Comparison of different Sub-Band Adaptive Noise Canceller with LMS and RLS

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Abstract—Sub-band adaptive noise is employed in various fields like noise cancellation, echo cancellation and system identification etc. It reduces computational complexity and improve convergence rate. In this paper we perform different Sub-band noise cancellation method for simulation. The Comparison with different algorithm has been done to find out which one is best.

Key words: Sub Band Adaptive Noise Cancellation with LMS, NLMS 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. NOISE

Noise means any unwanted signal. 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. There are different types of Noise explained below:

A. *White noise:*

White noise is a random signal or process with a uniform power spectral density, so it is a noise in which the frequency and power spectrum is constant and independent of frequency.

B. *Thermal noise*

That type of noise occurs in all transmission media and communication equipment, including passive devices. It arises from random electron motion and is characterized by a uniform distribution of energy over the frequency spectrum with a Gaussian distribution of levels.

C. *Shot noise*

Shot noise normally occurs when the variations in the electrical current produced by electrons emitted from a cathode. Shot noise is always associated with current flow, It stops when the current flow. Shot noise is independent of temperature. Shot noise is spectrally flat or has a uniform power density, meaning that when plotted versus frequency it has a constant value. Shot noise is present in any conductor not just a semiconductor.

D. *Atmospheric Noise*

Atmospheric noise is radio noise caused by natural atmospheric processes, primarily lightning discharges in thunderstorms.

E. *Flicker noise or 1/f noise*

The 1/f noise is found in many natural phenomena such as nuclear radiation and electron flow through a conductor. All systems contain some form of white noise, but many practical systems are also plagued by additional low-frequency noise components. When this low frequency noise has the common form in which the noise spectral density increases as the inverse frequency one speaks about 1/f noise.

F. *Impulse noise*

In that type of noise a non continuous series of irregular pulses or noise "spikes" of short duration, broad spectral density and of relatively high amplitude.

III. 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 inputs- 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 $H\hat{}$ 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[11].

IV. 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.

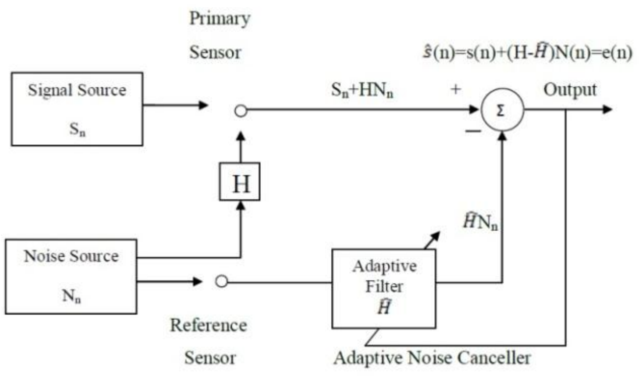

Fig. 1: Adaptive noise cancellation

A. LMS algorithm

The Least Mean Square (LMS) algorithm, introduced by Widrows 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 bench mark against that all other adaptive filtering algorithms are judged with iteration. The filter tap weights of the adaptive filter are updated according to the following formula [5]. 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. NLMS algorithm

In the standard LMS algorithm, when the convergence factor μ is large, the algorithm experiences a gradient noise amplification problem. In order to solve this difficulty, we can use the NLMS (Normalized Least Mean Square) algorithm. The correction applied to the weight vector w(n) at iteration n+1 is "normalized" with respect to the squared Euclidian norm of the input vector x(n) at iteration n [13]. The NLMS algorithm as a time-varying step-size algorithm

$$\mu(n) = \frac{\alpha}{c+\|x(n)\|^2} \quad (4.1)$$

Where: α is the NLMS adaption constant, which optimize the convergence rate of the algorithm and should satisfy the condition 0< α<2, and *c* is the constant term for normalization and is always less than 1.

$$W(n+1) = w(n) + \frac{\alpha}{c+\|x(n)\|^2} e(n)x(n) \quad (4.2)$$

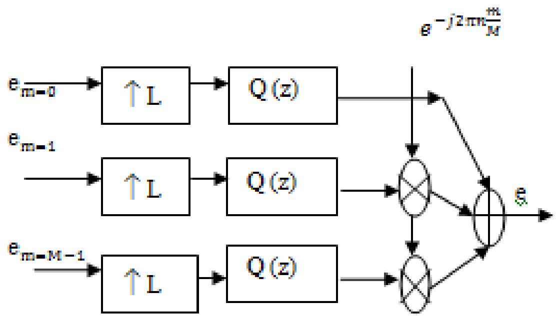

Fig. 2: Synthesis filter bank decomposition

C. 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. The RLS algorithm performs at each instant an exact minimization of the sum of the squares of the desired signal estimation errors [3]. These are its equations: To initialize the algorithm **P** (*n*) should be made equal to δ-1 where δ is a small positive constant. **[12]**

$$y(n) = \hat{w}(n).u(n) \quad (4.3)$$
$$\hat{w}(n) = \hat{w}(n-1) + \alpha^*(n) \quad (4.4)$$

