Simulation and Comparative Analysis of LMS and RLS Algorithms Using Real Time Speech Input Signal

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Abstract **- In practical application, the statistical characteristics of signal and noise are usually unknown or can't have been learned so that we hardly design fix coefficient digital filter. In allusion to this problem, the theory of the adaptive filter and adaptive noise cancellation are researched deeply. According to the Least Mean Squares (LMS) and the Recursive Least Squares (RLS) algorithms realize the design and simulation of adaptive algorithms in noise canceling, and compare and analyze the result then prove the advantage and disadvantage of two algorithms .The adaptive filter with MATLAB are simulated and the results prove its performance is better than the use of a fixed filter designed by conventional methods.**

KEY WORDS: Adaptive filters, Adaptive algorithm, RLS, LMS.

I. 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 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.

II. ADAPTIVE FILTERS

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.

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Adaptive filter can be divided into linear and nonlinear ad 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.

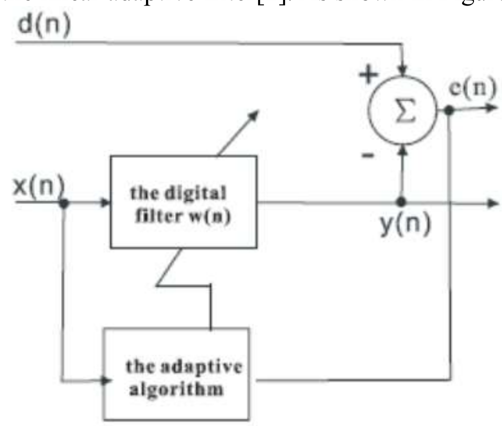

Figure 1: Adaptive filter scheme

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].

(i) LMS 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. [2]

$$y(n) = \hat{\mathbf{w}}^H(n).\mathbf{u}(n)$$
$$e(n) = d(n) - y(n)$$
$$\hat{\mathbf{w}}(n+1) = \hat{\mathbf{w}}(n) + \mu \cdot \mathbf{u}(n) \cdot e(n)$$

(ii) 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 $\mathbf{P}(n)$ should be made equal to δ^{-1} where δ is a small positive constant[2].

$$y(n) = \hat{\mathbf{w}}^H(n).\mathbf{u}(n)$$
$$\alpha(n) = d(n) - \hat{\mathbf{w}}^H(n-1)\mathbf{u}(n)$$
$$\pi(n) = \mathbf{u}^H(n).\mathbf{P}(n-1)$$
$$k(n) = \lambda + \pi(n).\mathbf{u}(n)$$
$$\mathbf{k}(n) = \frac{\mathbf{P}(n-1).\mathbf{u}(n)}{k(n)}$$
$$\hat{\mathbf{w}}(n) = \hat{\mathbf{w}}(n-1) + \mathbf{k}(n).\alpha^*(n)$$
$$\mathbf{P}'(n-1) = \mathbf{k}(n).\pi(n)$$
$$\mathbf{P}(n) = \frac{1}{\lambda}\left(\mathbf{P}(n-1) - \mathbf{P}'(n-1)\right)$$

