ISSN: 2454-1311

Evaluation of Adaptive Noise Cancellation using various Algorithm Techniques

Mr. Kulvinder Singh[1], Mr. Kapil Sachdeva[2]

[1]Research Scholar of P.G. College I.I.E.T. kinana, K.U.K. University, Haryana, India

[2]Assistant Professor in Department of ECE, I.I.E.T. College Kinana, jind, Haryana, India

Abstract— *Noise means any unwanted sound. In signal processing or computing noise can be considered as a random unwanted data or signal that does not have any meaning. It means that the noise is not being used to transmit with a signal, i.e. addition to the signal is called noise. Adaptive filter have the ability to adjust their impulse response to filter out the correlated signal in the input. They require little or no a priori knowledge of the signal and noise characteristics that is correlated in some sense to the signal to be estimated. Unlike Non-adaptive or fixed filters have static or fixed filter coefficients and are designed to have a frequency response chosen to alter the spectrum of the input signal in a desired manner. The adaptive algorithm generates this correction factor based on the two factor i.e. input signal and error signals (difference between output and the desired signal). In this paper we proposed LMS and RLS, kalman algorithms define three different coefficient update algorithms. And evaluating these three techniques & choose best one.*

Keywords— ***Noise, Adaptive filter, Matlab software, LMS, RlS, Kalman algorithm.***

I. INTRODUCTION

Adaptive noise cancellation (ANC) techniques for the acquisition of distortion product otoacoustic emissions (DPOAEs). The efficiency of an ANC algorithm for noise suppression was investigated using three microphones: one placed in the test ear, one in the non test ear for internal noise reference; one near the subject's head for external noise reference. Real-time results indicated that the use of an ANC algorithm in combination with standard averaging methods can reduce noise levels by as much as 10 dB beyond that obtained with standard noise reduction methods and probe attenuation alone.

Noise cancellation means to reduce noise signal from the original signal by comparing with the reference signal and the output in which noise component is present with the desired signal. One possible way to satisfy such a requirement to obtain a better recording of the desired signal is the use of a simple noise canceller.

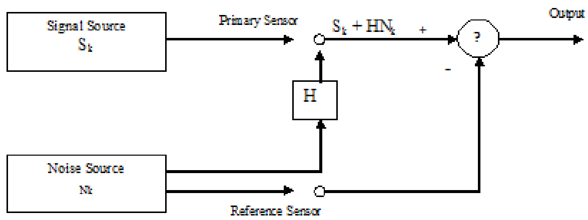

Fig. 1 Block Diagram of Noise Canceller

However, such an approach to subtract the reference noise signal directly from the primary signal is bound to fail because the noise signal at the reference sensor is not exactly the same as the delayed and/or filtered version of noise at the primary sensor. In some cases, this may even lead to an increase in the average power of the noise output. However, when proper provisions are enforced and the subtraction operation is controlled by an adaptive process, superior noise cancellation performance is obtained as compared to the previous approach.

Adaptive Noise Cancellation:

Adaptive noise cancellation is a technique in which noise is reduced by using adaptive filter. Adaptive filters are used especially for non stationary signals and environments. In contrast to the conventional filter design techniques, adaptive filters do not have constant filter coefficients and no priori information is known. Such a filter with adjustable parameters is called an adaptive filter. Adaptive filter adjust their coefficients to minimize an error application of adaptive filters includes adaptive noise cancellation, which is used to remove noise or interference from noisy speech signal. In ANC, the corrupted signal is passed through a filter that tends to suppress the noise while leaving the original signal unchanged.

In this setup, the signal path from the noise source is passed to the primary sensor as an unknown FIR channel H. The adaptive filter to the noise recorded at the reference sensor, and then an adaptive algorithm is used to train the adaptive filter to match or estimate the characteristics of the unknown channel H.If the estimated characteristics of the unknown channel have negligible differences as compared to the actual characteristics, the noise components in the corrupted signal can be cancelled to obtain the desired signal.

