

Adaptive Noise Canceller Using RLS Filter a Review

Somaya Hassan Al Shebane

Mariam Aboajela Msaad

Abdulatif A. Shaban

Libyan Academy, School of
 Applied Sciences and
 Engineering
 Tripoli, Libya

Faculty of Information Technology
 "network Department
 Tripoli, Libya
 meemee_02@yahoo.com;

Janzour Collage of Education
 University of Tripoli
 Tripoli, Libya
 Latif1995@yahoo.co.uk

Abstract- This paper describes the concept of adaptive noise cancelling . subtract noise from an input signal using the Recursive Least Squares (RLS) algorithm. The RLS adaptive filter uses the reference signal on the Input port and the desired signal on the Desired port to automatically match the filter response in the Noise Filter block. As it converges to the correct filter, the filtered noise should be completely subtracted from the "Signal+ Noise" signal, and the "Error Signal" should contain only the original signal.

Keyword – Noise Signal, Adaptive Filter, Noise Canceller, LMS Algorithm, RLS Algorithm.

I. INTRODUCTION

The adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics.

Adaptive filters are used, when it is necessary for the filter characteristics to be variable, adapted to changing conditions, when there is spectral overlap between the signal and noise or if the band occupied by the noise is unknown or varies with time. There are four main types of adaptive filter configurations Adaptive system identification, Adaptive noise cancellation.

Because of their self adjusting performance and in- built flexibility, adaptive filters are used in many diverse applications such as echo cancellation, radar signal processing, navigation systems, and equalization of communication channels and in biomedical signal enhancement. Adaptive filters are used, when it is necessary for the filter characteristics to be variable, adapted to changing conditions, when there is spectral overlap between the signal and noise or if the band occupied by the noise is unknown or varies with time.[1]

Noise Cancellation

Adaptive noise cancellation is often used to extract the desired speech from the given noisy speech. [3] The noise cancellation plays an important role in digital voice communication systems for e.g. Cell phones require adaptive noise cancellation to reduce further degradation of vo-coded speech. Active noise control (ANC) (also known as noise cancellation, active noise reduction (ANR) or anti-noise) is a method for reducing unwanted sound.

Adaptive FIR Filter

Adaptive filters are an important part of signal processing. Adaptive filters adjust its transfer function according to an optimizing algorithm.

Due to the complexity of the optimizing algorithms, most adaptive filters are digital filters.

Adaptive filters perform digital signal processing and adapt their performance based on the input signal . Adaptive filter can be classified as linear and non-linear adaptive filter.[2]

Recursive least squares (RLS) adaptive filter

The RLS Filter block recursively computes the least squares estimate (RLS) of the FIR filter weights. The block estimates the filter weights, or coefficients, needed to convert the input signal into the desired signal. Connect the signal you want to filter to the Input port. The input signal can be a scalar or a column vector. Connect the signal you want to model to the Desired port.[4]

The Recursive Least Squares (RLS) algorithm is used to subtract noise from an input signal. [4] The RLS adaptive filter uses the reference signal on the Input port and the desired signal on the desired port to automatically match the filter response in the Noise Filter block. As it converges to the correct filter, the filtered noise should be completely subtracted from the "Signal+ Noise" signal, and the "Error Signal" should contain only the original signal.

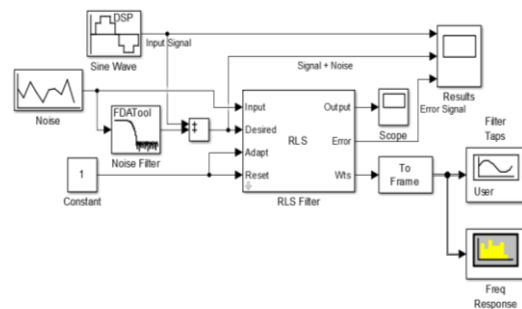


Figure1 . Simulink Model of RLS Adaptive Filter

1-Sine Wave Input

Output samples of a sinusoid. To generate more than one sinusoid simultaneously, enter a vector of values for the Amplitude, Frequency, and phase offset parameters.

2-Random Noise input

Output a random signal with uniform Gaussian (Normal) distribution. Set output repeatability to Non repeatable (block randomly selects initial seed every time simulation starts), Repeatable (block randomly selects initial seed once and uses it every time simulation starts), or Specify seed (block uses specified initial seed every time simulation starts, producing repeatable output).

3- Noise filter

The noise filter is used after the random noise input to scale the input noise signal.

4-RLS Filter

Computes filter weights based on the exponentially weighted recursive least squares (RLS) algorithm for adaptive filtering of the input signal. Select the adapt port check box to create an Adapt port on the block[5]. When the input to this port is nonzero, the block continuously updates the filter weights. When the input to this port is zero, the filter weights remain constant. If the Reset port is enabled and a reset event occurs, the block resets the filter weights to their initial values.

