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Study and Performance Analysis of Sub-Band Adaptive Noise Cancellation

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***Abstract* - Sub-band adaptive noise is employed in various field like noise cancellation, echo cancellation and system identification etc. It reduces computational complexity and improve convergence rate. In this paper we will perform Sub-band noise cancellation using LMS, RLS. Then compare all these algorithm to find out which one is best.**

***Keywords* - Sub-Band Adaptive Noise Cancellation, LMS and RLS Algorithms.**

I. Introduction

Noise problems have attenuated the growth of telecommunication system. So noise cancellation is a common technology is aimed at reducing unwanted ambient sound. It involved the use of optimum and statistical signal processing techniques to design signal processing that modified the characteristics and achieve a predefined application objects. In noise cancellation, echo cancellation, long distance communication signal involved is very undesirable. So we adapt unknown characteristics which may time variant or time invariant [1]. The main objective of adaptive noise cancellation is to detect an information bearing signal in noise.. Signal processing has become an important tool in almost all fields of science and engineering. In cases and long distance communication, the time variant behavior of the system involved is very undesirable. Therefore, processing techniques should adapt to the unknown characteristics which may be time invariant or variant [3]. The adaptive algorithms should be: simple, computationally efficient, implementable on the existing hardware platform and cost effective, one of the main objectives within adaptive signal processing is noise suppression, i.e., the detection of an information-bearing signal in noise [2].

II. Adaptive Noise Cancellation

The noise cancellation deals with adaptive filtering problem. The estimation error arises from the difference between the target signal and the filter output signal.

The mechanism of iteratively updating filter parameters comes from minimizing a cost function of the estimation errors. And the commonly used cost function is mean-square value or least-square value of the error signal. The parameters of an adaptive filter are usually randomly initialized or based on the a priori knowledge available to the system, and they are refined each time the new input sample is available. The mechanism for adjusting the parameters of an adaptive filter is referred to as an adaptive algorithm. There are various methods for adaptive noise cancellation like LMS and RLS [4]. In figure 1.1, it needed two input- a source signal (Sk +HNk) and a reference noise Nk. Reference noise is correlation with noise added with signal source. The adaptive filter processes the noise signal to produce $\hat{H}$ Nk where $\hat{H}$ is the estimated tap vector for H. This filter output is then subtracted from (Sk +HNk) the output to obtain the estimated desired output signal $\widetilde{Sk}$.

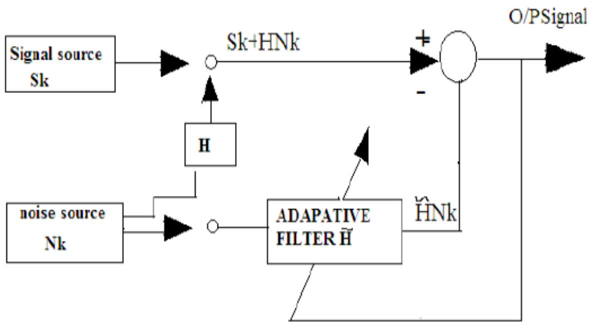

Fig.1.1. Adaptive noise cancellation

III. Adaptive Filter Algorithm

The Least Mean Square adaptive algorithm is the most widely used real time filtering algorithm due to its computing requirements. The adaptive algorithm chosen here is the LMS algorithm, because of its simplicity, hardware efficiency and stability.

A. LMS algorithm

The Least Mean Square (LMS) algorithm, introduced by Widow and Hoff in 1959, is an adaptive algorithm, which uses a gradient-based method of steepest decent. LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms LMS algorithm is relatively simple; it does not require correlation function calculation nor does it require matrix inversions. The LMS algorithm is a type of adaptive filter known as stochastic gradient-based algorithms as it utilizes the gradient vector of the filter tap weights to converge on the optimal wiener solution. It is this simplicity that has made it the benchmark against that all other adaptive filtering algorithms are judged with

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iteration. The filter tap weights of the adaptive filter are updated according to the following formula [5].

$$\widehat{W}(n+1)=\widehat{W}(n+1)+\mu.u(n).e(n)$$

Here x (n) is the input vector of time delayed input values, x (n) = [x(n) x (n −1) x (n-2)… x (n − N +1)] T. The vector w(n) = [w0(n) w1 (n) w2 (n)….w N-1(n)] T represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter μ is known as the step size parameter and is a small positive constant. This step size parameter controls the influence of the updating factor. Selection of a suitable value for μ is imperative to the performance of the LMS algorithm, if the value is too small the time the adaptive filter takes to converge on the optimal solution will be too long; if μ is too large the adaptive filter becomes unstable and its output diverges.

B. RLS algorithm

The RLS algorithms are known for their excellent performance when working in time varying environments but at the cost of an increased computational complexity and some stability problems. In this algorithm the filter tap weight vector is updated using.

$$w(n)=\overline{w}^{t}(n-1)+k(n)e_{n-1}$$

$$k(n)=u(n)/\left(\lambda+x^{t}(n)u(n)\right)$$

$$U(n)=w_{\lambda}^{-1}(n-1)x(n)$$

Where: λ is a small positive constant very close to, but smaller than 1.

