

Noise Cancellation in Communication Systems using LMS and RLS Algorithms

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Abstract: A review of noise cancellation in communication systems based on adaptive filter algorithms is presented. The modifying of signal characteristics in noise cancellation is relatively fast which involves the utilization of adaptive algorithms for quick coverage. The LMS and RLS algorithms play a vital role in noise cancellation to increase the convergence rate. Current communication applications require reducing the computational complexity in real-time implementation. The category of adaptive filters is automatically changing the parameters of its algorithms according to the input signal. Many adaptive filters are digitally executed to update the algorithm parameters. The original applications of an adaptive filter in numerous fields including system recognition and noise cancellation can be modeled by MATLAB simulator. The input signal of an adaptive filter is with the mixed algorithms and noise signal in the specific design of MATLAB to remove the noise from the original data and produce reliable data at the output port. The filter implementation shows the strength of the suggested structure and the capabilities of the proposed algorithms have been investigated and analyzed. The performance of the filter is promising to support the current and future communication systems in terms of noise cancellation.

Keywords: Noise Cancellation, Adaptive Filter, LMS Algorithm, RLS Algorithm, Communication System.

1.Introduction

Noise cancellation is an important technique in digital signal processing used to remove unwanted noise from a corrupted signal and recover the original information signal. In many communication systems, transmitted signals are often affected by various types of interference and environmental noise during transmission [1-3]. This unwanted noise degrades the quality of the received signal and affects the overall performance of the communication system. Therefore, effective noise cancellation methods are essential to improve signal quality and ensure reliable communication. Adaptive noise cancellation has gained significant attention in recent years because of its ability to automatically adjust filter parameters according to changing signal conditions. Unlike fixed filters, adaptive filters continuously update their coefficients to minimize the error between the desired signal and the filter output. Because of this adaptive capability, noise cancellation has become a fundamental technique used to eliminate noise associated with the original signal [4-7]. In the process of data transmission, the noise is automatically added to information from source to receiver part. The principle algorithms of noise cancellation have explained as illustrated in Figure 1.



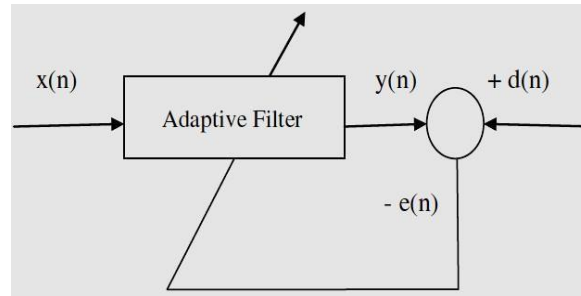


Figure1: Fundamental of Adaptive algorithms

The presence of noise in a communication channel significantly reduces signal quality [8-9]. As a result, suppressing and eliminating noise has become a major research area in signal processing. Early research introduced adaptive noise cancellation techniques to address this problem, and since then several methods and algorithms have been developed to improve system performance [10-12].

The primary goal of these algorithms is to process the input signal to get the pure signal without noise. In this manner, to remove the distortion from the original waveform, the useful signal is subtracted from the filtered signal. To minimize the noise signals, the algorithms are driven by the error signal to produce the filter coefficients. By using an adaptive filter in communication systems, an excellent solution could offer when signal processing requires significant improvement against fixed filter in conventional procedure. The objectives of this work are to review the current techniques of noise optimization and develop the performance of LMS and RLS algorithms

To select the adaptive filter, the following factors should be taken into account:

- a. Convergence rate
- b. Structure of the filter
- c. Requirements for computation
- d. Robustness of the filter

Figure 2 illustrates the basic structure of an adaptive noise cancellation system. In this system, a reference signal containing noise is provided to the adaptive filter. The adaptive filter processes this reference signal to estimate the noise component present in the primary input signal. The estimated noise is then subtracted from the primary input to generate an error signal, which represents the filtered output signal with reduced noise.