5-Filter Taps

Display a vector or matrix of time- domain, frequency domain, or user – specified data. Each column of a 2-D input matrix is plotted as a separate data channel. 1-D inputs are assumed to be a single data channel. For frequency – domain operation, input should come from a square such as the Magnitude FFT block, or a block with equivalent data Organization.

6-Frequency Spectrum Scope

Compute and display the periodogram of each input signal. Non-frame based inputs to the block should use the buffering option.

Simulation Results

The following figure shows the filter taps in the graph between filter coefficients and samples

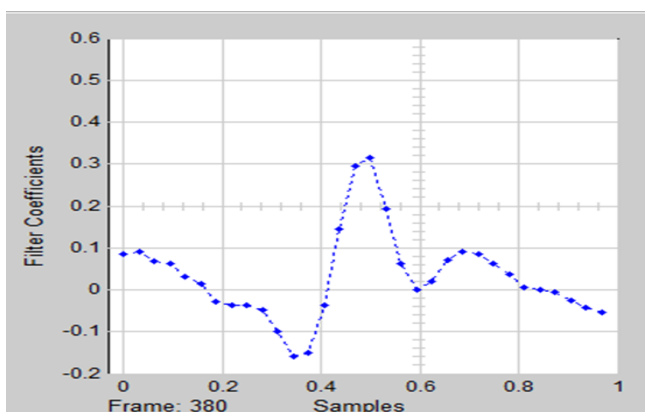


Figure 1.2 Filter taps of RLS filter

The SNR between original distorted signals in below diagram. is The following Figure 1.3 shows the result of the simulated RLS filter. It shows the input signal, which is a

sine wave. Then it shows the input signal with the noise signal is the type (uniform). Lastly it shows the error signal.

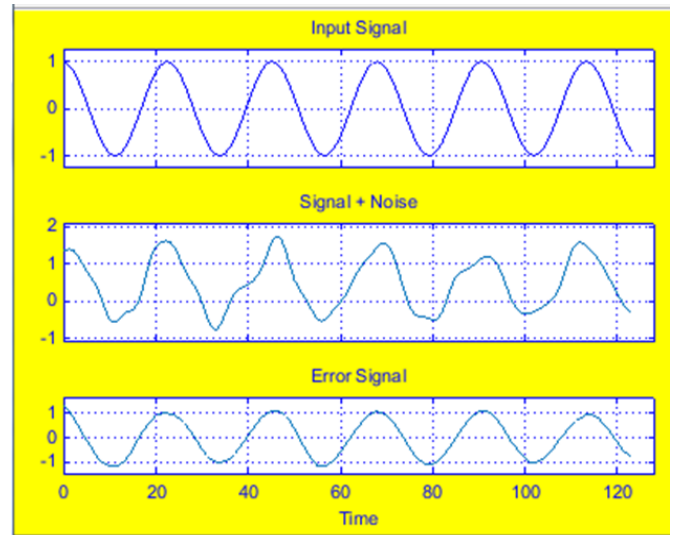


Figure 1.3 Simulation Results

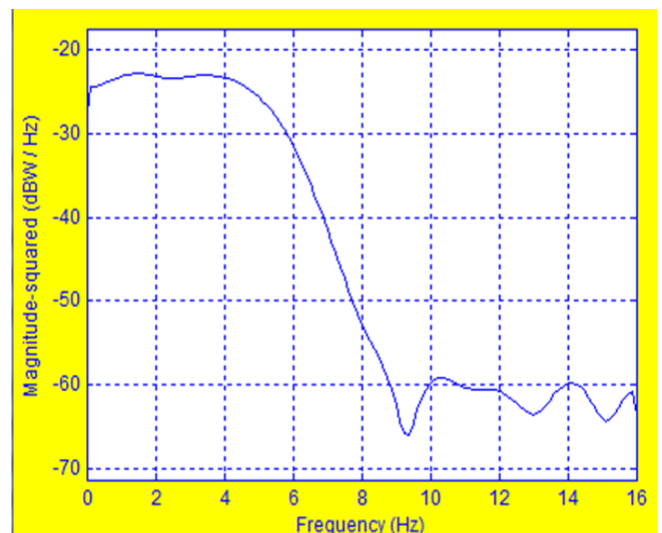


Figure 1.4 Frequency Response of RLS filter

Conclusion

In this paper, we have seen that the recursive least squares (RLS) algorithms have a faster convergence speed, the input signals are considered deterministic, while for the LMS and other similar algorithm they are considered stochastic. However, RLS algorithms involve more complicated mathematical operations and require more computational resources than the LMS algorithm.

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