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A survey on design and implementation of LMS and DLMS Adaptive Filter and its performance analysis based on FPGA

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Abstract— The most important issue for practical signal processing applications is removing noise, echo etc. The hardware implementation of adaptive filters is a challenging issue in real-time practical noise cancellation, echo cancellation, applications. An adaptive filter is a kind of filter that changes and updates its specifications according to the application automatically. Adaptive filter in general consists of two basic processes, filtering process and adaptive weight control process. This adaptive weight control mechanism may be different algorithms such as LMS,DLMS,RLS,NLMS which are used for error minimization.

This paper presents the survey on design and implementation of LMS and DLMS adaptive filter architectures on a Field Programmable Gate Arrays (FPGA) chip. The adaptive filters LMS and DLMS will be compared and analyzed on the basis of performance parameters such as Device utilization summary w.r.t. FPGA, Speed Factor and maximum operating frequency. The filter architecture is considered for designing and the VHDL hardware description language will be used for algorithm modeling. The practical results of simulation will be monitored using Modelsim.

Keywords— LMS, DLMS, FPGA, Adaptive filter, Modelsim, VHDL, Xilinx FPGA.

INTRODUCTION

Today, the main issues in most of the applications are noise signals or error signals due to which the originality of signals is lost. To overcome this problem, filters are used. Filters can be FIR or IIR. The filter is said to be optimum only when the statistical characteristics of the input data match the prior information on which the design of filter is based. The most efficient way for filters to be favourable is to use adaptive mechanism. But in some cases, the noise model is time varying and could not be removed by stationary- coefficient based filters. In advance, Adaptive filters are employed that could adapt their coefficients by changing the filter inputs. Adaptive filters may contain FIR or IIR filters. FIR filters are commonly used as they used forward paths only and are stable. The LMS algorithm is a well-known adaptive algorithm for updating filter coefficients in dynamic and unknown environments. However, the delay in the feedback error for updating the weights according to the LMS algorithm does not favors its pipeline implementation under high sampling rate condition. For that purpose the delayed LMS (DLMS) algorithm for pipeline implementation of LMS. The improved or delayed version of LMS is called as DLMS adaptive filter.

Microprocessors, microcontrollers and digital signal processors (DSP) chips perform fetching, decoding and execution stages sequentially and not simultaneously using fixed hardware and architecture. Therefore, they could not process data simultaneously. The FPGA chips are reconfigurable and can process data and information simultaneously for different processing applications. FPGAs have been used in a wide range of applications such as network communication, video communication and processing and cryptographic applications. The objective of the project is to present implementation of least mean square (LMS) adaptive filter and Delayed least mean square (DLMS) architectures on a Spartan Field Programmable Gate Arrays (FPGA) chip and compare their parameters.

LITERATURE REVIEW

Various researchers have been done to improve the performance parameters values for the given algorithms. Some of which are listed below.

Fohl and Matthies [1] implemented an adaptive filter on FPGA to investigate the applicability of this chip as a hardware base for real-time audio processing and they concluded that the FPGA is so suited for complex real-time audio processing. A 64-tap 9-bit LMS adaptive FIR filter for active noise control (ANC) was implemented on Altera Cyclone II FPGA considering a 24 KHZ uniform random noise signal.

Elhossini et al [2] has proposed three different architectures for implementing a least mean square (LMS) adaptive filtering algorithm, using a 16 bit fixed-point arithmetic representation. These architectures were implemented using the Xilinx multimedia board as an audio processing system. The Virtex-II FPGA chip is used to implement the three architectures.They had showed that using a pure hardware implementation results in a much higher performance with somewhat lower flexibility.

Kim and Poularikas [3] in their paper developed classic ANC, variable step size ANC and SCAF ANC for removing noise in speech signals and compared those schemes according to their performance and computation complexity. In the paper, an adjusted step size LMS (least mean squares) algorithm is proposed for possible improvements in the performance of adaptive FIR filters in nonstationary environments. Simulation results of comparing SCAF with a fixed step size LMS algorithm were presented.