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

$$y_{n-1(n)}=w^{t}(n-1)x(n)$$

$$e_{n-1}=d(n)-y_{n-1}(n)$$

In the RLS Algorithm the estimate of previous samples of output signal, error signal and filter weight is required that leads to higher memory requirements [6].

IV. Sub-band Noise Cancellation

Adaptive noise cancellation can be usefully applied in situations where it is required to cancel an interfering noise from a given signal that is a mixture of the desired signal an interference noise. Two useful applications of the adaptive noise cancellation operation are presented in the following section. The places where long-impulse-response filter are needed to employ adaptive filtering in sub band. In sub band adaptive filtering, both the input signal and desired signal are split into frequency sub-band [6]. Each sub-band has shorter impulse response than full band. An adaptive filter is essentially a digital filter with self-adjusting characteristics and tracking capacities. 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. An adaptive filter is essentially a digital filter with self-adjusting characteristics. It adapts, automatically, to changes in its input signals [5]. An adaptive filter consists of two parts: one is a digital filter with adjustable coefficients and another is an adaptive algorithm which is used to adjust or modify the coefficients of the filter [8]. The most important properties of adaptive filter work is that it can work effective in unknown environment, and to track the input signal of time-varying characteristics. Adaptive filter has been used in communications, control etc.

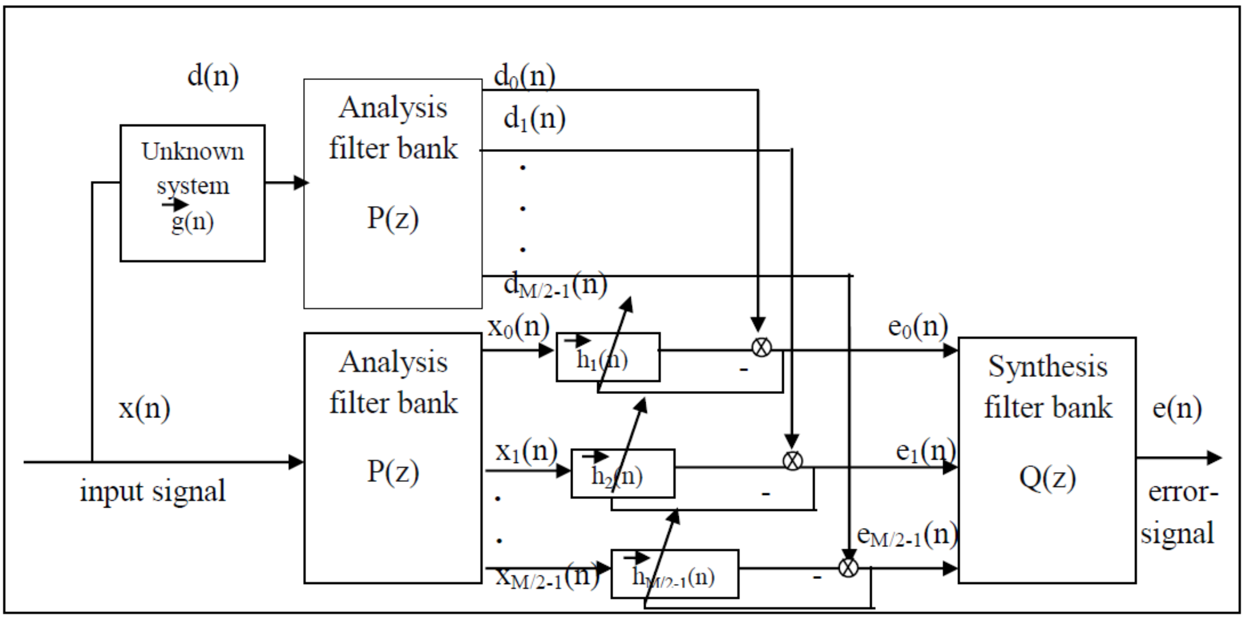

Fig.1.2. Sub-band Adaptive noise cancellation

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A. Deccimation and interpolation

Decimation (also known as down-sampling or compression) of a digital signal x (k) by a factor of L means reducing its sampling rate L times is shown in figure 1.3. Decimation is achieved by retaining only every Lth Sample of the signal. Interpolation (up-sampling) means increasing a signal rate by L times is shown in figure 1.4 [9].

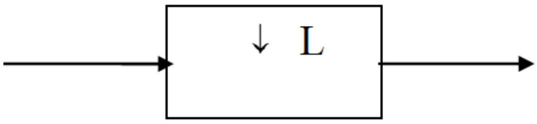

Fig.1.3. Decimation by factor L

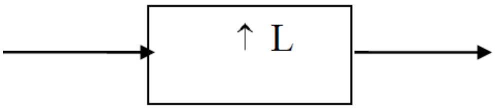

Fig.1.4. Interpolation by factor L

V. Simulation and Result

In this Paper we have to do simulation with Matlab [5] and compare LMS, RLS .From comparison we find that LMS is used for stationary environment and have good tracking but it have very low convergence speed. RLS have very high conversion speed as compare to LMS. Figure 1.5 shows that input wave is a sine wave then it shows a noise +input wave and lastly it shows a wave after cancellation noise.

