Volume 2, Issue 11, November 2012 ISSN: 2277 128X
International Journal of Advanced Research in Computer Science and Software Engineering
Research Paper
Available online at: www.ijarcsse.com

Design of Adaptive Noise Canceller Using RLS Filter a Review

Reena Rani
Senior Lecture ECE Deptt.
REMTI, Shamli, Uttar Pradesh

Dushyant Kumar(Asst. Prof.)
H.C.T.M. Technical Campus
Kaithal Haryana

Narinder Singh
H.C.T.M. Technical Campus
Kaithal Haryana

Abstract: This paper describes the concept of adaptive noise cancelling. It is an alternative method of estimating signals corrupted by additive noise. In this paper we describe the noise cancellation using the Recursive Least Squares (RLS) to remove the noise from an input signal. 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. The filtered noise should be completely subtracted from the "noisy signal" of the input Sine wave & noise input signal, and the "Error Signal" should contain only the original signal.

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

1. Introduction

The tremendous growth of development in the digital signal processing area has turned some of its specialized areas into fields themselves. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications. 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. 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 [3].

2. Basic Concept of the Noise Canceller

Noise cancellation makes use of the notion of destructive interference. When two sinusoidal waves superimpose, the resulting waveform depends on the frequency, amplitude and relative phase of the two waves. If the original wave and the inverse of the original wave encounter at a junction at the same time, total cancellation occur. The challenges are to identify the original signal and generate the inverse without delay in all directions where noises interact and superimpose.

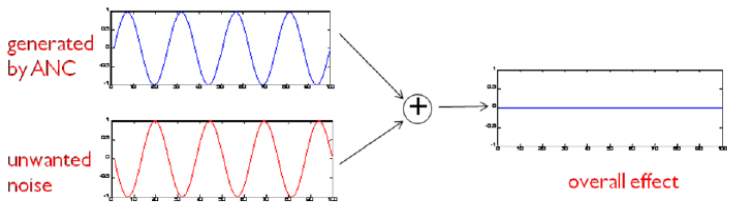

Figure 2.1 Basic concept of noise canceller

The common adaptive algorithms that have found widespread application are the Least Mean Squares (LMS) and the Recursive Least Squares (RLS).

3. Recursive Least Squares (RLS) Adaptive Filters

The Recursive Least Squares (RLS) algorithm is based on the well-known least squares method [2]. The least-squares method is a mathematical procedure for finding the best fitting curve to a given set of data points. This is done by minimizing the sum of the squares of the offsets of the points from the curve. The RLS algorithm recursively solves the least squares problem. In the following equations, the constants λ and δ are parameters set by the user that represent the forgetting factor and regularization parameter respectively. The forgetting factor is a positive constant less than unity, which is roughly a measure of the memory of the algorithm; and the regularization parameter's value is determined by the signal-to-noise ratio (SNR) of the signals. The vector $\hat{w}$ represents the adaptive filter's weight vector and the M-by-M matrix P is referred to as the inverse correlation matrix. The vector π is used as an intermediary step to computing the gain vector k. This gain vector is multiplied by the a priori estimation error ξ (n) and added to the weight

vector to update the weights. Once the weights have been updated the inverse correlation matrix is recalculated, and the training resumes with the new input values.[6] A summary of the RLS algorithm follows:
Initialize the weight vector and the inverse correlation matrix P.
$W^H(0) = \bar{0}$……………………………….…….3.1
P (0) $=\delta^{-1}$………………………..……………...3.2
Where δ =
Small positive constant for high SNR
Large positive constant for low SNR
For each instance of time n =1, 2, 3 …compute:
$\pi(n) = P(n-1)u(n)$....................................3.3
$k\ (n) = \frac{\pi(n)}{\lambda + u^H(n)\pi(n)}$…………………………..3.4
$\xi(n) = d(n) - w^H(n-1)u(n)$…………….3.5
$w(n) = w(n-1) + k(n)\bar{\xi}(n)$…………..….3.6
And

$$P(n) = \lambda^{-1}p(n-1) - \lambda^{-1}k(n)u^H(n)p(n-1) \text{……3.7}$$

An adaptive filter trained with the RLS algorithm can converge up to an order of magnitude faster than the LMS filter at the expense of increased computational complexity.

3.1 Recursive Least Squares (RLS) algorithm
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.

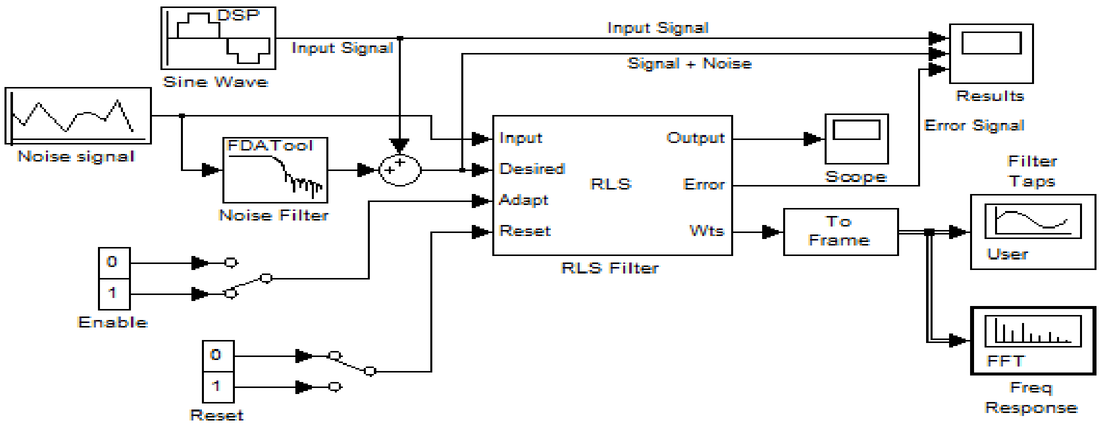

Figure 3.1 Noise canceller using Recursive Least Squares filter [8]

