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Implementation of an efficient and optimized architecture of LMS Adaptive filter to reduce the noise and error using DLMS

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ABSTRACT:
In this paper, I present an efficient and optimized architecture for the implementation of a least mean square adaptive filter. For achieving lower adaptation-delay, noise-cancellation, error and area, I use a least mean square algorithm.

Keywords:
Adaptive filters; least mean square algorithms; fixed-point arithmetic; circuit optimization

I. INTRODUCTION

We know that adaptive digital filters are used in wide applications in several digital signal processing (DSP) applications, such as, noise and echo cancelation, system identification, and channel equalization etc. Here i use a finite impulse response (FIR) filters whose weights are updated by Widrow-Hoff by using a least mean square (LMS) algorithm, because this will be considered as the simplest known adaptive filter. In adaptive digital filters least mean square (LMS) algorithm is considered as the simplest known adaptive filter. By comparing other filters the LMS adaptive filter is the most popular one not only due to its simplicity but also due to its satisfactory convergence performance. Here I use a delayed (DLMS) algorithm because the conventional LMS algorithm does not support pipelined implementation due to its recursive behaviour, and DLMS which allows pipelined implementation of the filter.

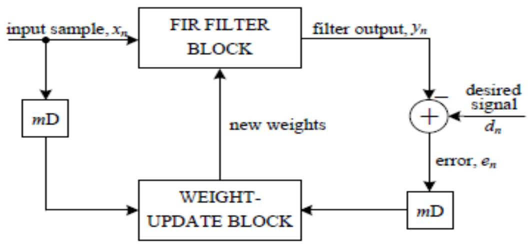

Fig.1. Structure of conventional delayed LMS adaptive filter

II. REVIEW OF DELAYED LMS ALGORITHM

The weights of LMS adaptive filter during the nth iteration are updated according to the following equations [2].

$$W_{n+1} = w_n + \mu . e_n . x_n \quad \text{......1}$$

Where

$$e(n)=d(n)-y(n) \quad \text{and} \quad y(n)=w(n) * x(n) \quad \text{......2}$$

d_n is the desired response, y_n is the filter output, and en denotes the error computed during the nth iteration. µ is the step-size, and N is the number of weights used in the LMS adaptive filter.

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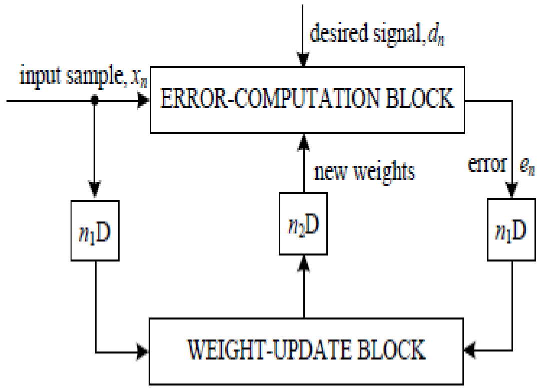


Fig.2. Structure of modified delayed LMS adaptive filter.

III. THE PROPOSED ARCHITECTURE

As shown in Fig. 2, there are two main computing blocks in the adaptive filter architecture, namely (i) the error computation block and (ii) weight-update block. In this Section, we discuss the design strategy of the proposed structure to minimize the adaptation-delay in the error-computation block, followed by weight-update block.

IV. SIMULATION METHODOLOGY

The methodology is used for the comparison between the performances of Adaptive FIR filter using Least Mean Square (LMS) algorithms. This work used MATLAB FDA tool and Simulink environment for the filter realization. We have used a Various Parameters for the simulation are Filter Structure (direct form), Filter length (32), Filter type (type2), adaptive algorithm (LMS), Simulation time (30, 50, 80), Design method (Kaiser Window _=0.5), and frequency specifications (Normalized 0-1). Figure-2 shows the MATLAB implementation of adaptive FIR Filter using LMS algorithm. Which have four outputs; input signal, input signal with noise signal, filter output and error signal produced at scope 1 and weight output (Wts) are produced at the vector scope. LMS Filters can be used to reduce the noise by separating the error between the actual output and desired signal mixed.