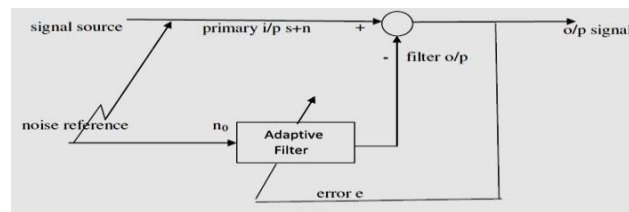


Figure2: The setup of adaptive noise cancellation

This paper proposes the algorithms shown in Figure 3 to optimize the noise in the channel which supports the task of an adaptive filter in noise cancellation. The rapid growth in most communication systems requires efficient algorithms to control the noise and attenuation in the order that produced by different sources.

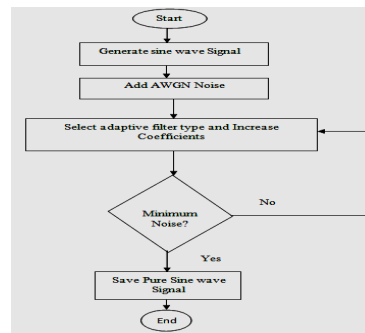


Figure3:Proposed algorithms for noise cancellation optimization

2. Literature Review

Noise cancellation in communication systems has been an important research topic in digital signal processing due to the increasing demand for high-quality and reliable signal transmission. In many communication environments, transmitted signals are affected by unwanted noise and interference during the transmission process. These disturbances degrade the quality of the received signal and reduce system performance. Therefore, effective noise reduction techniques are required to suppress noise and improve signal clarity. Early research in this area mainly focused on the use of fixed filtering techniques such as low-pass, high-pass, and band-pass filters. Although these filters are simple to implement and computationally efficient, they are only effective for stationary noise conditions and cannot adapt to time-varying noise environments.

Several studies have compared the performance of LMS and RLS algorithms in noise cancellation systems. These comparisons are generally based on parameters such as convergence speed, signal-to-noise ratio (SNR) improvement, mean square. The LMS algorithm is one of the simplest adaptive algorithms used in signal processing. It adjusts filter coefficients iteratively to minimize the mean square error between the desired signal and filter output.

The RLS algorithm is another powerful adaptive filtering technique. It provides faster convergence and better performance compared to LMS but requires higher computational complexity.

3. Noise Control Using Fir-LMS Adaptive Filter

The reduction of the unwanted amount of noise in the channel can be achieved by the active noise control system using FIR-LMS adaptive filter algorithms. Typically, the noise signal is introduced by a device chain in the system or from channel medium.

As a result, the noise close to the data source should be determined and then suppressed by generating an anti-noise signal equal to an error signal using adaptive filter algorithms. The band-limited range from 80KHz to 300KHz which represents the baseband signal of many communication systems with specific filter length could be simulated by MATLAB command window to get filter response as shown in Figure 4.

To estimate the secondary path of propagation, the noise controller response showing in Figure 5 is written in MATLAB code. Classically, the filter length in most cases could not be for sufficient control. This figure illustrates the coefficients of actual and estimated propagation path to demonstrate the accuracy of secondary path response. Through the operation of the filter, the performance of a noise control system does not harm the residual error.

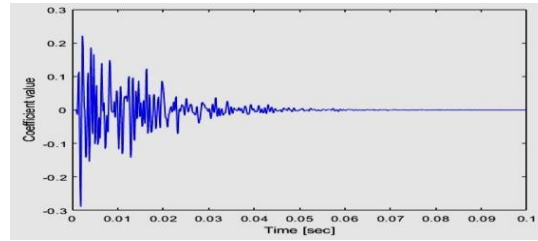


Figure4:FIRfilterresponsefornoisecontrol

The applications of the noise control system involve the coming noise by channel. To emphasize the difference between the original and noise signals, the adaptive filter has been enabled. The estimated time delay model shown in Figure 6 is designed and simulated in SIMULINK block set, and the waveforms of mixed signal are illustrated in Figure 7 and figure 8.

