Noise Cancellation in Communication System Using Adaptive Filters by RLS Algorithm

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Abstract : In practical , the characteristics of signal and noise are unknown so that we hardly design a coefficient digital filter. An adaptive filter is a special category of filter that automatically changes its parameters in harmony with input signal by keeping error into account, using some optimizing algorithms. Adaptive filters find applications in various fields such as, noise and echo cancellation, system identification, channel equalization, adaptive inverse system configuration and adaptive linear prediction and so on. This research employment is based on the implementation of adaptive filtering algorithms for noise and echo cancellation Least Mean Square (LMS), Normalized Least Mean Square(NLMS), Recursive Least Square (RLS) algorithms. MATLAB is used as a Software tool. In allusion to this problem, the theory of the adaptive filter and adaptive noise cancellation are researched deeply. The Recursive Least Squares (RLS) algorithms are used in noise cancelling, comparing and analysing the result. It gradually adjusts the error functions and increases the speed of convergence. Compared to most of its competitors, RLS exhibits extremely fast convergence. However, this benefit comes at the cost of high computational complexity. This has been observed by simulation results.
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***IndexTerms-* Adaptive filters, Adaptive algorithm, RLS, noise, simulation, convergence rate, MATLAB.**

I. INTRODUCTION

When digital communication operates in higher bandwidth or frequency range there are chances of generating Inter- Symbol Interference (ISI) [1]. To overcome this problem of ISI, Finite Impulse Response (FIR) filters and Infinite Impulse Response (IIR) filters are used [2]. FIR Filters are suitable in the case of no feedback or if the system is linear [3]. If the system is nonlinear then IIR filters are suitable [4]. IIR filters work elegantly for nonlinear signals only if and if not producing error surface [5]. In the process of communication systems, often to deal with some unforeseen signal, noise or time-varying signals, if only by a two FIR and IIR filter of fixed coefficient cannot achieve optimal filtering[3]. An adaptive filter adjusts its frequency response automatically to improve its performance. Owing to the self- adjusting performance and in-built flexibility, adaptive filters are used in diverse applications.

We use adaptive filters where it is must for the filter characteristics to be variable and adapted to changing condition and when there is a spectral overlap between the signal and noise. Adaptive algorithms are used to modify the coefficients of digital filter[8]. The frequency and space diversities require a bandwidth overhead. But in general, bandwidth is quite expensive. These signal diversity techniques were used in analog radio and are easily adapted to digital systems that undergo highly selective interference. The amplitude equalizers are designed to flatten the received spectrum to correct the spectral shape. An amplitude equalizer is often used in conjunction with frequency or space diversity. This can provide a sufficient equalization for specific channels. However, to adequately characterize the effects of all channel types, the multichannel equalizer is adopted.

II. RELATED WORK

To obtain better MSE and improved convergence rate RLS algorithm is used. The RLS algorithm performs exact minimization of the sum of the squares of the desired signal estimation errors at every instance[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[2].

The following block diagram describes the overall working of a signal (sine wave) when passed into the noise cancelation blocks and also through the adaptive filters. The error and the noise signals are detected by using the appropriate adaptive algorithms in the communication system to obtain the required desired signal at the output.

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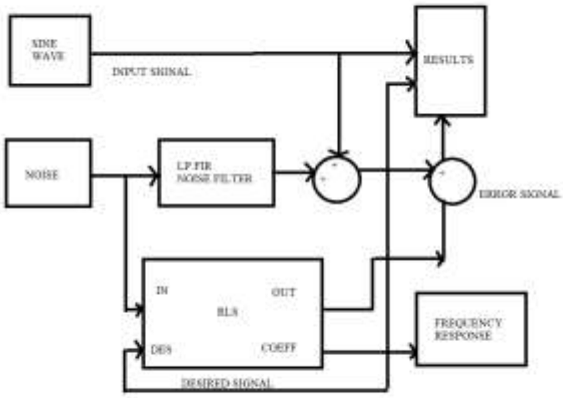

Fig. 1 Block diagram adaptive filter for noise cancellation

$$y(n) = w^H.u(n)$$

$$\propto (n) = d(n) - w^H(n-1)u(n)$$

$$\pi(n) = u^H.\text{P(n-1)}$$

$$k(n) = \lambda + \pi(n).u(n)$$

$$k(n) = \frac{P(n-1).u(n)}{k(n)}$$

$$w(n) = w(n-1) + k(n).\propto^* (n)$$

$$P(n-1) = k(n).\pi(n)$$

$$P(n) = \frac{1}{\lambda}(P(n-1) - P(n-1))$$

The RLS algorithm is based on the least squares. The RLS algorithm has fast convergence than LMS algorithm. It has more complexity than LMS but high convergence speed, makes it pretty good choice[7].

