

Real time active noise cancellation using adaptive filters following RLS and LMS algorithm

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Abstract – In this paper we describe adaptive noise cancellation and demonstrate it using real time speech signals. The adaptive filter design discussed here is based on two algorithms RLS (Recursive Least Square) and LMS (Least Mean Square) and a comparison has been drawn based on their performance. Adaptive filters find application because of their dynamic nature and they work on the principle of destructive interference.

Key Words: Noise signal, Adaptive filter, RLS algorithm, LMS algorithm, Simulink

1. INTRODUCTION

Noise has proven to be the bottleneck in deciding the performance of communication system and its random nature makes it difficult in designing these systems. Filtering is an application of signal processing which is used to remove unwanted and spurious signals. The two types of filtering used are: fixed and adaptive. The difference being, adaptive filters do not require prior information and hence are more effective. 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. The two efficient algorithms for designing of adaptive filters are RLS and LMS algorithm.

2. LMS algorithm

One of the most widely used algorithm for noise cancellation using adaptive filter is the Least Mean Squares (LMS) algorithm. LMS adaptive filters are easy to compute and are flexible. The method uses a “primary” input containing the corrupted Signal and a “reference” input containing noise correlated in some unknown way with the primary noise. The reference input is adaptively filtered and subtracted from the primary input to obtain the signal estimate. A desired signal corrupted by additive noise can often be recovered by an adaptive noise canceller using the least mean squares (LMS) algorithm. It then enhances the SNR. LMS adjusts the adaptive filter coefficients and modify them by an amount proportional to the instantaneous estimate of

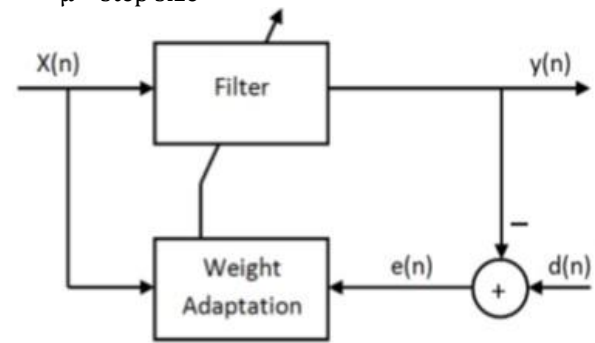
the gradient of the error surface. It neither requires correlation function calculation nor matrix inversions. LMS minimizes the power in the error. Minimization of mean square error is achieved by the repetitive procedure incorporated in it to make successive corrections in the direction of negative of the gradient vector which is represented in the following equations:

$$y(n) = F(n) \cdot x(n) \dots\dots\dots (i)$$

$$e(n) = d(n) - y(n) \dots\dots\dots (ii)$$

$$F(n + 1) = F(n) + \mu \cdot x(n) \cdot e(n) \dots\dots\dots (iii)$$

Where,
 $y(n)$ = filter output
 $x(n)$ = input signal
 $e(n)$ = error signal
 $d(n)$ = desired output
 μ = step size



Simulink specifications:

- Filter length- 32
 - Step size (μ) - 0.08
 - Leakage factor- 1
 - Initial value of filter weight – 0
- The design method uses a FIR Kaiser window filter of second order with beta - 0.5. The value of μ (μ) is critical. If μ is too small, the filter reacts slowly. If μ is too large, the filter resolution is poor. The selected value of μ is a compromise.