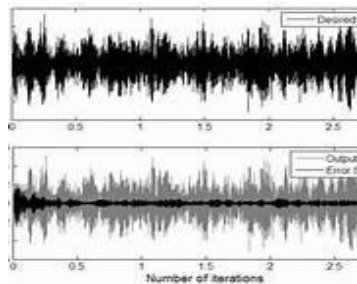


Figure 5: Secondary identification of the adaptive filter

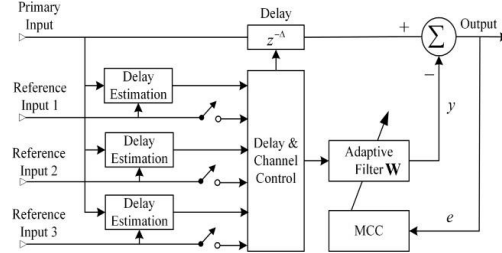


Figure 6:DelayTimeEstimationModel

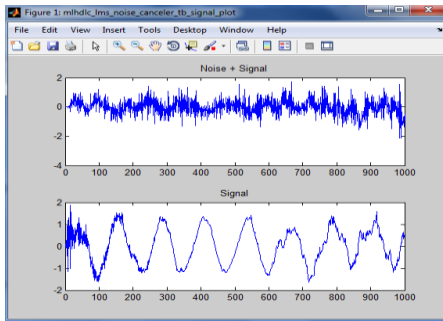


Figure 7: Noisy and Original Signal Estimated by LMS filter

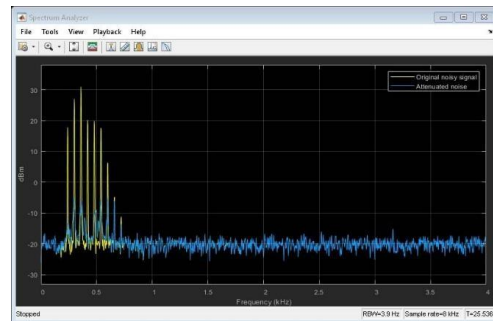


Figure7:Original and noise signal estimated by the adaptive filter

4. NoiseControlUsingRLSAdaptiveFilter

To extract useful information from a noisy signal, the Recursive Least Squares (RLS) adaptive filtering algorithm is used. In communication systems, the transmitted signal is often corrupted by noise during transmission. In this work, the original signal is represented by a sine wave mixed with Additive White Gaussian Noise (AWGN), which introduces errors and degrades signal quality.

The RLS algorithm estimates and removes the noise component by minimizing the error between the desired signal and the filter output. The adaptive noise cancellation system consists of a primary input (signal + noise), reference noise input, adaptive filter, and error output. The adaptive filter processes the reference noise signal and generates an estimate of the noise, which is subtracted from the primary input to produce the filtered output.

The adaptive noise cancellation block diagram is shown in Fig. 9, and the system model is implemented using the SIMULINK block set in MATLAB. This simulation helps analyze the performance of the RLS adaptive filter in reducing noise and improving signal quality.

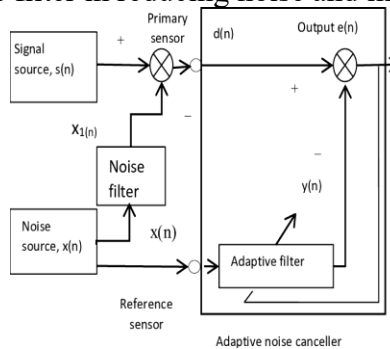


Figure8:AdaptiveNoiseCancellationAlgorithms

Figure 10 shows the input and output waveform of adaptive noise cancellation model. The capability of suggested algorithms is apparent in the error estimation and removes from the original signal after mixed with the noise signal. The first waveform represents pure input sine wave and the second waveform shows the attenuated signal by error effect. Hence, the error signal has isolated from the original signal properly. Figure 11 illustrates the frequency response of FIR-RLS filter and low passes filtering. Figure 12 shows the FIR filter taps