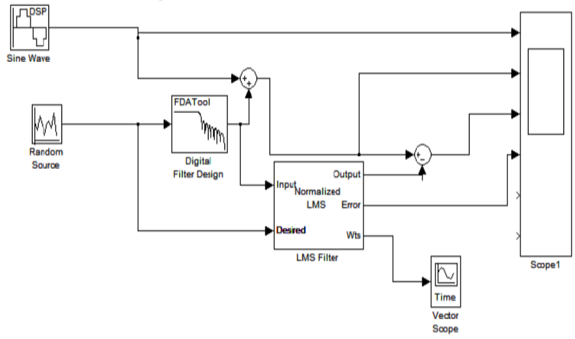
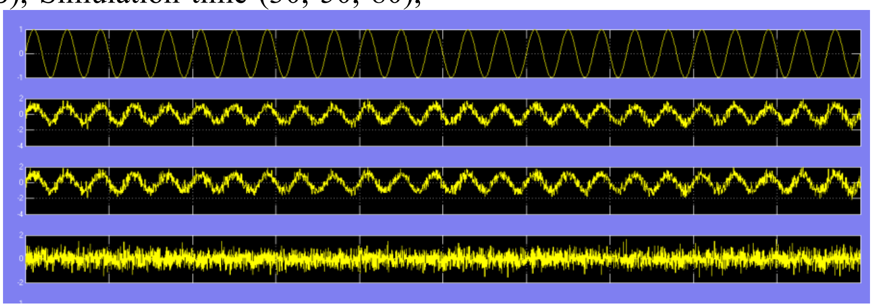


Fig.3. Implemented Adaptive FIR filter using LMS algorithm

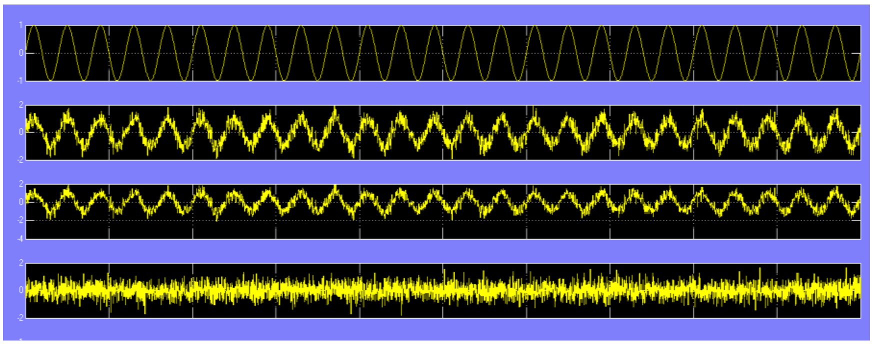
IV. EXPERIMENTAL RESULTS

a) For noise cancellations

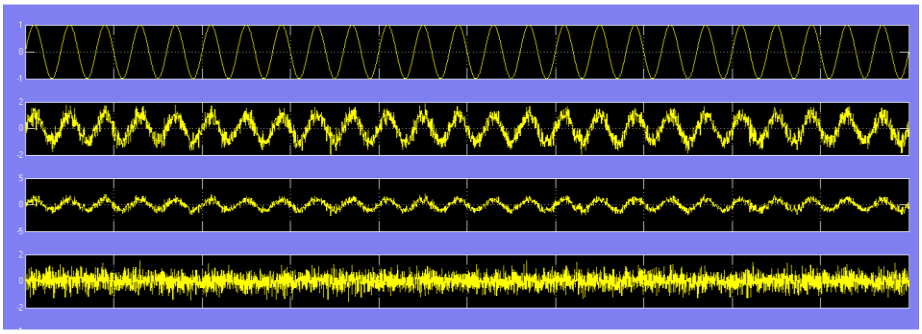
Figure a, b and c shows the output of Adaptive FIR filter using LMS algorithm for the simulation time t=30, 50, 80 second respectively. From the result it is observed that as the simulation time increases the error or noise from the signal is effectively removed or decreases using LMS filtering. More the simulation time more the reduction of noises from the signal.



