Orissa Journal of Physics
ISSN 0974-8202
© Orissa Physical Society
Vol. 24, No.1
February 2017
pp. 161-169

Simulation and Comparative Analysis of Adaptive Noise Cancellations using LMS, NLMS AND RLS Algorithms

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Received: 2.12.2016 ; Revised : 18.12.2016 ; Accepted :21.1.2017

Abstract. This paper focused on the adaptive noise cancellation of speech signal using Least Mean Square (LMS), Normalized Least Mean Square (NLMS) and Recursive Least Square methods. Adaptive noise cancellation is an alternative way of cancelling noise present in a corrupted signal. In this technique, evaluation of distorted signal by additive noise or interference achieved with no a priori estimates of signal or noise. A comparative study is carried out using LMS, NLMS and RLS methods. Result shows that these methods have potential in noise cancellation and can be used for system identification problem and variety of applications. Computer simulations for all cases are carried out using MAT LAB software.

Keywords: LMS, NLMS, RLS, Noise Cancellation, Adaptive Filtering, System Identification, Convergence rate, MATLAB CODES

1. Introduction

In the process of digital signal processing, 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[2]. Under such circumstances, we must design adaptive filters, to track the changes of signal and noise. Adaptive Filter is that it uses the filter parameters of a moment ago to automatically adjust the filter parameters of the present moment, to adapt to the statistical properties that signal and noise unknown or random change [1], in order to achieve optimal filter. Based on in-depth study of adaptive filter, based on the least mean squares ,normalized least mean square algorithm and recursive least squares are applied to the adaptive filter technology to the noise, and through the simulation results prove that its performance is usually much better than using conventional methods designed to filter fixed.

2. Adaptive Filter

The so-called adaptive filter, is the use of the result of the filter parameters a moment ago, automatically adjust the filter parameters of the present moment, to adapt to the unknown signal and noise, or over time changing statistical properties, in order to achieve optimal filtering [3]. Adaptive filter has "self-regulation" and "tracking" capacities. Adaptive filter can be divided into linear and nonlinear adaptive filter. Non-linear adaptive filter has more signal processing capabilities. However, due to the non-linear adaptive filter more complicated calculations, the actual use is still the linear adaptive filter [2]. As shown in Figure.

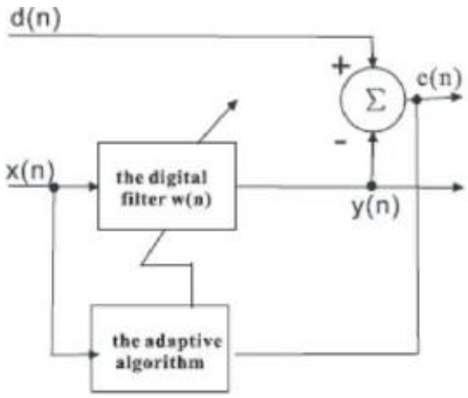

Fig.1. Adaptive Filter Scheme

The figure above is given the general adaptive filtering display: digital filter carries on filtering on the input signal x(n) , produce output signal y(n). Adaptive algorithm adjusts the filter coefficient included in the vector w (n), in order to let the error signal e(n) to be the smallest. Error signal is the difference of useful signal d(n) and the filter output y(n).Therefore, adaptive filter automatically carry on a design based on the characteristic of the input signal x(n) and the useful signal d(n)[4]. Using this method, adaptive filter can be adapted to the environment set by these signals. When the environment changes, filter through a new set of factors, adjusts for new features[3] .The most important properties of adaptive filter is that it can work effective in unknown environment, and to track the input signal of time-varying characteristics [5]. Adaptive filter has been widely used in communications, control and many other systems. Filter out an increase noise usually means that the contaminated signal through the filter aimed to curb noise and signal relatively unchanged. This filter belongs to the scope of optimal filtering [6], the pioneering work started from Wiener, and Kalman who work to promote and strengthen. For the purpose of the filter can be fixed, and can also be adaptive. Fixed filter designers assume that the signal characteristics of the statistical computing environment fully known, it must be based on the prior knowledge of the signal and noise. However, in most cases it is very difficult to meet the conditions; most of the practical issues must be resolved using adaptive filter. Adaptive filter is through the observation of the existing signal to understand statistical properties, which in the normal operation to adjust parameters automatically [7], to change their performance, so its design does not require of the prior knowledge of signal and noise characteristics. Adaptive filter is used for the cancellation of the noise component which is overlap with unrelated signal in the same frequency range [2].