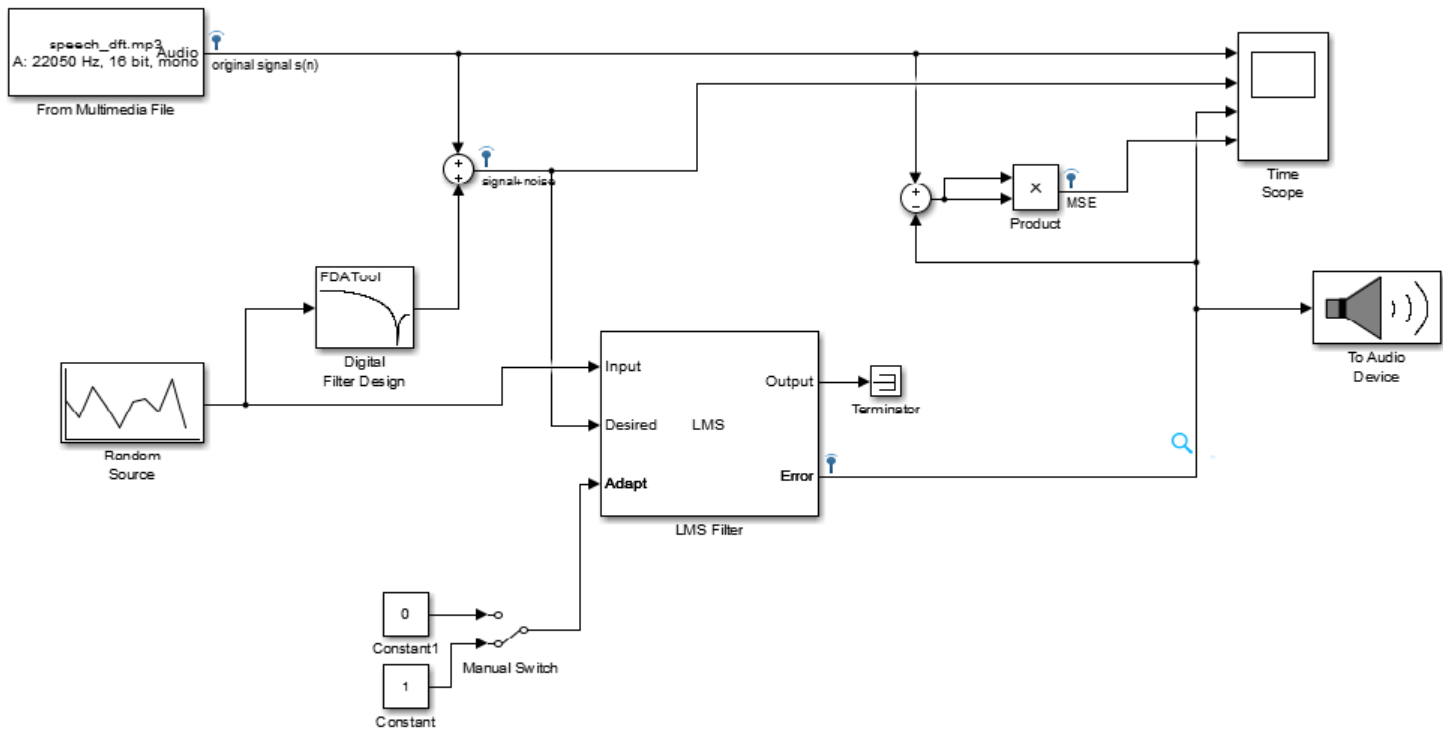


Figure 1 LMS schematic

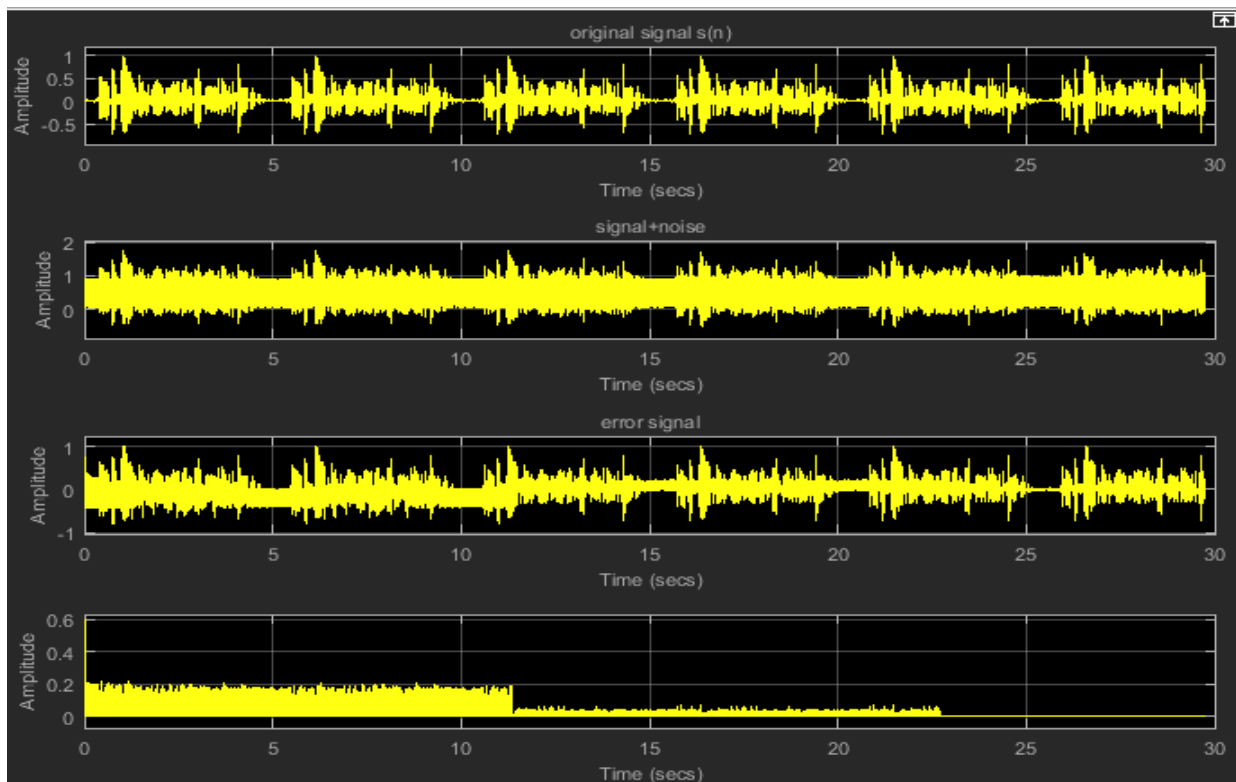


Figure 2 Simulation using Simulink

3. RLS algorithm

The Recursive Least Squares (RLS) algorithm was introduced in order to provide superior performance compared to those of the Least Mean Squares (LMS) algorithm at the expense of increased computational complexity. In the RLS algorithm, an estimate of the autocorrelation matrix is used to decorrelate the voice signal. Also, the quality of the steady state solution keeps on improving over time, eventually leading to an optimal solution. The RLS algorithm recursively solves the least squares problem. In the following equations, the constants λ and δ are user defined that represent the forgetting factor and regularization parameter respectively. The forgetting factor is a positive constant less than unity, which gives 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.

At every moment, Recursive least squares (RLS) algorithm performs a precise minimization of the whole of the squares of the wanted sign estimation error. The processing starts with known initial conditions also, in light of the information contained in the new data samples, Recursive least squares (RLS) algorithm redesigns the old estimates. These are its equations to introduce the algorithm $P(n)$ (inverse correlation matrix) should be made equivalent to where δ (regularization component) is a little positive constant.

- $y(n) = F(n) \cdot u(n)$ (i)
- $\alpha(n) = d(n) - F(n) u(n)$ (ii)
- $\pi(n) = P(n-1) u(n)$(iii)
- $k(n) = \lambda + \pi(n) u(n)$(iv)
- $K(n) = P(n-1)u(n)/k(n)$ (v)
- $F(n) = F(n-1) + K(n) \alpha(n)$(vi)
- $P1(n-1) = K(n) \cdot \pi(n)$ (vii)
- $P(n) = \{ P(n-1) - P1(n-1) \} / \lambda$ (ix)