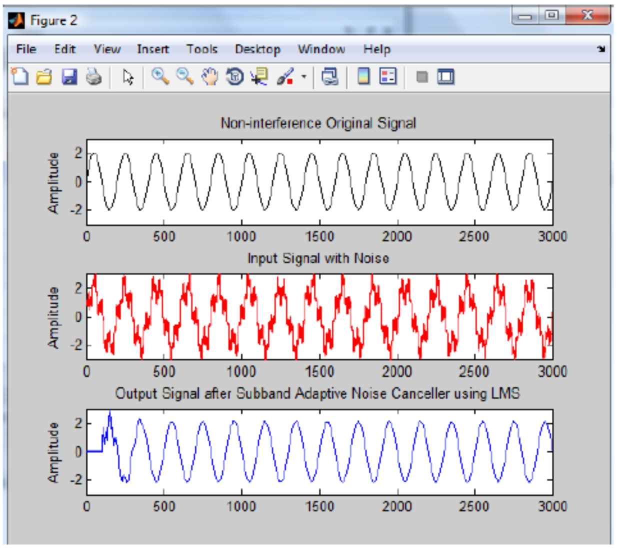

Fig.1.5. Output of subband adaptive noise cancellation.

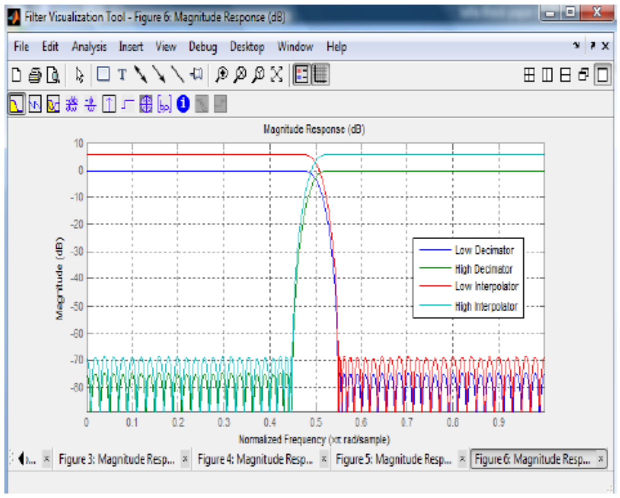

Fig.1.6. Frequency response of subband

VI. Conclusion

RLS is best suited for noise cancellation as compare to LMS because it have best convergence behavioral. But practical implementation of the algorithm have very high complexibility and also have poor numerical properties. we also conclude that subband have high performance as compare to full band noise cancellation.

References

[1] B. Widrow et al., “Adaptive noise cancellation: Principles and applications”. Proc. IEEE, vol. 63, no. 12, Dec. 1975

[2] “Adapative active noise” Sixth international congress on sound and vibration 5-8 July 1999, Copenhagen, Denmark Mosquera, J.A. Gomez, F. Perez, M. Sobreira

[3] Comparative Tracking Performance of the LMS and RLS Algorithms for Chirped Narrowband Signal Recovery Paul C. Wei, *Member, IEEE*, Jun Han, *Student Member, IEEE*, James R. Zeidler, *Fellow, IEEE*, and Walter H. Ku, *Member, IEEE* Transactions on signal processing VOL. 50. NO. 7, JULY 2002

[4] “An Efficient Adaptive Noise Cancellation Scheme Using ALEand NLMS Filters” Jafar Ramadhan Mohammed1, Muhammad Safder Shafi2, Sahar Imtiaz2, Rafay Iqbal Ansari2, and Mansoor Khan International Journal of Electrical and Computer Engineering (IJECE) Vol.2, No.3, June 2012.

[5] Shadab Ahmad, Tazeem Ahmad “Implementation of Recursive Least Squares (RLS) Adaptive Filter for NoiseCancellation” IJSET Volume No.1, Issue No.4, pg : 46-48 01 Oct. 2012

[6] S. Sandeep Pradhan and V. U. Reddy “A New Approach to Subband Adaptive Filtering”IEEE Transaction On Signal Procesing, Vol. 47, NO. 3, March 1999.

[7] Acoustic Noise Cancellation Using Sub-Band Decomposition and Multirate Techniques Thamer M.J. Al-Anbaky University of Technology-Dept. of Electrical & Electronic Eng.-Baghdad/Iraq

[8] “A Review of Advances in Subband Adaptive Filtering” Ali O. Abid Noor, Salina Abdul Samad and Aini Hussain World Applied Sciences Journal 21 (1): 113-124, 2013.

[9] Richard Briggs “LMS based active noise cancellation method using Sub-band”IEEE annual international conference New York city USA,Augest 30-September 3,2006.

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