Vella, and Debono [4] illustrated an LMS adaptive filter in a line echo cancellation scheme and different architectures were used to implement multiplication blocks for decreasing hardware utilization and increasing computation speed. The ever increasing data rates used in communication systems bring along the need for faster adaptive filtering systems that are capable of handling the echo tail generated. This paper describes the implementation of such an adaptive filter on a Xilinx Spartan 3 FPGA.

PROPOSED WORK

The most efficient way for filters to be optimum is to use adaptive mechanism. An adaptive filter is a filter that can updates its specifications according to application. The theory of widely used algorithm named as least mean square (LMS) algorithm was developed by widrow and Hoff in1960. LMS algorithm is an important member of the family of stochastic gradient algorithms. A significant feature of the LMS algorithm is its simplicity. It does not require matrix inversion nor pertinent correlation function.The simplicity of LMS algorithm made it the standard against other linear adaptive filtering algorithm.

The LMS algorithm is a linear adaptive filtering algorithm which consists of two processes. A filtering process, which involves firstly computing the output of filter in response to input signal and secondly generating an estimation error by comparing this output with desired response. Figure 1 shows simple block diagram of adaptive filter. An adaptive process, which involves the automatic adjustment of parameters of filter in accordance with estimation error.

In filtering process the filter may be FIR or IIR, FIR filters are commonly used as they used forward paths only and are stable. This adaptive weight control mechanism may be different algorithms such as LMS,DLMS,RLS,NLMS which are used for error minimization. LMS algorithm is a least mean square algorithm used among different algorithms due to its simplicity, low computational processing tasks and high robustness.

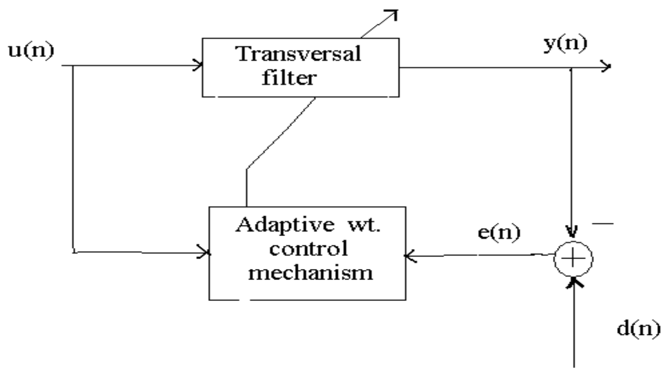

FIG.1 BLOCK DIAGRAM OF ADAPTIVE FILTER

The combination of these two processes form a feedback loop as shown in fig. 1. First we have transversal filter around which LMS algorithm is built, responsible for filtering process and we have adaptive control process on tap weights of filter thus called as adaptive weight control mechanism.

Details of transversal filter are presented in fig 2. The tap inputs x(n),x(n-1)........x(n-N+1) form the elements of N-by-1 tap input vector x(n) where (N-1) is the no. of delay elements. The tap weights w_0 (n),w_1(n), ...w_{N-1}(n) form the elements of N-by-1 tap weight vector w(n). The value computed for this vector using LMS algorithm represents an estimate whose expected value may come close to weiner solution w_0 as the no. of iterations n, approaches infinity. During filtering process, the desired response d(n) is supplied for processing alongside the tap input vector x(n). Given this input , the transversal filter produces an output y(n) used as an estimate of desired response d(n).

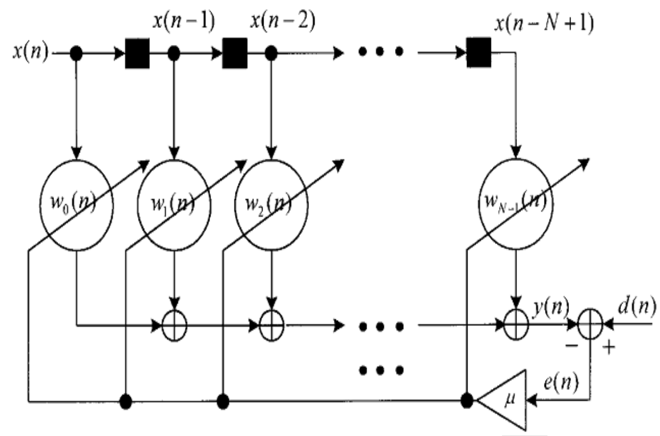

FIG. 1 LMS ADAPTIVE FILTER

Accordingly following equations are defined,

Filter output:

$$y(n) = w(n)\, x(n) \qquad \text{..............(1)}$$

Estimation error:

$$e(n) = d(n) - y(n) \qquad \text{.............(2)}$$

In adaptive weight control mechanism a scalar version of inner product of estimation error e(n) and tap input x(n-k) for k= 0, 1, 2,……., N-2, N-1; is processed. The result so obtained defines correction applied to tap weight $w_k(n)$ at iteration (n+1). The scaling factor used in this computation is denoted by positive quantity 'μ' called as step size parameter. Step size (μ) plays an important role in deciding the error and then weights. But there is a bound on this step size which is denoted as

$$0 < \mu < 2/MS_{max} \qquad \text{………………..(3)}$$

Where,

S_{max} is the value of power spectral density of tap input x(n) and M is the filter length.

Tap weight adaptation:

$$w(n+1)=w(n) + \mu x(n)e(n) \qquad \text{………………(4)}$$

If we observe, that LMS algorithm requires only 2M+1 multiplications & 2M additions per iterations, where M is the number of tap weights. This is repeated and the new weights minimize the error signal. In other words the computational complexity of LMS algorithm is less than the steepest descent algorithm which is time consuming.

DLMS is a delayed least mean square algorithm which has a pipelined architecture.In the conventional LMS adaptive filter, the estimated signal in each data interval is computed and subtracted from the desire signal. The error is then used to update the tap coefficients before the next sample arrives. In some practical applications, the LMS adaptation scheme imposes a critical limit on its implementation. LMS algorithm uses the feedback-error corresponding to the nth iteration for updating the filter weights to be used for computing the filter output for the (n+1)th iteration. The DLMS algorithm is similar to the LMS algorithm, except that in case of DLMS algorithm, the weight increment terms to be used in the current iteration are estimated from the error value and input samples corresponding to a past iteration. Figure 3 shows the structure of conventional DLMS algorithm.

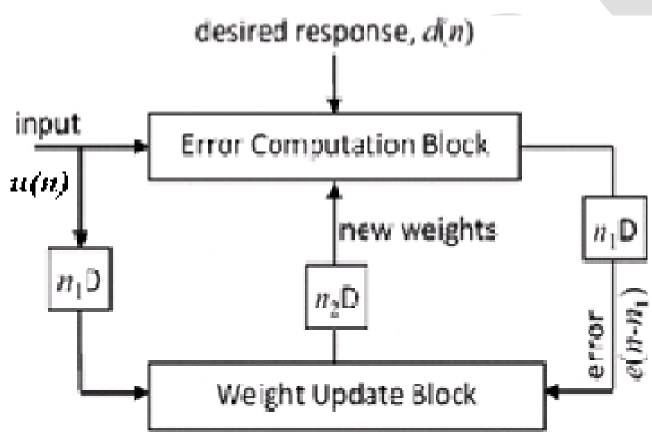

FIGURE 2 DLMS ADAPTIVE FILTER

The weight update equation algorithm is given by,

$$w(n + 1) = w(n) + \mu\, u(n - D)\, e(n - D) \qquad \ldots\ldots\ldots\ldots\ldots(5)$$

From the figure 3 it is clear that the error signal is delayed by n_1 number of cycles and then given to weight update block. Same amount of delay is provided to the input signal. In the weight update block these signals are utilized according to the equation 6.Then further n_2 number of cycles delay is given and then this output is distributed to FIR filter in terms of new weights.

CONCLUSION

Here in this paper the study of adaptive digital LMS and DLMS filters is proposed. The detailed study on the different architectures i.e. on LMS and DLMS is efficiently carried out. It is observed that the LMS filters do not support pipelining, thus for handling critical paths and further error minimization DLMS adaptive filter is used. The delay introduced in the DLMS filter at the input as well as at the error output block updates the weights further. Thus, the new updated weights will improve the performances.

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