III. SIMULATION RESULTS

(a) Noise reduction with LMS

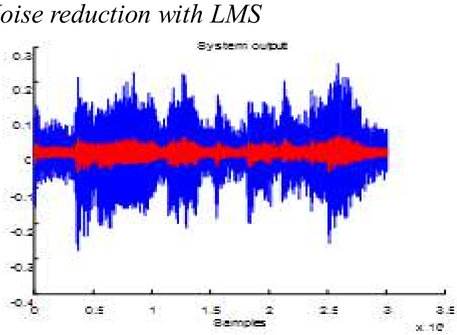

Figure 2 : True and Estimated output

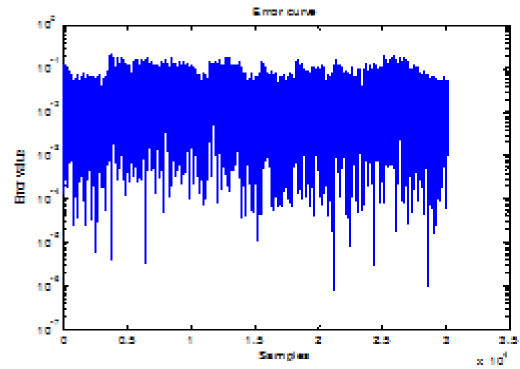

Figure 3: Error Curve

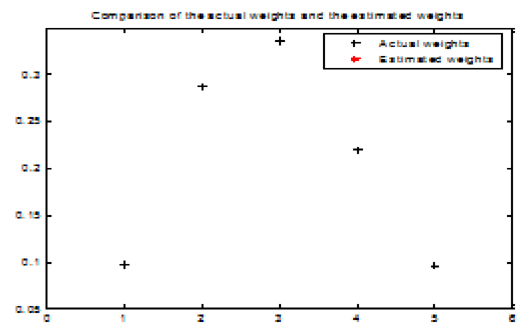

Figure 4: plot of estimated weights and filter weights

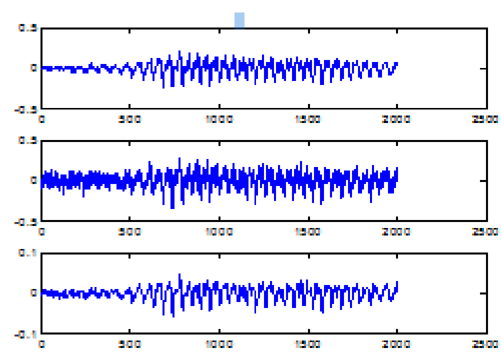

Figure 5 : Original wave , noisy wave and filtered wave

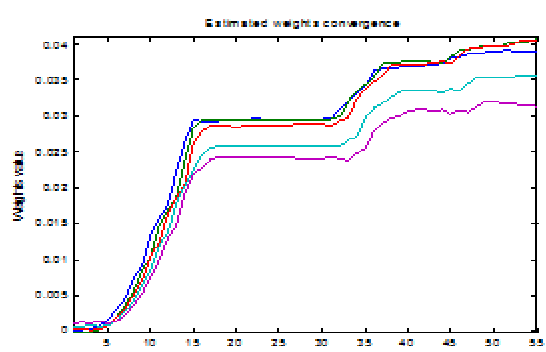

Figure 6 : Coefficient convergence estimate

(b) Noise reduction with RLS

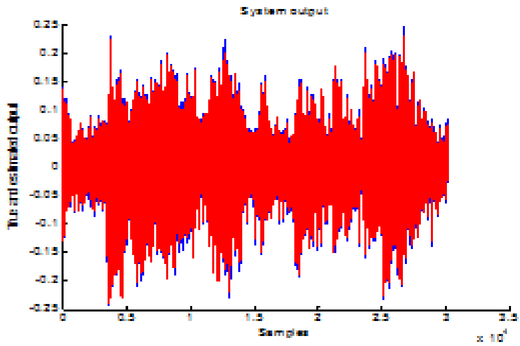

Figure 7: True and Estimated output

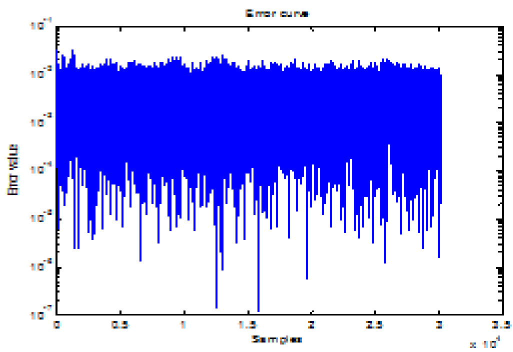

Figure 8 : Error curve

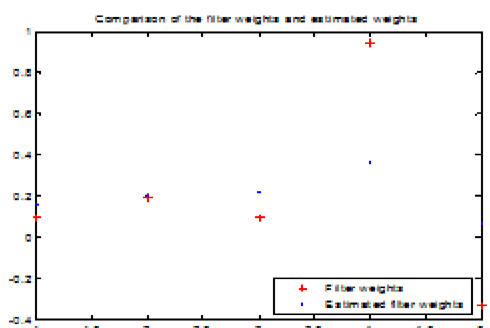

Figure 9: plot of estimated weights and filter weights

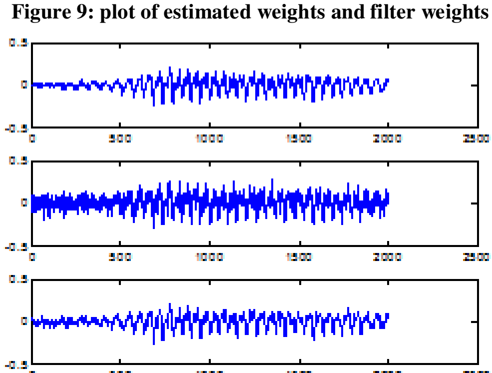

Figure 10: Original wave , noisy wave

and filtered wave

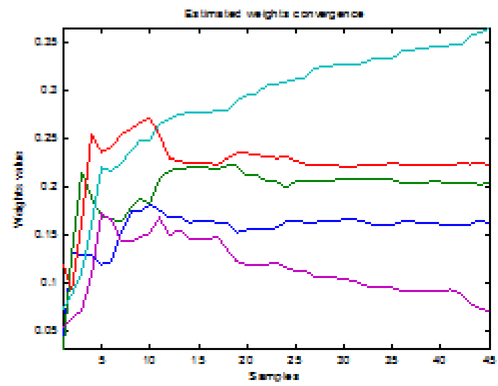

Figure 11: Coefficent convergence estimate

IV. Comparative Analysis of LMS and RLS Algorithms

The simulation results are achieved using real time speech input signal in MATLAB environment. The simulation results show that more than LMS algorithm and RLS algorithm in the area to cancel the noise has very good results, to complete the task of noise reduction. LMS filters filtering results is relatively good, the requirements length of filter is relatively short, it has a simple structure and small operation and is easy to realize hardware. But the shortcomings of LMS algorithm convergence rate is slower, but the convergence speed and noise vector there is a contradiction, accelerate the convergence speed is quicker at the same time noise vector has also increased. Convergence of the adaptive for the choices of gain constant μ is very sensitive. The noise signal and signal power when compared to larger, LMS filter output is not satisfactory, but we can step through the adjustment factor and the length of the filter method to improve. RLS algorithm filter the convergence rate is faster than the LMS algorithm, the convergence is unrelated with the spectrum of input signal, filter performance is superior to the least mean squares algorithm, but its each iteration is much larger operation than LMS. The required storage capacity is large, is not conducive to achieving a timely manner, the hardware is also relatively difficult to achieve the task of noise reduction.

V. 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, adaptive filter the application of space is also very extensive. For example, we know areas that the system identification, 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.

VI. FUTURE WORK

The application can be extended for the noise reductin in the speech for the hearing aids in the noisy environment like crowd noise, car noise, cockpit noise, aircraft noise etc. With modified RLS and LMS algorithm convergence speech can be increased as per the real time requirement fast algorithm can be developed.

VII. REFERENCES:

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