II. PROBLEM FORMULATION

The noise cancellation is to estimate the noise signal and to subtract it from original input signal plus noise signal and hence to obtain the noise free signal. There is an alternative method called adaptive noise cancellation for estimating a signal which changes continuously like speech and corrupted by an additive noise or interference. And the reference input of the input signal is adaptively filtered and subtracted from the primary input signal to obtain the estimated signal. Adaptive Noise Cancellation is a technique used to remove an unwanted noise from received signal using adaptive filter. Adaptive filters have the ability to adjust their impulse response to filter out the correlated signal in the input. Adaptive Filters Adjust their coefficients as non stationary signal as the input. For adjusting the coefficients we have different algorithms. Here we compare the algorithms LMS (least mean square) and NLMS (Normalised least mean square), RLS (Recursive Least Square) and Kalman Filter for changing its coefficients according to the change in the signal and estimate the MSE. All these algorithm having different advantage and disadvantage. Here Comparison of all three adaptive filters algorithms which find the minimum signal to noise ratio. The algorithm having minimum signal to noise ratio is near to original signal and having other factor better like less filter length and convergence rate. Computer simulations for all cases are carried out using Mat lab software and experimental results are presented that illustrate the usefulness of Simulation & Performance Analysis of ANC using Adaptive Filters.

III. DESIGN & IMPLEMENTATION

For design and implementation we use simulation using MAT LAB software. Simulink tool is an environment of MATLAB for multi-domain simulation and model-based Design for dynamic and embedded systems. It provides us an interactive graphical environment and a customizable set of block libraries that let helps us to design, simulate, implement, and test a variety of time-varying systems, including communications, controls, signal processing, video processing, and image processing. Adding more products to extend Simulink software as a multiple modeling domains, as well as it provide tools for designing, implementation, and verification and validation of the tasks.

Simulink is integrated with MATLAB, providing immediate access to an extensive range of tools that let us develop algorithms, analyze and visualize simulations, create batch processing scripts, customize the modeling environment, and define signal, parameter, and test data.

Simulink Model For NLMS Algorithms

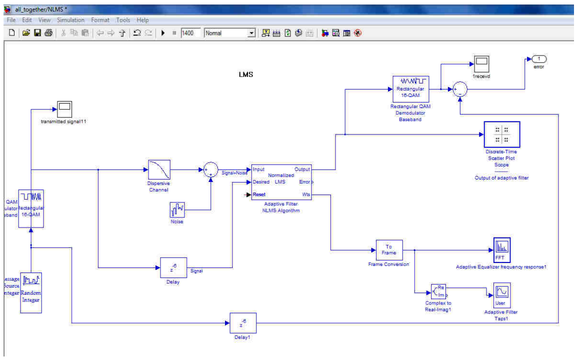

Fig. 2 Simulink Model For NLMS Algorithms

Simulink Model For RLS Algorithms

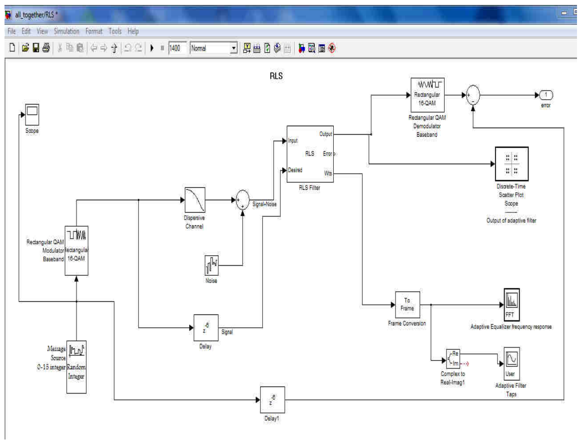

Fig. 3 Simulink Model For RLS Algorithms

Simulink Model For Kalman Algorithms

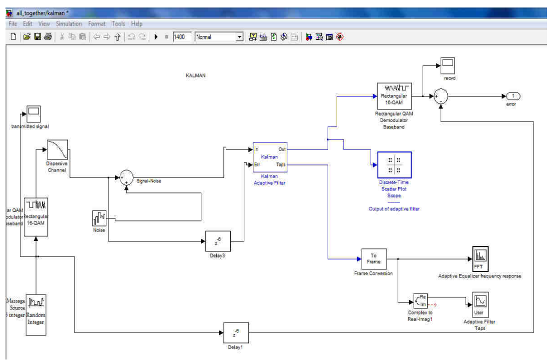

Fig. 4 Simulink Model For Kalman Algorithms

Simulink Model For All Together Algorithms (NLMS, RLS, KALMAN)

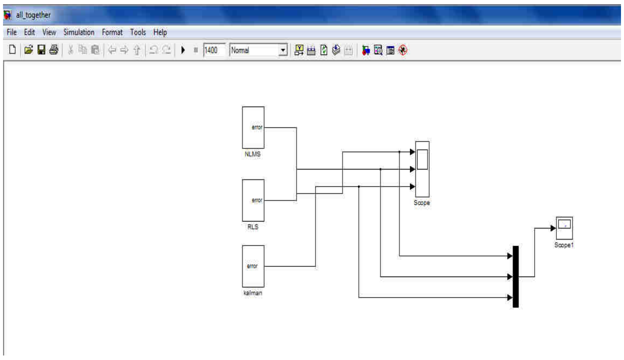

Fig. 5 Simulink Model For All Together NLMS, RLS, Kalman Algorithms

IV. RESULTS & GRAPHS

A. LMS Model Results:-Here NLMs Algorithms is used .NLMS is a special case of LMS. In NLMS step size is time varying which increase the convergence rate of the filter.

