem of mean square error and RLS have problem of complexity. But all these algorithms have RLS shows minimum error using less filter length and better convergence rate. However RLS is better, but in general LMS is used due to its simplicity and high stability

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