III. Proposed RLS Algorithm in the Cancellation of Noise in the Communication System

The proposed paper shows how to use an RLS filter to extract useful information from a noisy signal. The information bearing signal is a sine wave that is corrupted by addictive white Gaussian noise.

The adaptive noise cancellation system assumes the use of two microphones. A primary microphone in the communication system picks up the noisy input signal, while secondary microphone receives the noise that is uncorrelated to the noise picked up by the primary microphone.

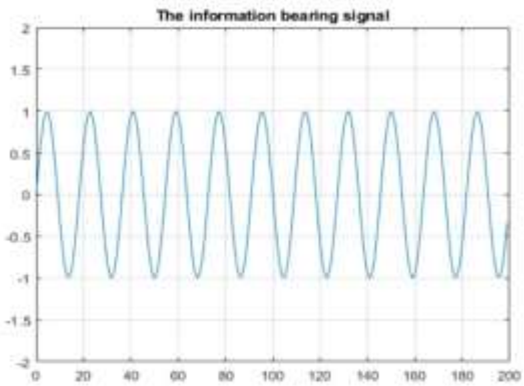

Fig. 2 Message signal carrying information

The noise picked up by the secondary microphone is the input for the RLS adaptive filter(fig.3). The noise that corrupts the sine wave is a lowpass filtered version of this noise. The sum of the filtered noise and the information bearing signal is the desired signal for the adaptive filter.

The secondary filter receives the output from the primary microphone and detects the noise combined in the message signal and tries to eliminate it using the adaptive filter which is performed by RLS algorithm.

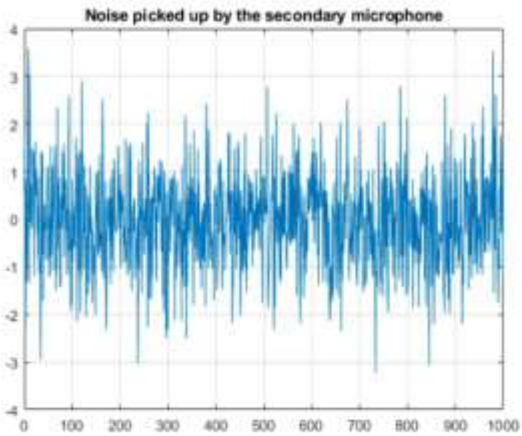

Fig. 3 Noise recognised by the secondary signal

The noise corrupting the information bearing signal is a filtered version of ‘noise’.After running the RLS adaptive filter for 1000 iterations(fig.4). Though the adaptive filter converges, the filtered noise should be completely subtracted from the “signal+noise”. Also the ‘e’, should contain only the original signal.

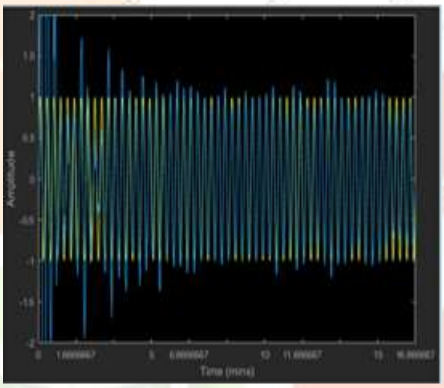

Fig. 4 Simulated result of the noise cancellation in the message signal

The plot shows the convergence of the adaptive filter response to the response of the FIR filter(fig.5).

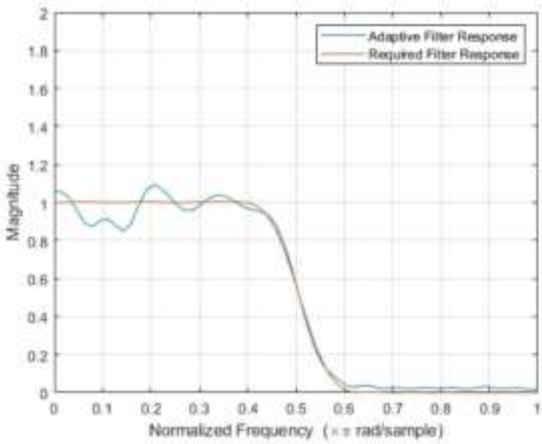

Fig. 5 Plot comparing adaptive and required filter response

IV. Conclusion

In this paper, we have given analysis on the convergence in the adaptive filter response to the response of the filter. Analysis and computer simulation show that the RLS algorithm is quite sensitive to the power of the measurement noise and provides better tracking capability in a communication system. The performance comes at the cost of computational complexity and considering the large FIR order required for echo cancellation, this is not feasible for real time implementation. In future we can also perform echo cancellation using other adaptive algorithms. The adaptive filters are mostly used where the statistical parameters of the system are unknown or in a non-stationary environment. The use of adaptive filtering provides us new signal processing capabilities which were not possible with ordinary fixed filters

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