Where,

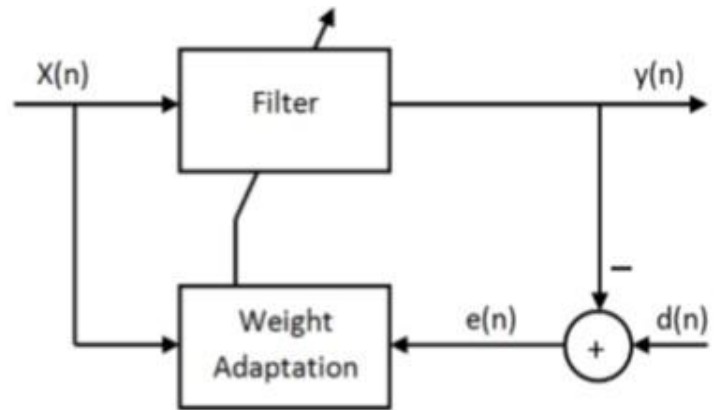
$F(n)$ = filter coefficients

$K(n)$ = gain vector

λ = forgetting factor

$P(n)$ = inverse correlation matrix of the input signal

$\alpha(n), \pi(n)$ = positive constant.



Simulink specifications:

Filter length- 32

Forgetting factor- 1

Initial value of filter weight – 0.5

The design method uses a FIR Kaiser window filter of second order with beta- 0.5

In RLS algorithm, the forgetting factor (λ) has to be chosen carefully such that its value should be very close to one in order to ensure stability and convergence of the RLS algorithm. However, this in turn poses a limitation for the use of the algorithm because small values of β may be required for signal tracking if the environment is non-stationary Z