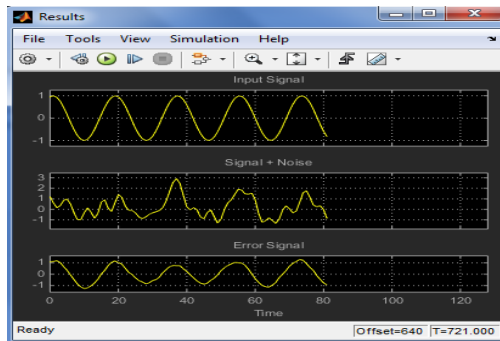


Figure 10: Input and output waveforms of the RLS model

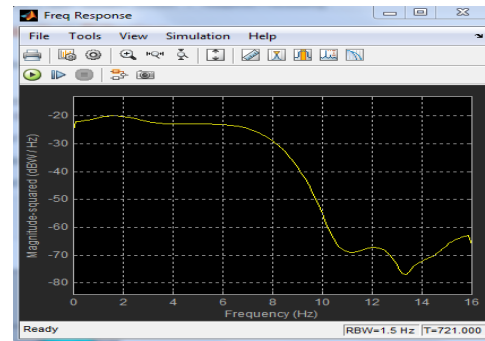


Figure 11: Frequency response of FIR-RLS filter.

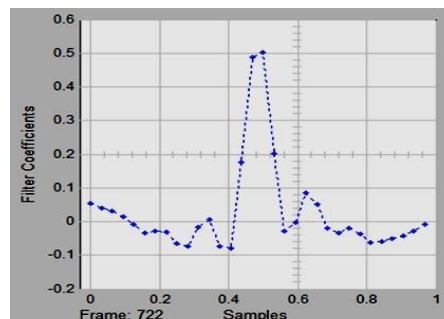


Figure12: FIR filter

5. System Model

The system model used in this work is based on an adaptive noise cancellation technique designed to remove unwanted noise from a corrupted signal. In communication systems, the

transmitted signal often gets mixed with noise during transmission through the channel. This noise reduces the quality of the received signal and affects the accuracy of the communication process. Therefore, an effective noise cancellation system is required to estimate and suppress the noise component while preserving the original signal.

In the proposed system, the input signal consists of a desired signal mixed with Additive White Gaussian Noise (AWGN). A reference noise signal that is correlated with the noise present in the primary input is provided to the adaptive filter. The adaptive filter processes the reference noise and generates an estimate of the noise component. This estimated noise is then subtracted from the primary input signal to produce an error signal, which represents the filtered output with reduced noise.

The adaptive filter continuously updates its coefficients based on the error signal in order to minimize the difference between the desired signal and the filter output. This iterative process allows the system to adapt to changing noise conditions and improve signal quality. The system model is implemented using MATLAB, which enables simulation and analysis of the noise cancellation performance using adaptive filtering algorithms.

A. Adaptive Filtering

An adaptive filter adjusts its coefficients automatically to minimize error.

Error Signal:

$$e(n) = d(n) - y(n)$$

Where:

$$d(n) = \text{Desired signal} \quad y(n) = \text{Filter output} \quad e(n) = \text{Error signal}$$

B. LMS Algorithm

The Least Mean Square (LMS) algorithm is one of the most popular adaptive filtering algorithms used in digital signal processing. It is mainly used in noise cancellation systems because of its simplicity and low computational complexity. The LMS algorithm updates the filter coefficients iteratively in order to minimize the mean square error between the desired signal and the output signal.

In communication systems, signals transmitted through channels are often corrupted by noise. The LMS algorithm helps in adapting the FIR filter coefficients automatically to remove unwanted noise and improve the quality of the signal.

Weight Update Equation:

$$w(n+1) = w(n) + \mu e(n)x(n)$$

Where:

$$w(n+1) = \text{Step size (learning rate)} \quad e(n) = \text{Error signal}$$

$$x(n) = \text{Input vector}$$

C. RLS Algorithm

The Recursive Least Squares (RLS) algorithm is an advanced adaptive filtering technique widely used in digital signal processing for noise cancellation. The main objective of the RLS algorithm is to minimize the error between the desired signal and the output signal of the adaptive filter.

