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Implementation of adaptive LMS Algorithm for Cancellation of Noisy Magnetic Resonance Imaging Environment Using FPGA

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Abstract — *This paper describes the method of adaptive LMS algorithm for noise cancellation. In MRI method acoustic noise is produce due to the communication between medical staff and patient also produces due to high magnetic (3-T) field and higher power. This acoustic noise interferes with diagnosis and imaging process. In MRI system level of strong acoustic noise is greater than 120dB SPL (Sound pressure level). In MRI system signal to noise ratio is very low due to available of strong acoustic noise. For removing the noise and recovering the original speech signal in MRI we employ a two channel adaptive least mean square algorithm. In this paper, the main aim is remove the noise in speech signal and improve the quality of speech signal especially for noisy magnetic resonance imaging environment*

Keywords*:-Speech enhancement, MRI noise, least mean square, background noise, FPGA.*

I. INTRODUCTION

In many speech related systems like telecommunication, biomedical signal processing, automatic speech recognition, etc background noise is common problem. This background noise is degraded the quality and intelligibility of the original speech signal. So, we need speech enhancement technique. Speech enhancement technique can be classified as single channel and dual channel. In single channel, there is only one microphone and therefore only one acquisition channel or mixture is available which will just give spectral information about signal. The performance of single channel is limited. In dual channel speech enhancement techniques, a reference signal for the noise is available and hence adaptive noise cancellation technique can be applied.

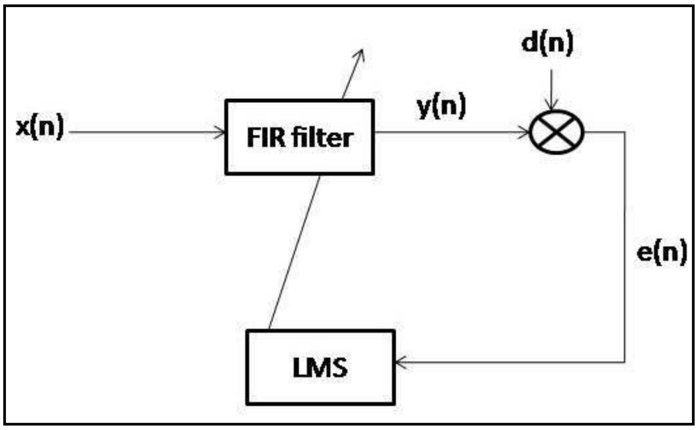

Fig.1 Block diagram of LMS algorithm [2]

Shown in Fig.1 in the two channel speech enhancement technique the first channel x (n) is dedicated to measure the original speech signal and noisy signal is called primary signal. The second channel d(n) is used to measure information about background noise is called reference signal. In all speech enhancement method, the primary signal and reference signal are assumed to be uncorrelated. Y (n) is the output of the FIR filter. Here, we are use direct form FIR filter. e(n) is error signal obtained by subtracting filter output and desired signal and it is feedback to LMS filter.

II. IMPLEMENTATION OF TWO CHANNEL LMS ALGORITHM

The output of the filter y (n) is obtained by convolution process of input signal and weight vector function w (n).

$$y\ (n)= x\ (n) * w(n) \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(1)$$

Error signal is obtained by subtracting reference signal from the filter output.

$$e\ (n) = y\ (n) - d\ (n) \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots(2)$$

The tap weight vector of filter is automatic updated according to estimation error. Weight updated is given by below equation. Where, μ is the step size parameter.

$$W\ (n+1) = w\ (n) + 2\mu e\ (n)\ x\ (n) \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ (3)$$

Implementation of LMS algorithm is very simple and less stable compare to other adaptive algorithm like normalize least mean square algorithm and recursive leas algorithm. LMS algorithm has a fixed step size for each iteration. The only disadvantage is its weak convergence. Each iteration the LMS algorithm requires 2N additions and 2N+1 multiplication it is indicate less complexity. Where n= 0, 1, 2..., n. Convergence speed is depend on step size parameter when step size parameter is increased need more iteration is required for convergence.

III. SIMULATION RESULT

An adaptive LMS filter is design with a direct-form FIR filter coded in VHDL code and with LMS algorithm written in VHDL code executing on the Xilinx ISE 14.1. The system architecture of adaptive LMS filter consists of main four modules such as 16 bit adder, 16 bit subtract or, multiplier and delay. The simulation results obtained practically are found to be equivalent to the result obtained by mathematical calculation.