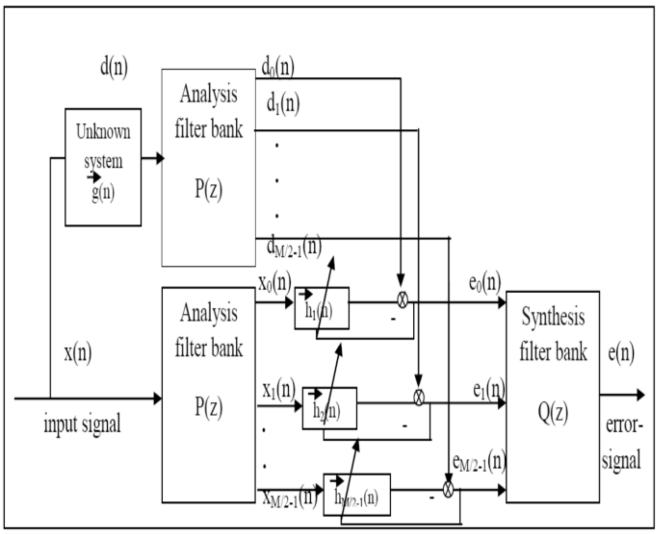

Fig. 3: Sub-band Adaptive noise cancellation

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 [4].

V. 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 full band adaptive filter is not used so we have 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.[14].

A. Decomposition Synthesis Filters Bank

The design of synthesis filters banks (Fig.5) reduces the design of a single synthesis prototype filter Q(z)[10]. When the designing the synthesis filters bank, the focus on the performance of the analysis filter bank as a whole.

VI. SIMULATION & RESULTS

1) By simulation of sub band adaptive noise cancellation in matlab we have to conclude that RLS algorithm is the best as compare to LMS.
2) As we increase the order of sub band then error signal will be low so signal output will be more correct. I.e. in 2order,4order and 8 order sub band noise cancellation 8 order sub-bands have high output and lowest error.
3) We also compare the LMS, NLMS and RLS with different values of μ.

