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Noise Cancellation using Adaptive Filter

Sheetal Pawar[1], Sanjana Raj[2], Suraj Juware[3], Nitish Sayam[4]
[1]*Professor, Dept. of Electronics and Telecommunication Engg., Dr. D. Y. Patil Inst. of Technology, Pune, India*
[2,3,4]*U.G. Student, Dept. of Electronics and Telecommunication Engg., Dr. D. Y. Patil Inst. of Tech., Pune, India*

***Abstract*: This paper presents a new architecture for noise cancellation by implementation of adaptive filter by using Least mean square algorithm(LMS) for hardware realization The performance, as measured by the signal-to-noise ratio between the signal and its estimate, is compared to that of the commonly used filtering method. It is shown, through simulation that the proposed canceller has, on the average, better performance than the RLS canceller. Its computational time is about 10% that of the other digital filter in the cases studied, which is a substantial improvement.**

***Keywords*: Adaptive filter, LMS algorithm**

1. Introduction

Adaptive Signal Processing is concerned with the design, analysis, and implementation of systems whose structure changes in response to the incoming data. In practice, signals of interest often become contaminated by noise or other signals occupying the same band of frequency. The goal of any filter is to extract useful information from noisy data. There are numerous methods for performing weight updated of adaptive filters. Adaptive filters are widely used in various digital signal processing (DSP) applications like channel equalization, noise cancellation and echo cancellation. The tap-delay finite impulse response (FIR) filter coefficients are updated by Widrow-Hoff least mean square (LMS) algorithm. The direct form FIR filter for the implementation of LMS adaptive filter results in either zero or lower adaptation-delay. For an inner-product performance operation the critical path is large to obtain the filter response. If the input signal has high sampling period, the critical path will exceed the sampling period. To over-come this, pipelined concept is introduced in the LMS algorithm, but the LMS algorithm will not support pipelined implementation. So, delayed LMS algorithm is introduced in the adaptive FIR filter which accepts the pipelined concept. Here we will discuss about LMS adaptive filter with fixed point implementations. A normal fixed filter is designed in advance with knowledge of the statistics of both the signal and unwanted noise; the adaptive filter continuously adjusts to a changing environment through the use of recursive algorithms.

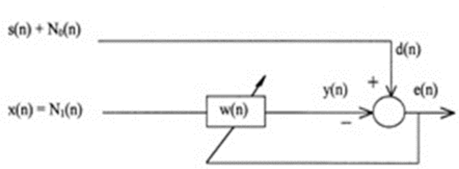

Fig. 1. Adaptive noise cancellation configuration

2. Background work of LMS Adaptive filters

A. LMS Filters

Least mean squares (LMS) algorithms are a class of adaptive filter used to mimic a desired filter by finding the filter coefficients that relate to producing the least mean square of the error signal (difference between the desired and the actual signal). It is a stochastic gradient descent method in that the filter is only adapted based on the error at the current time. It was invented in 1960 by Stanford University professor Bernard Widrow and his first Ph.D. student, Ted Hoff.

1) Related work

For every cycle the LMS algorithm based filter gets an error value and the output that equals to the difference between the present filter output and the desired signal. The resultant error signal is used to update the filter coefficient. The weights of LMS adaptive filter during the nth iteration are updated according to the following equations:

$d(n+1) = d(n) + \mu e(n) X(n)$

Where

$d(n) = [d0(n), d1(n), \ldots\ldots. d(N-1)(n)]$

$X(n) = [x(n), x(n-1), x(n-2) \ldots.. x(n-N+1)]T$

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

$y(n) = PT(n) X(n)$

$w(n)$ = desired response

Where

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$X(n) = [x(n), x(n-1), x(n-2) \ldots.. x(n-N+1)]T$

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

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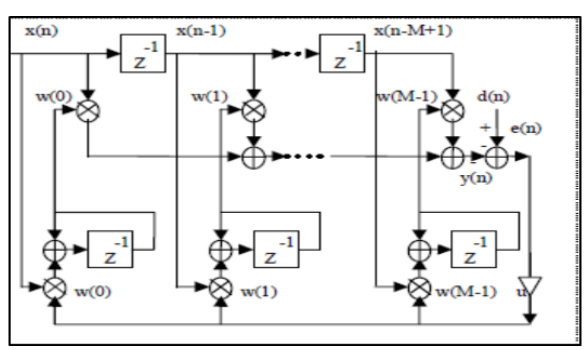

Fig. 2. Block diagram of LMS filter

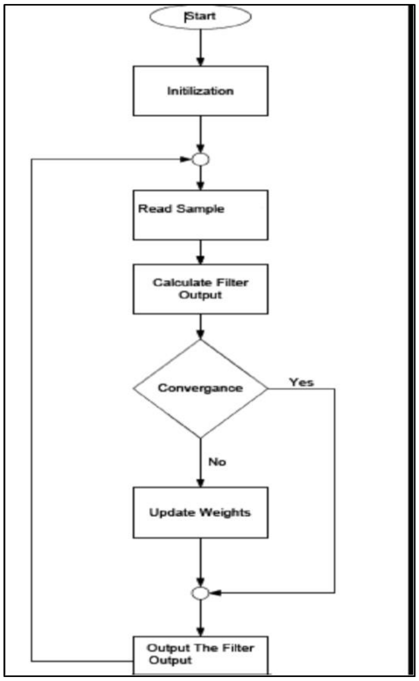

Fig. 3. Flow chart of LMS algorithm

3. Future work

The main focus of this project has been to increase the quality of services and system capacity. For that purpose, low power LMS equalizer are implemented. Its performance is evaluated and tested in FPGA. Compare to the existing system the new proposed system has high speed, low power and area. Also with little modification the LMS equalizer can be converted to decision feedback equalizer which offers the advantage of faster convergence and lesser noise enhancement. The need for equalizers arises from the fact that the channel has amplitude and phase dispersion which result in the interference of the transmitted signals with one another. The adaptive filter removes the interference in the transmission system.

4. Conclusion

This paper presented an overview on noise cancellation using adaptive filter.

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