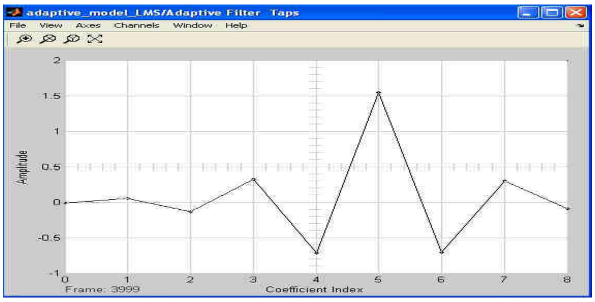

Fig. 6 Adaptive Filter Taps of NLMS

Figure 6 shows the NLMS algorithm tap value of the filter which used for the change of the coefficient.

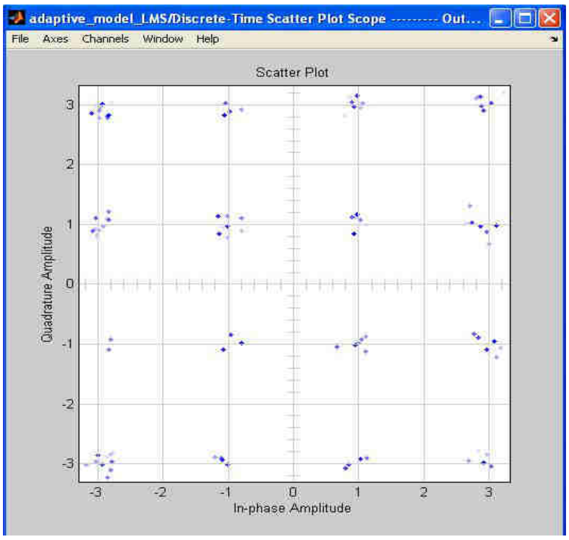

Fig. 7 Discrete time scatter plot of NLMS

Figure 7 shows that the Scatter plots of NLMS is a modulated signal, to reveal the modulation characteristics, such as pulse shaping or channel distortions of the signal.