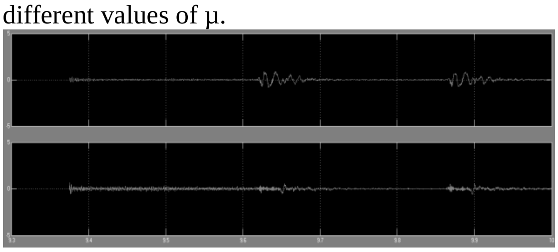

Fig. 4: Output, Error signal with LMS Algorithm for step size=0.1

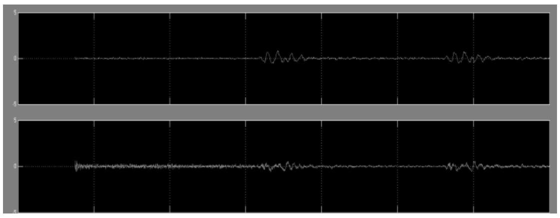

Fig. 5: Output, Error signal with LMS Algorithm for step size=0.01

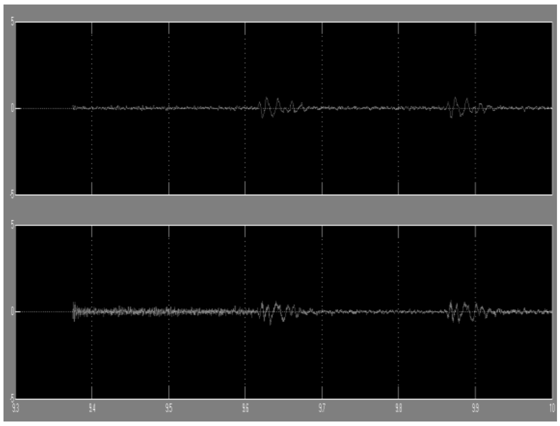

Fig. 6: Output, Error signal with LMS Algorithm for step size=0.001

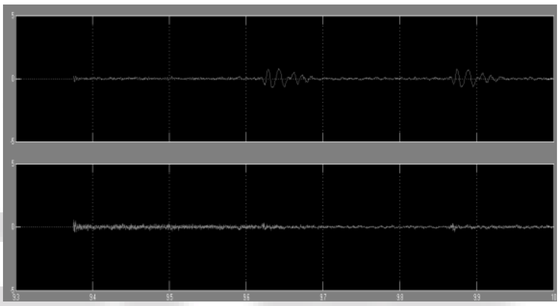

Fig. 7: Output, Error signal with RLS Algorithm for step size=0.1

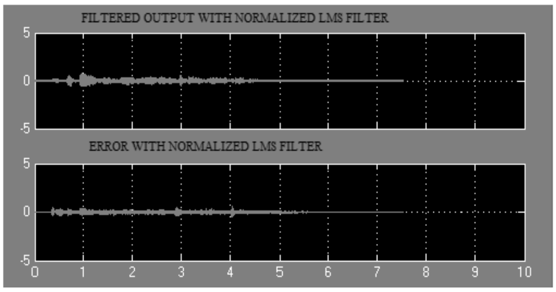

Fig. 8: Output, Error signal with Normalized LMS Algorithm for step size=0.001

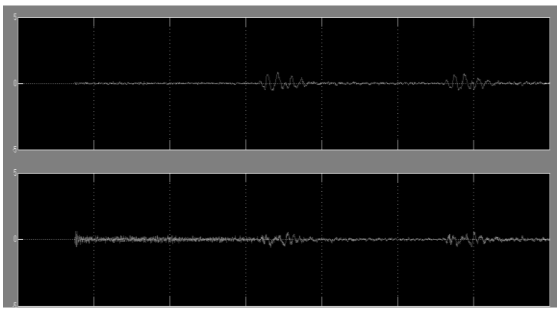

Fig. 9: Output, Error signal with Normalized LMS Algorithm for step size=0.01

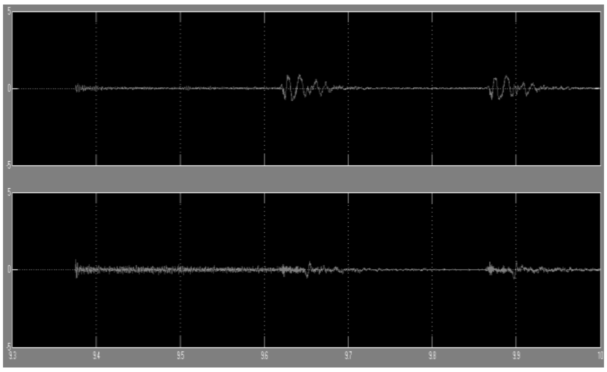

Fig. 10: Output, Error signal with Normalized LMS Algorithm for step size=0.01

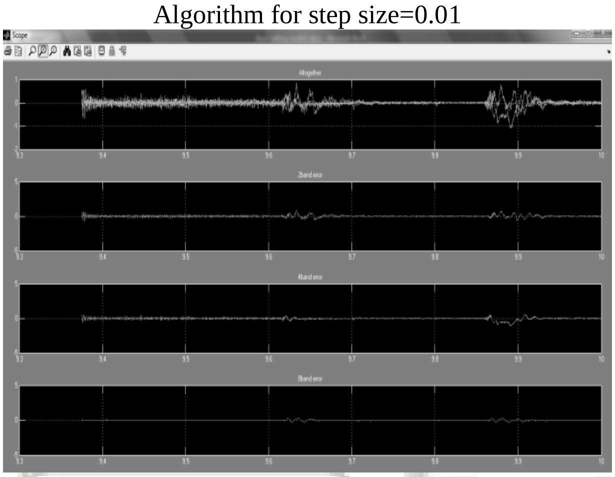

Fig. 11: Error signal with Normalized LMS Algorithm for 2 order, 4 orders and 8 order subband adaptive noise cancellation.

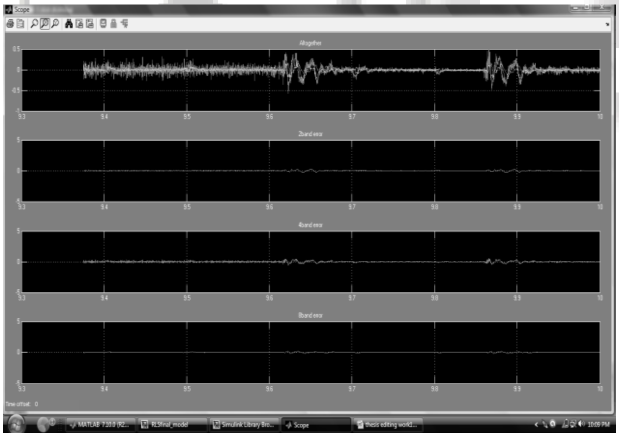

Fig. 12: Error signal with RLS Algorithm for 2nd, 4th and 8th order sub band adaptive noise cancellation

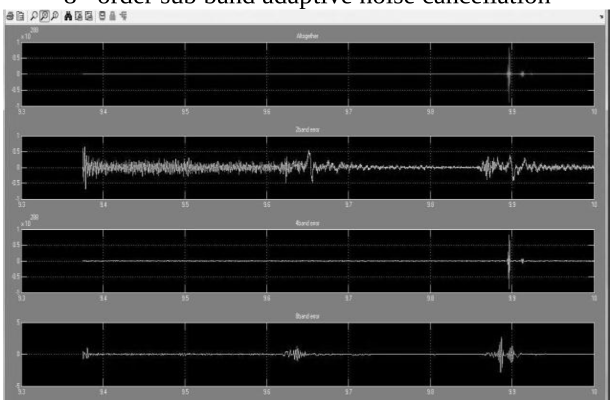

Fig. 13: Error signal with NLMS Algorithm for 2nd, 4th and 8th order sub band adaptive noise cancellation

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