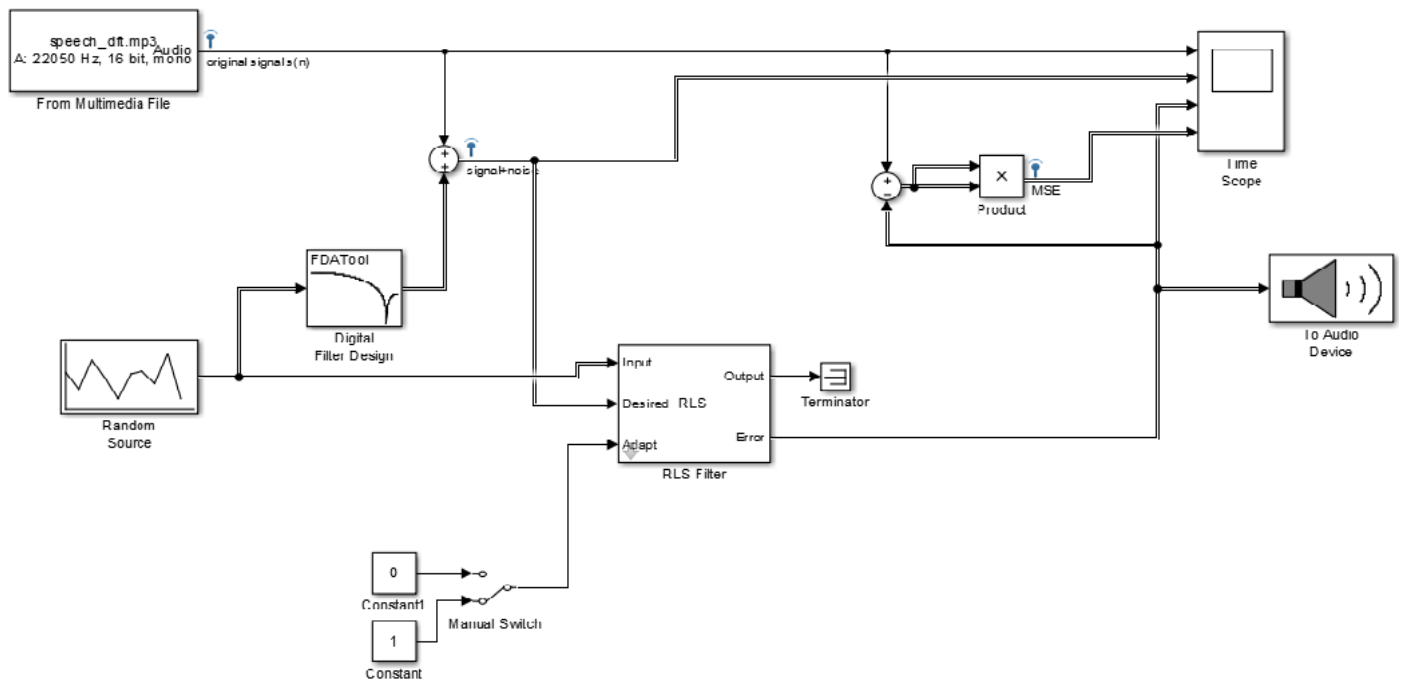


Figure 3 RLS schematic

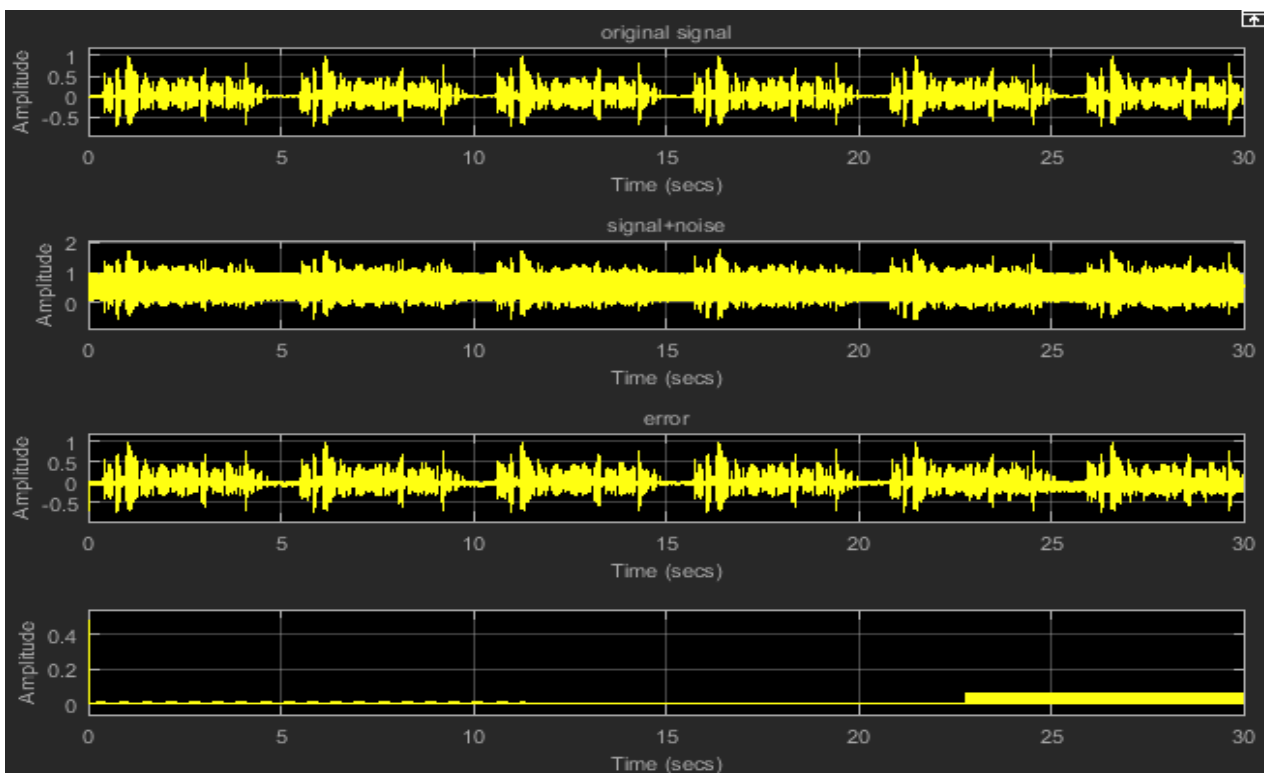


Figure 4 Simulation using Simulink

4. CONCLUSIONS

In this paper, a performance comparison between the LMS and RLS algorithms has been drawn using the SIMULINK. The simulations have been done with real time voice signal. Simulations have shown that the RLS algorithm outperforms the LMS algorithm but this high performance is with a trade-off with the high computational complexity of the RLS algorithm. One of the disadvantages of the RLS algorithm inspite of its higher convergence rate is its stability if the autocorrelation matrix is singular.

Furthermore, the performance of the aforementioned algorithms were similar to what has been investigated by the SIMULINK software.

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