3. Adaptive Noise Cancellation

The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unchanged. And as we mentioned above, this is an adaptive process, which means it cannot require a priori knowledge of signal or noise characteristics [8].

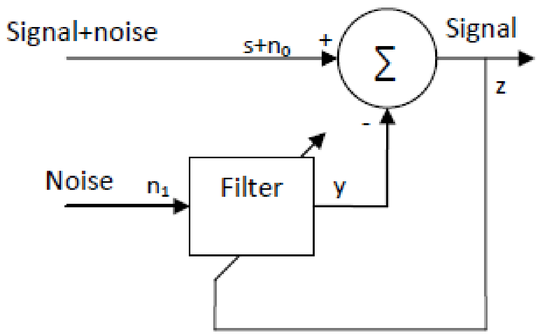

Fig.2. Adaptive Noise Cancellations

To realize the adaptive noise cancellation please refers the figure 2 where, we use two inputs and an adaptive filter. One input is the signal corrupted by noise (Primary Input, which can be expressed as S_{n0}) [9]. The other input contains noise related in some way to that in the main input but does not contain anything related to the signal (Noise Reference Input**,** expressed as n_1). The noise reference input pass through the adaptive filter and an output y is produced as close a replica as possible of n_0. The filter readjusts itself continuously to minimize the error between n_0 and y during this process. Then the output y is subtracted from the primary input to produce the system output $z = s + n_0 - y$, which is the de-noised signal [10]. Assume that s, n_0, n_1 and y are statistically stationary and have zero means. Suppose that s is uncorrelated with n_0 and n_1, n_1 is correlated with n_0. We can get the following equation of expectations in (1)

$$E[z^2] = E[s^2] + E[(n_0 - y)^2] \quad (1)$$

When the filter is adjusted so that $E[z^2]$ is minimized, $E[(n_0 - y)^2]$ also minimized. So the system output z can serve as the error signal for the adaptive filter.

4. Adaptive Algorithms:

Least-Mean-Square Algorithm

The LMS is the most used algorithm in adaptive filtering. It is a gradient descent algorithm; it adjusts the adaptive filter taps modifying them by an amount proportional to the instantaneous estimate of the gradient of the error surface. It is represented in following equations.

$$J(n) = E|e^2(n)|$$

$$y(n) = \hat{\mathbf{w}}^H(n).\mathbf{u}(n)$$
$$e(n) = d(n) - y(n)$$
$$\hat{\mathbf{w}}(n+1) = \hat{\mathbf{w}}(n) + \mu \cdot \mathbf{u}(n) \cdot e(n)$$
$$d(n) = \boldsymbol{w}_o^T \boldsymbol{x}(n) + n(n) \quad (2)$$

Where ***wo*** is a vector which represents the system to be identified and *n (n)* is the noise

Normalized Least-Mean-Square Algorithm:

A general form of the adaptive filter is illustrated in Figure 1, where ***d(n)*** is a desired response (or primary input signal), ***y(n)*** is the actual output of a programmable digital filter driven by a reference input signal ***x(n),*** and the error ***e(n)*** is the difference between ***d(n)*** and *y(n).* The function of the adaptive algorithm is to adjust the digital filter coefficients to minimize the mean-square value of ***e (n).***

A technique to adjust the convergence speed is the Normalized LMS (NLMS) algorithm. The NLMS is shown as follows:

$$w(n+1) = w(n) + \mu(n)x(n)e(n) \tag{3}$$

u (n) is adaptive step size which is computed as

$$\mu(n) = \frac{\alpha}{L\hat{P}_i(n)}, \qquad 0 < \alpha < 2 \tag{4}$$

P(n) is the estimated power of ***x(n)*** at time n , *L* is order of the filter, and *a* is the normalized step size. An exponential window is used to estimate the power of x(n)

$$\hat{P}_i(n) = (1-\beta)\hat{P}_i(n-1) + \beta x^2(n) \tag{5}$$

where β is a smoothing parameter, which is in terms of its equivalent (exponential) window length

$$M \equiv \frac{1}{\beta} \tag{6}$$

RLS Algorithm:

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[2].