Unlike the LMS algorithm, which updates filter coefficients based on the current error, the RLS algorithm uses information from previous input samples to calculate the filter coefficients. Because of this property, the RLS algorithm provides faster convergence and better noise reduction performance compared to LMS.

The RLS algorithm is particularly useful in communication systems where signals are affected by rapidly changing noise conditions. By quickly adapting the filter coefficients, the RLS algorithm improves the quality of the received signal.

The RLS algorithm works by minimizing the weighted least squares cost function. It continuously updates the filter coefficients using both current and past signal samples.

The error signal is calculated as:

$$e(n) = d(n) - y(n)$$

where

$e(n)$ = error signal $d(n)$ = desired signal

$y(n)$ = output of the adaptive filter

6. Results and Discussion

The performance of the proposed LMS and RLS adaptive algorithms is evaluated using MATLAB and SIMULINK with the objective of noise cancellation. In this study, a sine wave signal mixed with Additive White Gaussian Noise (AWGN) is used as the input signal for the adaptive filtering algorithms. The adaptive filter processes the noisy signal and significantly reduces the noise present in the communication channel.

The difference between the desired signal and the adaptive filter output represents the error signal, which indicates the amount of noise remaining in the processed signal. During the filtering process, the adaptive algorithms update their coefficients iteratively in order to minimize this error. The simulation results show that the adaptive filter gradually reduces the error value and produces an output signal that closely matches the original desired signal.

The error performance of the Normalized LMS and RLS algorithms for different noise amplitudes is presented in Table I. The results indicate that the RLS algorithm generally provides better noise reduction performance and faster convergence compared to LMS. However, the LMS algorithm requires lower computational complexity and performs efficiently during transmission in certain scenarios. Therefore, both algorithms have their own advantages, and the selection of an appropriate algorithm depends on the specific requirements of the communication system.

Table 1: Error Calculation in LMS and RLS algorithms

An amplitude of Noise Signal	LMS algorithm Error	RLS algorithm Error
200mv-50%ofthe originalsignal	4.607×10^{-4}	2.1264×10^{-5}
100mv-40%ofthe original signal	4.664×10^{-4}	2.2464×10^{-5}
50mv- 30%ofthe originalsignal	4.702×10^{-4}	2.4764×10^{-5}
20mv-10%oforiginalsignal	4.851×10^{-4}	2.6964×10^{-5}

7. Conclusion

This report presented an adaptive noise cancellation approach for improving signal quality in communication systems using LMS and RLS algorithms. In communication systems, transmitted signals are often corrupted by noise during transmission through the communication channel. Therefore, effective noise reductions techniques are required to recover the original signal and improve communication reliability. The study focused on adaptive filtering methods, particularly the Least Mean Square (LMS) and Recursive Least Squares (RLS) algorithms, and analyzed their performance in noise cancellation applications.

The LMS algorithm offers a simple structure and low computational complexity, making it suitable for real-time and low-power communication systems. However, its convergence speed is relatively slower and depends on the step-size parameter. On the other hand, the RLS

algorithm provides faster convergence and better noise suppression performance due to its recursive computation method, although it requires higher computational resources. Simulation-based analysis using MATLAB and SIMULINK demonstrated that both algorithms effectively reduce noise from signals corrupted by Additive White Gaussian Noise (AWGN) and improve overall signal clarity.

The results indicate that while RLS provides better convergence speed and lower error, the LMS algorithm remains efficient for systems with limited computational resources. Therefore, the selection of the appropriate adaptive algorithm depends on the specific requirements of the communication system. The proposed adaptive noise cancellation approach provides an effective solution for improving signal quality and can be applied in various communication applications such as wireless communication, speech processing, biomedical signal analysis, and audio enhancement systems.

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