3.1.1 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.
3.1.2 Random Noise input
Output a random signal with uniform Gaussian (Normal) distribution. Set output repeatability to Nonrepeatable (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).

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

3.1.4 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. When the input to this port in 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.

3.1.5 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 organisation.

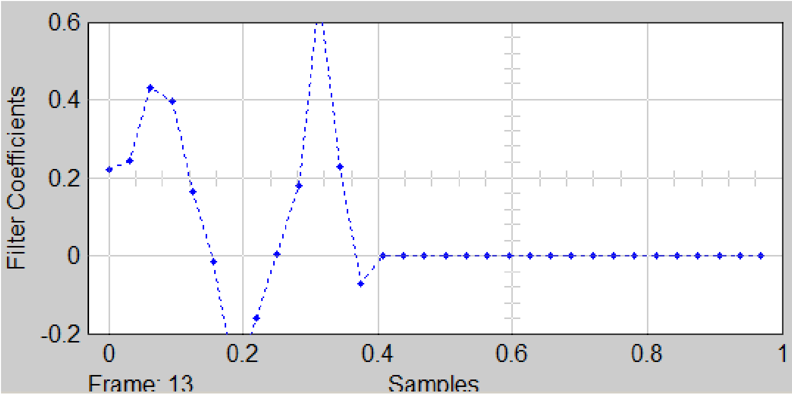

Figure 3.2 Filter taps

3.1.6 Frequency Spectrum Scope

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

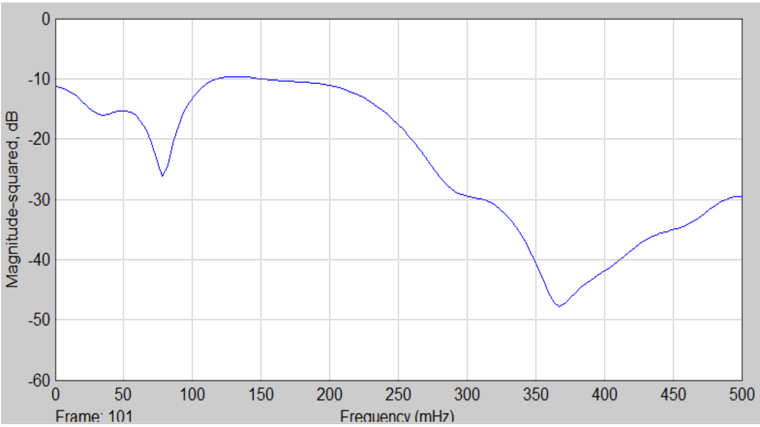

Figure 3.3 Frequency Response of noise canceller

4. Results

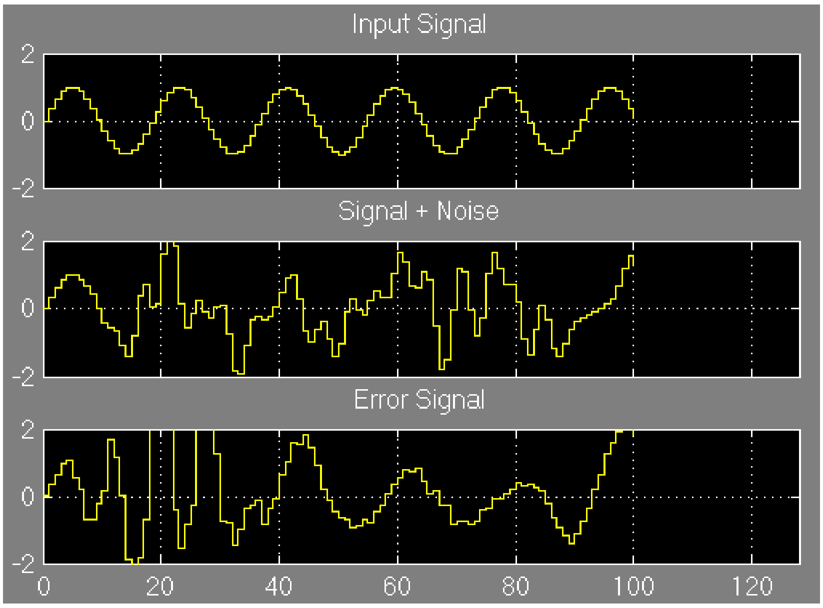

Figure 4.1 Results of RLS Filter

5. Conclusion

In this paper, we have seen that the recursive least squares (RLS) algorithms have a faster convergence speed and in the derivation of the RLS algorithm, 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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