$$J(n) = \sum_{i=0}^{k} \lambda^{n-i} e^2(n)$$

$$
\begin{aligned}
y(n) &= \hat{\mathbf{w}}^H(n).\mathbf{u}(n) \\
\alpha(n) &= d(n) - \hat{\mathbf{w}}^H(n-1)\mathbf{u}(n) \\
\pi(n) &= \mathbf{u}^H(n).\mathbf{P}(n-1)
\end{aligned}
$$

$$
\begin{aligned}
k(n) &= \lambda + \pi(n).\mathbf{u}(n) \\
\mathbf{k}(n) &= \frac{\mathbf{P}(n-1).\mathbf{u}(n)}{k(n)} \\
\hat{\mathbf{w}}(n) &= \hat{\mathbf{w}}(n-1) + \mathbf{k}(n).\alpha^*(n) \\
\mathbf{P}'(n-1) &= \mathbf{k}(n).\pi(n) \\
\mathbf{P}(n) &= \frac{1}{\lambda}\left(\mathbf{P}(n-1) - \mathbf{P}'(n-1)\right)
\end{aligned} \qquad (7)
$$

5. Simulation Results

1. The LMS Solution:

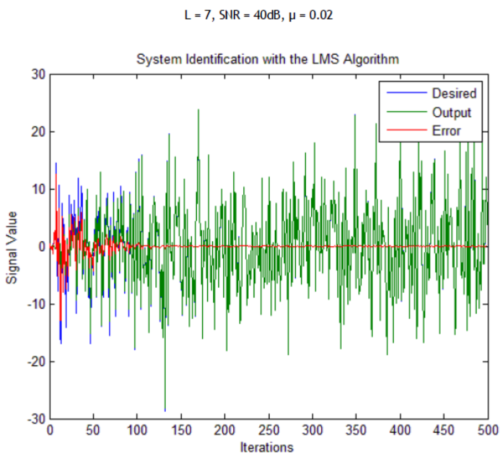

Fig. 3. Desired signal, filter output and error of the LMS algorithm for the given system identification problem

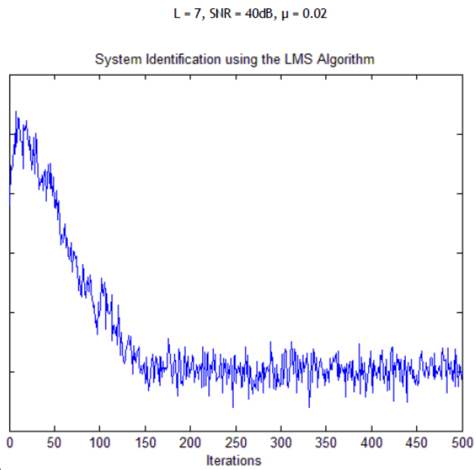

Fig.4. Mean-squared error of the LMS algorithm

2. The NLMS Solution:

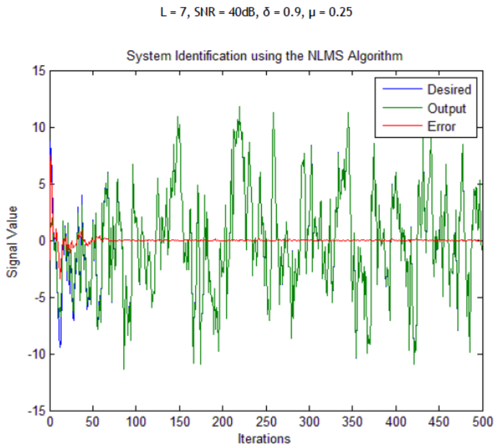

Fig.5. Desired signal, filter output and the error of the NLMS algorithm for the system identification given problem

