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FPGA Implementation of Lms Filter

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Abstract:— **LMS algorithm is one of the algorithm which is used in digital Signal Processing used to minimize the noise present in the input signal. This algorithm is best due to its simplicity and efficiency. The algorithm is implemented in many of the applications.FPGA implementation of the LMS algorithm is done using Simulink model.**

***Index Terms*—FPGA (Field Programmable Gate Arrays),LMS(Least Mean Square).**

I. INTRODUCTION

The LMS algorithm was introduced by Widrow and Hoff in 1959. LMS generates the filter coefficients to avoid the interference at the input of the signal. LMS supports iterative procedure which identifies corrections which leads to the minimum MSE. LMS algorithm is simple and it does not require a function for matrix inversions. It is a most commonly used adaptive algorithm. It provides stable and robust performance for various signal conditions. Convergence speed is less compared to other complicated algorithms. These are used were the statistical parameters that changes over a period of time. The step size is varied according to the input of the signal to avoid the noise. Fig.1 shows adaptive filter.

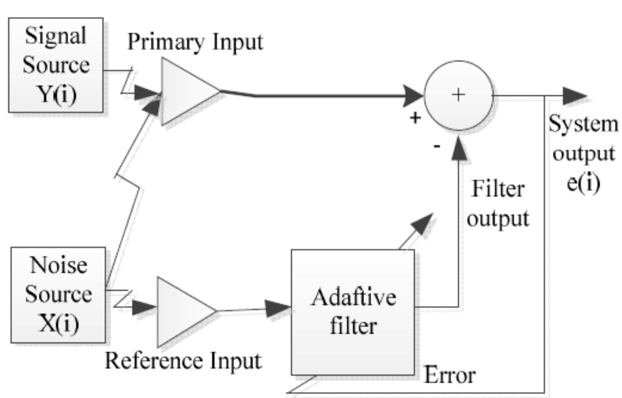

Fig 1.Block Diagram of Adaptive Filter

II. LMS ALGORITHM

The most excellent characteristic of LMS algorithm is robustness and simplicity. It has a simple structure and ease of implementation in terms of hardware. The main advantage of LMS is that it does not require prior knowledge of signal statistics. It identifies filter weights which reduces the noise. LMS is simple, low computational complexity and has fast convergence rate.

∇_k is obtained by the equation:

$$\nabla_k = \frac{dJ}{dW} = -2P + 2RW = 0 \qquad (1)$$

The LMS algorithm is a practical method to obtain W_k in real time

$$W_{opt} = R^{-1}P \qquad (2)$$

W_{k+1} *is given by*

$$W_{k+1} = W_k + 2\mu e_k X_k \qquad (3)$$

where:

$$e_k = y_k - W_k^T X_k \qquad (4)$$

II. ADVANTAGES AND APPLICATIONS OF LMS

Advantages

1. This system has an advantage over other implementation algorithm in terms of area, power and speed.
2. Area consumption is less.
3. Less power consumption.
4. Faster convergence.
5. Simple to implement.

Applications

1. The algorithm is used to eliminate noise contained in input signals.
2. This technique has been applied in various communication and industrial appliances such as hands free phones, machineries and transformers.
3. This technique has been implemented in biomedical and image processing applications.

III. SIMULATION AND RESULTS

MATLAB simulations of LMS, RLS and NLMS were used to investigate the effectiveness of the adaptive filters for recovery of a signal corrupted with noise. Simulation results were also used to investigate the differences between LMS, RLS and NLMS to determine which would be better suited to be implemented in hardware.LMS uses huge number of registers, gates and LUTs which is shown in Fig 2.

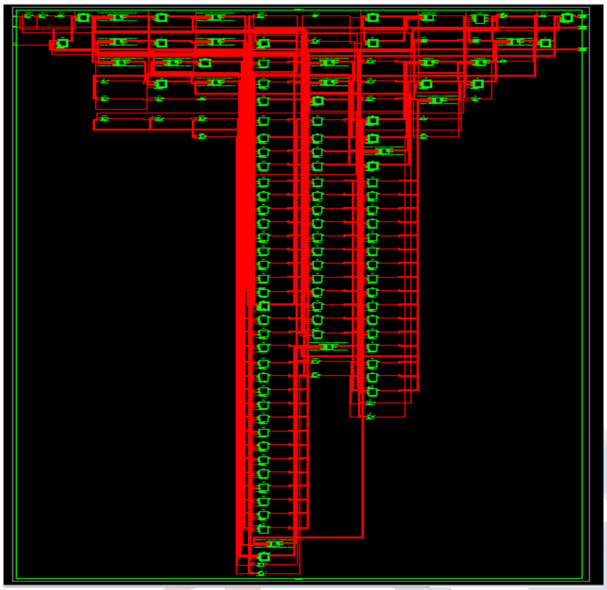

Fig. 2 RTL Schematic of Adaptive Filter Using LMS Algorithm

The design summary of the filter is shown in Fig. 3.

Device Utilization Summary (estimated values)			
Logic Utilization	Used	Available	Utilization
Number of Slice Registers	371	126800	0%
Number of Slice LUTs	207	63400	0%
Number of fully used LUT-FF pairs	149	429	34%
Number of bonded IOBs	51	210	24%
Number of BUFG/BUFGCTRLs	1	32	3%
Number of DSP48E1s	8	240	3%

Fig. 3 Design Summary of Adaptive Filter Using LMS

The noise generated by the filter is shown in Fig. 4.and the Simulink model is shown in Fig.5.

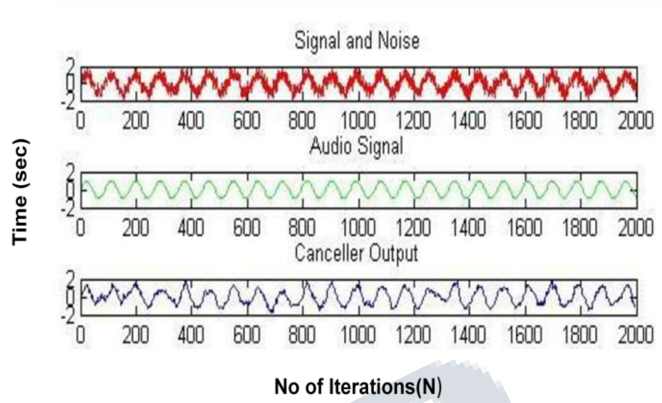

Fig. 4 Noise Canceller Output Waveform for LMS

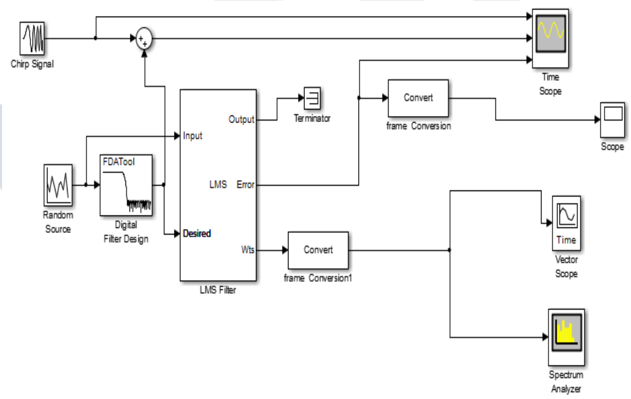

Fig. 5 Simulink Model of LMS Algorithm

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