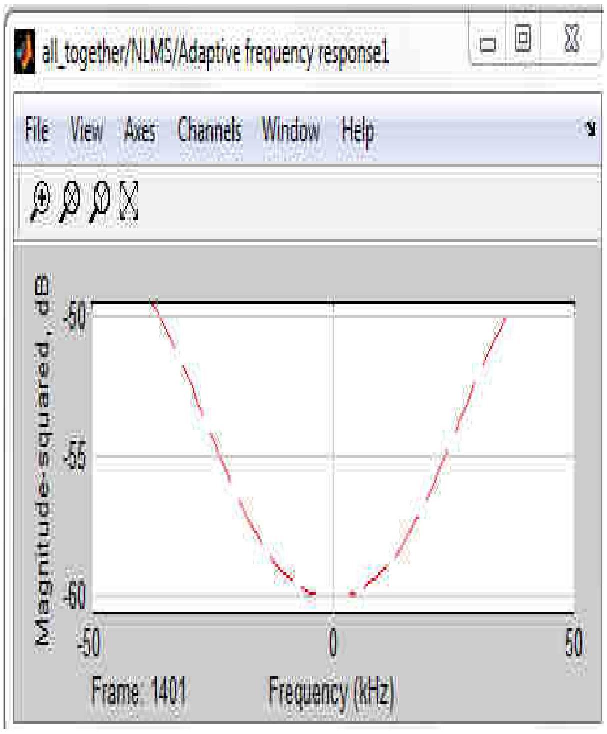

Figure 8 Frequency response of NLMS

Fig. 8 shows the mean square error magnitude in db of NLMS

B) <u>RLS(Recursive least Mean Square) Model Results</u>

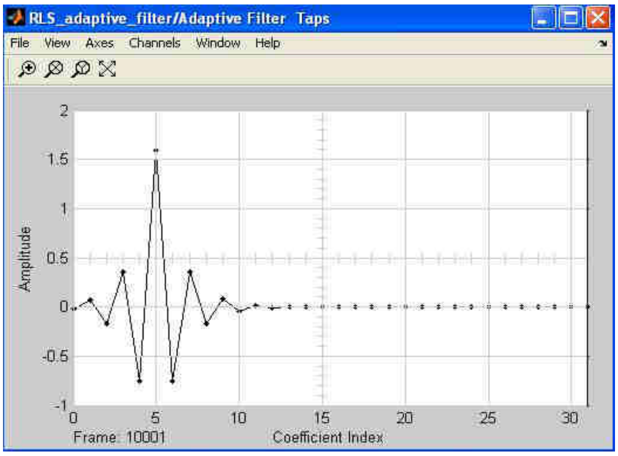

Fig. 9 Adaptive Filter Taps Of RLS

Fig. 9 Shows the RLS Algorithm Tap Value of the Filter which Used For the Change of the Coefficient

ISSN: 2454-1311

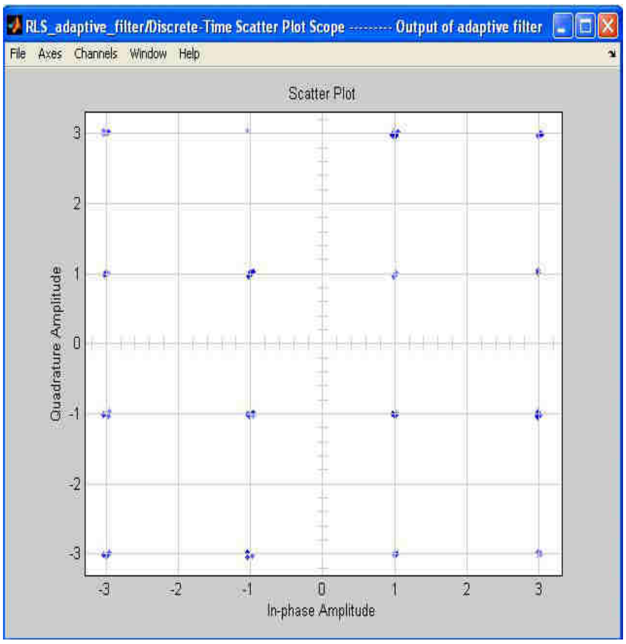

Fig. 10 Discrete time scatter plot of RLS

Figure 10 shows that the Scatter plots of RLS is a modulated signal, to reveal the modulation characteristics, such as pulse shaping or channel distortions of the signal.

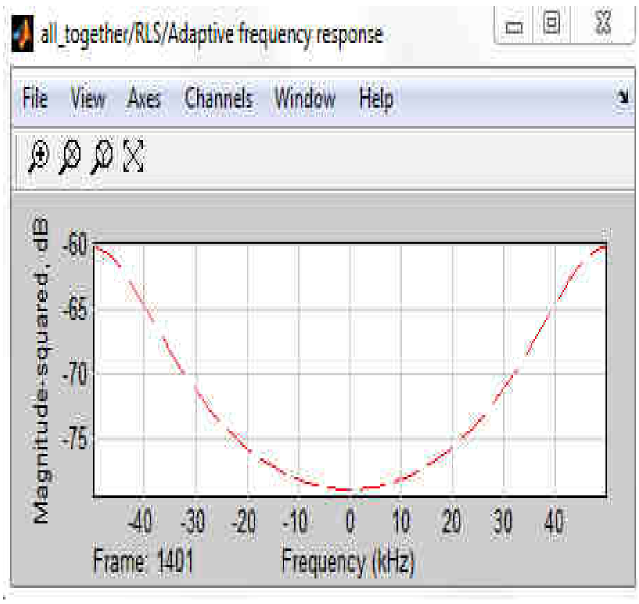

Fig. 11 Frequency response of RLS

Figure 11 shows the mean square error magnitude in db of RLS algorithm

C.KALMAN Filter Model Result:

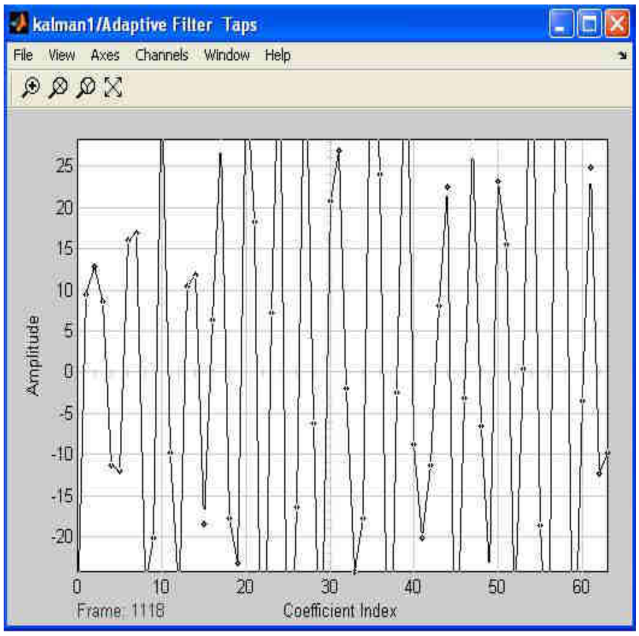

Fig. 12 Adaptive Filter Taps of Kalman Filter

Figure 12 shows the Kalman Filter tap value of the filter which used for the change of the coefficient.

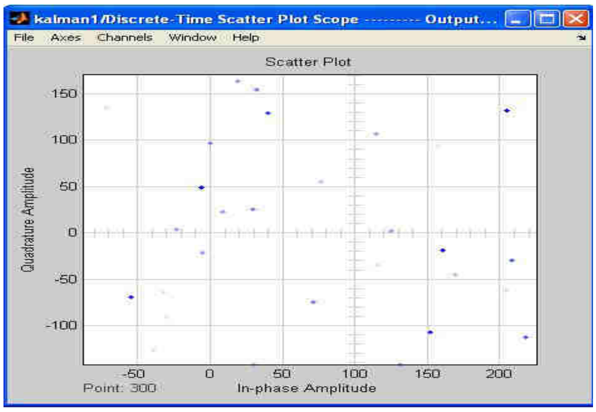

Fig. 13. Discrete time scatter plot of Kalman Filter

Figure 13 shows that the Scatter plots of Kalman filter is a modulated signal, to reveal the modulation characteristics, such as pulse shaping or channel distortions of the signal.

