Active Noise Cancellation System

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Abstract - An active noise cancellation system has been designed and implemented. Both LS & RLS methods were used to establish the effectiveness. MATLAB was used to design and test a least mean square (LMS) and a recursive least square (RLS) adaptive filter for the project. Results obtained from the MATLAB model to determine efficient method for the filter.

Keywords - Weiner filter, Least Squares, Recursive Least Squares

I. INTRODUCTION

The goal of the project was to design and implement an active noise cancellation system using an adaptive finite impulse response (FIR) filter. This active noise cancellation system would be used to increase the signal-to-noise ratio (SNR) of a signal by decreasing the power of the noise. The study of active noise cancellation is a rapidly developing area. With the concern for noise pollution on the rise, methods of reducing noise are in greater demand. Active noise cancellation systems with adaptive filters are considered an effective method for reducing unwanted information (i.e., noise).

II. ADAPTIVE Filters

Adaptive filters consist of the three basic components: the adaptive filter $h(n)$ the error $e(n)$ and the adaptation function $y(n) = x(n) * h(n)$and $e(n) = d(n) - y(n)$ as shown in Figure 1. The goal of the system in Figure 1 is to adapt the filter in such a way that the input digital signal $x(n)$, is filtered to produce an output signal $y(n)$, that will minimize the error signal $e(n)$, when subtracted from the desired signal $d(n)$. The arrow through the adaptive filter is standard notation to indicate that the filter is adaptive. This means that all of the filter coefficients can be adjusted in such a way that the mean square error is to be minimized. The adaptive filter can be an FIR or IIR filter or even a non-linear system. To ensure the stability of the adaptive algorithm, most adaptive filters use an FIR type.

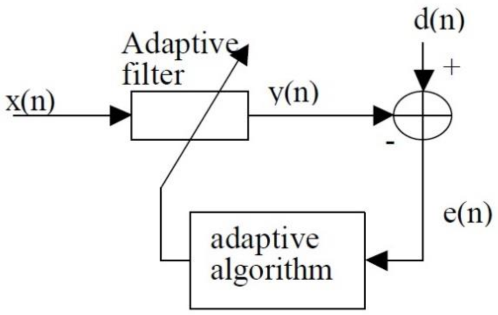

Figure 1 Adaptive Filter

Active noise cancellation increases the signal-to-noise ratio of a signal by decreasing the noise power in the signal by attempting to cancel noise signals. Applications consist of adaptive noise cancellation, echo cancellation, adaptive beamforming, biomedical signal processing, and others.

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.

$$y(n) = X(n).h^*(n)$$

Where $h = \left(X(n)^H X(n)\right)^{-1} X(n)^H d(n) = R_{xx}{}^{-1}.r_{dx}$

And the error we are trying to minimize is given by,

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

ii. RLS Algorithm

Recursive least square (RLS) is another algorithm for adaptive filters. This algorithm attempts to directly update the auto and cross-correlation matrices in order to approach the Wiener-Hopf equation. The RLS algorithm attempts to directly update its estimate of the optimum coefficients to approach the Wiener-Hopf equation.

$$R_{XX}(n+1) = R_{XX}(n) + X\,(n).X^T(n)$$

$$r_{dx}(n+1) = r_{dx}(n) + \, d(n).X\,(n)$$

Using these to update our values for each new input, we calculate the filter coefficients with the following:

$$f(n+1) = R_{XX}{}^{-1}(n+1)\,.\,r_{dx}(n+1)$$

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With a new input sample $f(n)$, and desired output value $d(n)$, we can summarize the RLS algorithm as

1. Update the input history vector $f(n)$.
2. Compute the filter output using the previous set of filter coefficients $b(n-1)$

$$y(n) = f^T(n)b(n-1)$$

3. Compute the error

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

4. Compute the Kalman gain vector

$$k(n) = \frac{R^{-1}(n-1)f(n)}{\lambda + f^T(n)R^{-1}(n-1)f(n)}$$

5. Update the matrix $R^{-1}(n)$ for the next iteration

$$R^{-1}(n) = \lambda^{-1}[R^{-1}(n-1) - k(n)f^T(n)R^{-1}(n-1)]$$

6. Update the filter coefficients for the next iteration.

$$b(n) = b(n-1) + k(n)e(n)$$

III. CASE STUDY

In laboratory, we have been provided audio files, one of which contained a song corrupted with the helicopter noise and other file contained the reference audio of the helicopter which has to be filtered using different methods. We implemented both LMS and RLS algorithms in MATLAB and observed the filtering of the helicopter noise out of the song.

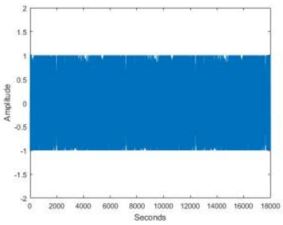

Figure 2 Noise Reference

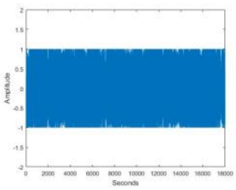

Figure 3 Signal with noise

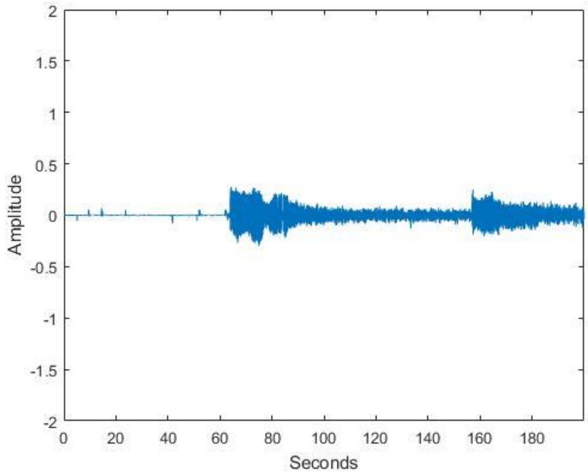

Figure 4 Desired Signal, derived from MATLAB Output

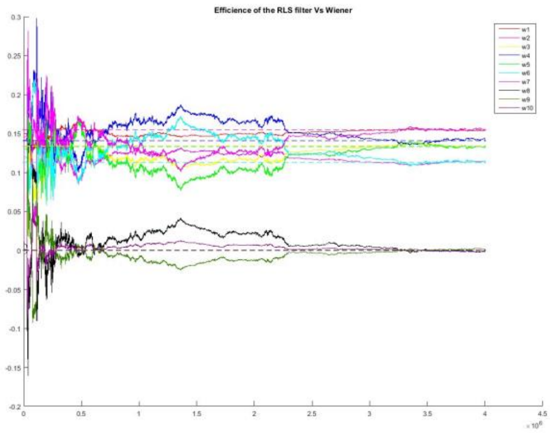

Figure 5 Efficiency of RMS vs LMS

Fig. (2) shows the Noise reference provided which is helicopter noise and Fig. (3) represents the signal with noise which needs to be filtered. After filtering we get the desired signal which is shown in Fig. (4). Efficiency of the RLS algorithm is represented in the Fig. (5) and detailed analysis is don't in the next section.

IV. COMPARATIVE ANALYSIS OF LMS AND RLS ALGORITHMS

The simulation results are achieved using provided 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. But the shortcomings of LMS algorithm convergence rate is slower, but the convergence speed and noise vector there is a contradiction,

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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 RLS algorithm was able to achieve more reduction in the mean square error over the LMS algorithm. In spite of this fact, the LMS is more widely used due to the complexity inherent in the RLS algorithm. The RLS algorithm requires a matrix inverse calculation at every time step.

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.

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