(a)



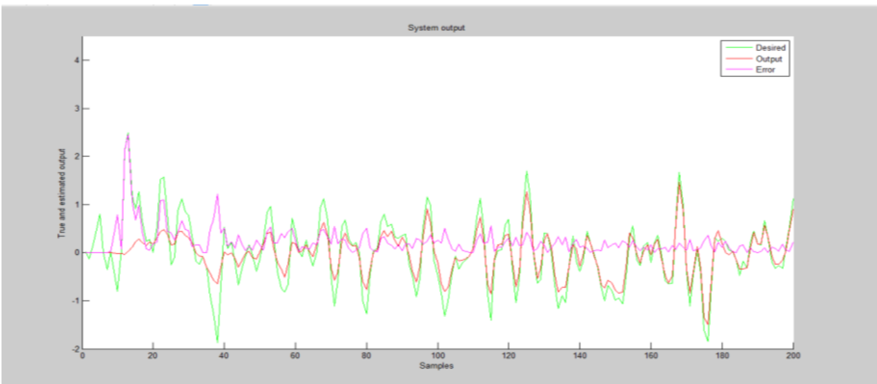
(b)



(c)

Fig.4. For noise cancellations (a) output of Adaptive FIR filter using LMS algorithm for simulation time 30 seconds (b) for simulation time 50 and (c) for simulation time 80

b) To remove the error from the signals



(d)

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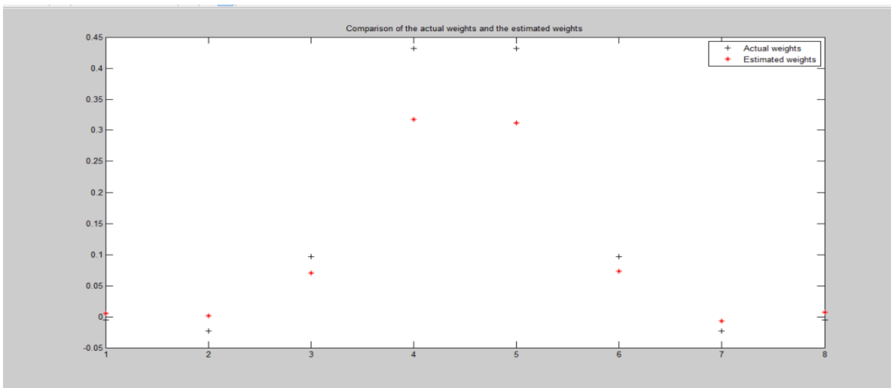
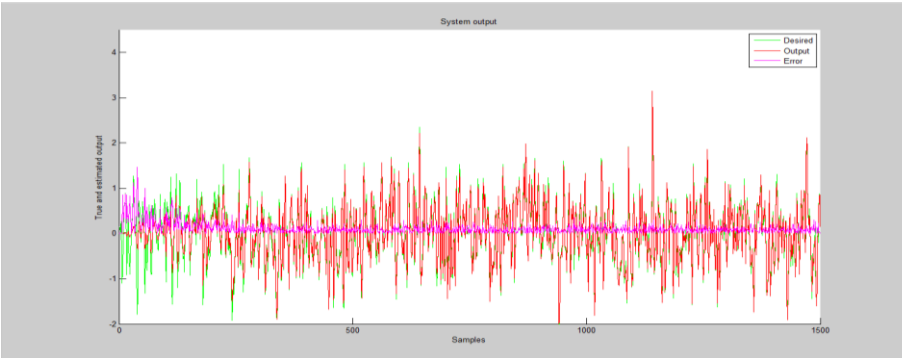
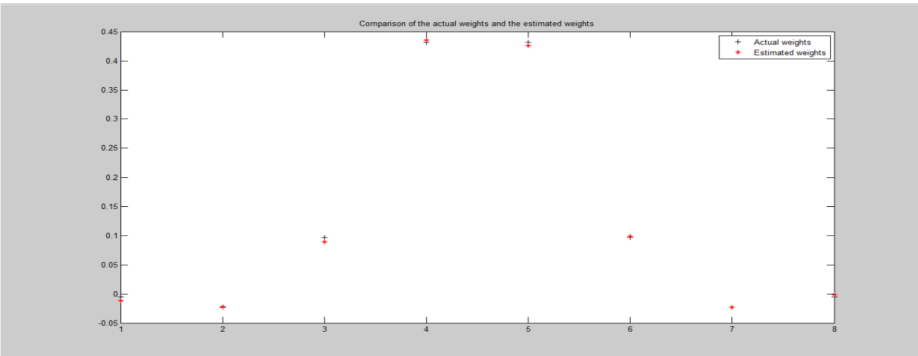


(e)



(f)



(g)

Fig.5. For error removing for (d) & (e) system order is 8 and number of iterations is 200 and For (f) & (g) system order is 8 and number of iterations is 1500

V. CONCLUSION

From the result i can conclude that the error and noise can be removed from an adaptive filter by using least mean square (LMS) algorithm.

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