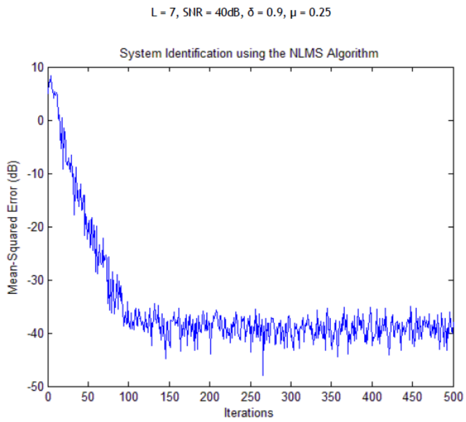

Fig.6. Mean-squared-error of the NLMS algorithm

3. The RLS Solution:

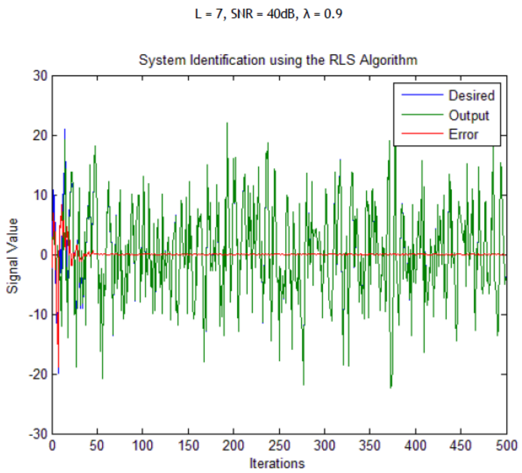

Fig.7: Desired signal, filter output and the error of the RLS algorithm for the given system identification problem

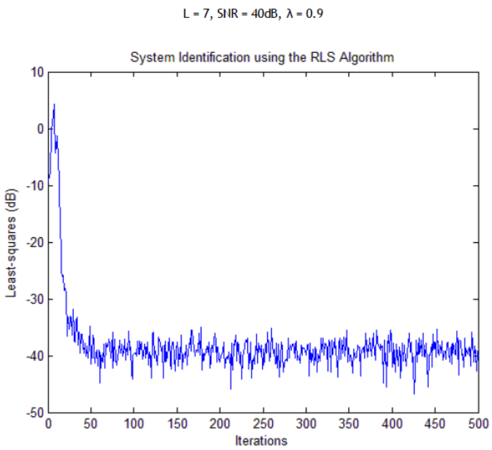

Fig.8: Weighted least-squares of the RLS algorithm

6. Comparisons of Results

In real-time applications, it is very important to analyse all the important details before we choose an adaptive algorithm. A small difference could result in elevated cost of implementation, or in a weak system, which is not stable in all variable changes, or even the solution is impossible to be implemented. The choice between using one algorithm instead of another, to the system identification problem, depends mainly on the following factors:

- Rate of Convergence: number of iterations required by the algorithm, to converge to a value close to the optimum Wiener solution in the mean-square sense. If the algorithm has a fast rate of convergence, it means that the algorithm adapts rapidly to a stationary unknown environment.
- Computational cost: when we talk about computational cost, it includes implementation cost, amount necessary to implement the algorithm in a computer and the number of arithmetic operations. The order of the operations is also important as well as the memory allocation, which is the space necessary to store the data and the program.
- Tracking: capacity of the algorithm to track statistical variations in a stationary unknown environment.

In this particular application, those three factors have has been analysed. The tracking factor has been analysed in two stages, one after a few hundred of iterations and the other after a few thousand iterations.

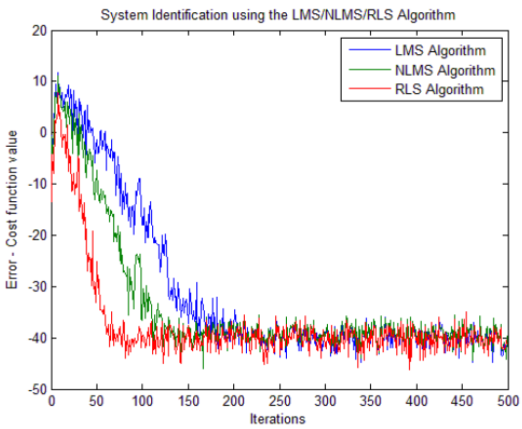

Fig.9. Comparison between the rates of convergence of the three proposed algorithms

The figure 9 above shows a comparison between the rates of convergence of the three proposed algorithms. Investigating the results shown in figure 9, it can be detected that the rate of convergence of the RLS algorithm is faster than the other two. Indeed, it can be twice faster than the NLMS and three times faster than the LMS algorithm.

L = 7, SNR = 40dB, $\mu = 0.02$(LMS), $\delta = 0.9$, $\mu = 0.25$(NLMS), $\lambda = 0.9$

7. Conclusion

Adaptive filtering is an important basis for signal processing, in recent years has developed rapidly in various fields on a wide range of applications. In addition to noise cancellation for system identification, adaptive filter the application of space is also very extensive. For example, we know areas that the adaptive equalizer, linear prediction, adaptive antenna array, and many other areas. Adaptive signal processing technology as a new subject is in depth to the direction of rapid development, especially blind adaptive signal processing and use of neural networks of non-linear adaptive signal processing, for the achievement of intelligent information processing system, a good prospect.

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