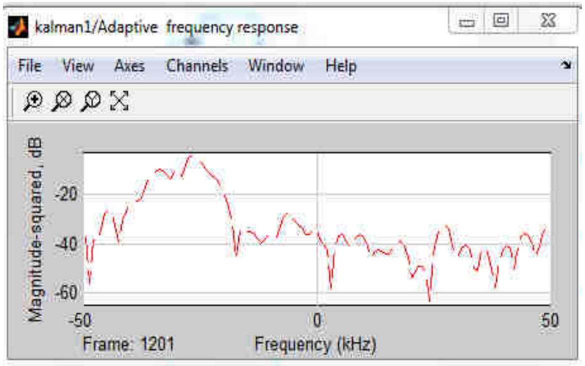

Fig. 14 Frequency response of Kalman filter

Fig. 14 shows the mean square error magnitude in db of Kalman filter algorithm

Error Signal

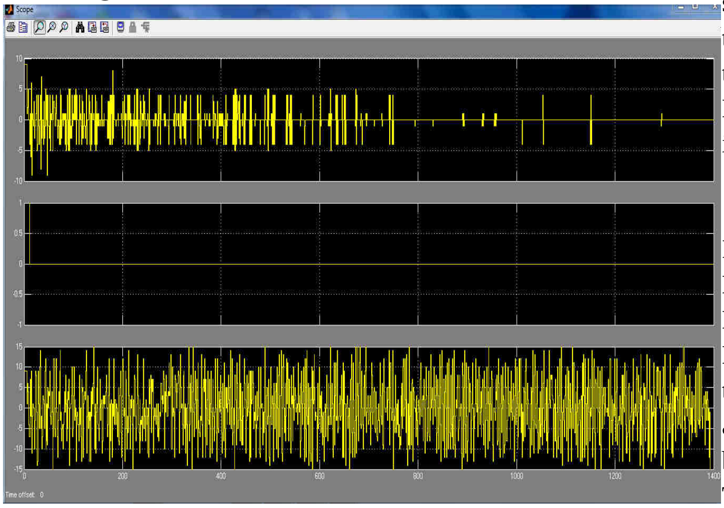

Fig. 15 Error Signal By NLMS,RLS,KALMAN

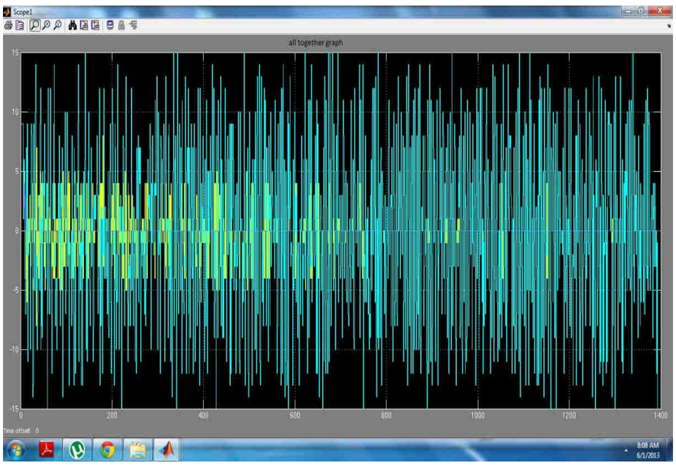

Fig. 16 Error Signal By All Together

To ensure a high-quality product, diagrams and lettering MUST be either computer-drafted or drawn using India ink.

Figure16 captions appear below the figure, are flush left, and are in lower case letters. When referring to a figure in the body of the text, the abbreviation "Fig." is used. Figures should be numbered in the order they appear in the text.

V. CONCLUSION

From the observations, it is concluded that:-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.

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.

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.

However RLS is better, but in general LMS is used due to its simplicity and high stability.

VI. FUTURE SCOPE

In this paper, compare the adaptive filters which include NLMS (Normalised Least Mean Square) special case of LMS algorithm, RLS(Recursive Least Mean Square) and Kalman Filters and find the minimum error algorithm But this minimum error algorithm has problem of complicity and stability. So we need to work on other research for better performance.

To study of other adaptive algorithms need so that their stability for application to Noise Cancellation is more. Modified RLS required so that more stable, faster convergence rate of filter and the reduction in filter length possible.

VII. REFERENCES

[1] Rafael E. Delgado, Ozcan Ozdamar, Syed Rahman, and Carlos N. Lopez “Adaptive Noise Cancellation in Multimicrophone System for Distortion Product Otoacoustic Emission Acquisition” IEEE TRANSACTIONS ON BIOMEDICAL ENGINEERING, VOL. 47, NO. 9, SEPTEMBER 2000.

[2] Paulo A. C. Lopes and Jos´e B. Gerald “*New Normalized LMS Algorithms Based on the Kalman Filter*” Fundac¸ ˜ao para a Ciˆencia e Tecnologia” project POSC/EEA-CPS/59401/2004.

[3] Debi Prasad Das, Swagat Ranjan Mohapatra, Aurobinda Routray,and T. K. Basu “*Filtered-s LMS Algorithm for Multichannel Active Control of Nonlinear Noise Processes*” IEEE TRANSACTIONS ON SPEECH AND AUDIO

PROCESSING, VOL. 14, NO. 5, SEPTEMBER 2006 .

[4] Ali A. Milani,Issa Panahi, Richard Briggs "*LMS-based Active Noise Cancellation Methods for fMRI Using Sub-band Filtering*" IEEE 28th IEEE EMBS Annual International Conference , Aug 30-Sept 3, 2006.

[5] Jashvir Chhikara and Jagbir Singh "*Noise cancellation using adaptive algorithms*" International Journal of Modern Engineering Research (IJMER) www.ijmer.com Vol.2, Issue.3, pp-792-795 ISSN: 2249-6645, May-June 2012.

[6] Jaijit Singh Changotra, Ramanand Harijan "*An Adaptive filter as Noise Cancellation by using LMS /Newton Algorithm*" International Journal on Computer Science and Engineering (IJCSE) ISSN : 0975-3397 Vol. 4 No. 08 0975-3397 Vol. 4 No. 1449,08 Aug 2012.

[7] Jesper Rindom Jensen, Jacob Benesty, Mads Græsbøll Christensen, and Søren Holdt Jensen "*Non-Causal Time-Domain Filters for Single-Channel Noise Reduction*" IEEE TRANSACTIONS ON AUDIO, SPEECH, AND LANGUAGE PROCESSING, VOL. 20, NO. 5, JULY 2012.

[8] Vimala.C and Radha.V "*Optimal Adaptive Filtering Technique for Tamil Speech Enhancement"* International Journal of Computer Applications (0975 – 8887) Volume 41– No.17, March 2012..