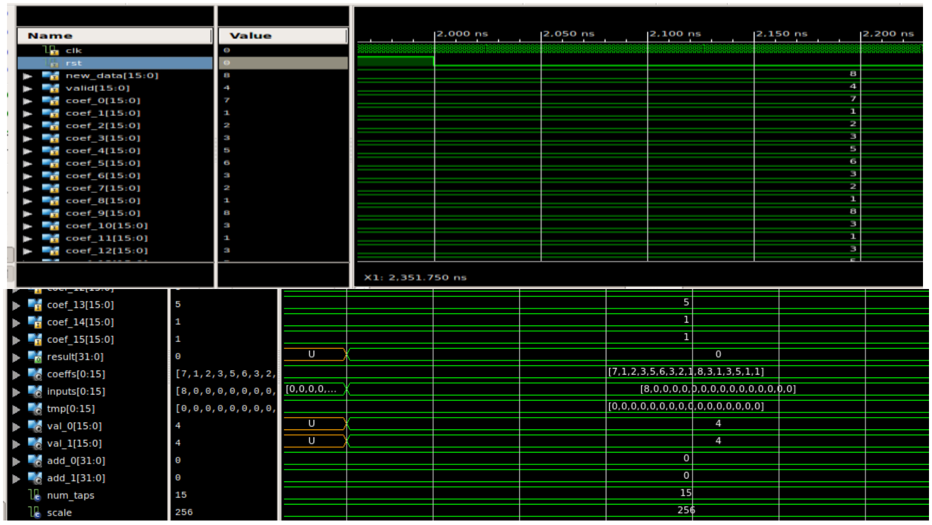

Fig.2. Simulation result of FIR filter algorithm

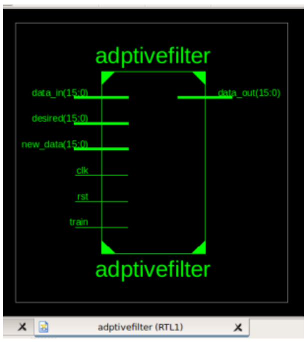

Fig.3. RTL view of adaptive LMS algorithm

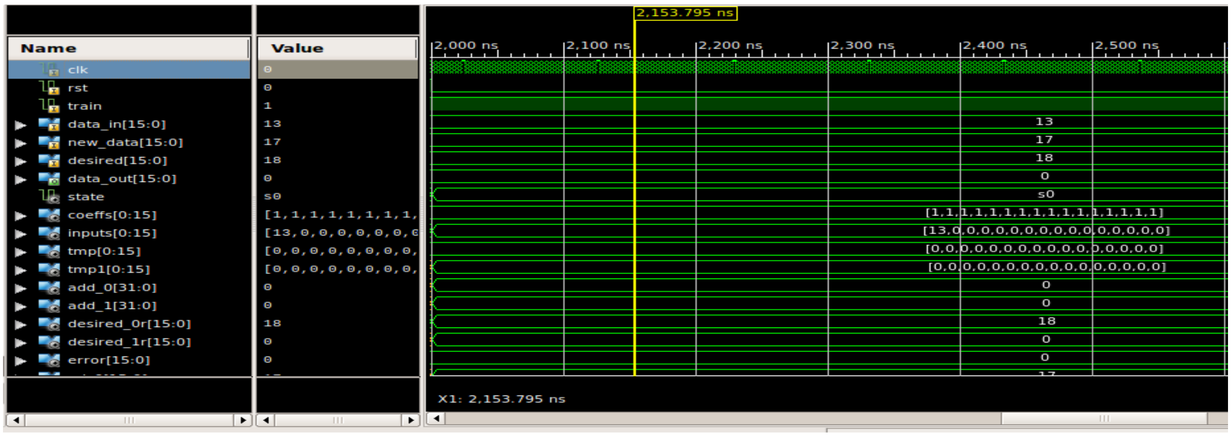

Fig.4. Simulation result of adaptive LMS filter

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Table 1. Devices utilization summary: Selected Device: 3s50pq208-5

Number of Slice Flip Flops	701 out of 1536
Number of bonded IOBs	67 out of 124

Table 2. Timing Summary

Minimum period	15.579ns
Maximum Frequency	64.187MHz

Table 3. List of parameter

Filter length	Step size parameter	Noise variance	MSE
16	4	13kHz	0.14

IV. CONCLUSION

FPGA provide an attractive platform for many complex signals processing algorithm.LMS algorithm is more suitable for MRI noise. Fixed step size parameter provides a low convergence speed. User can define the filter length and step size parameter as per applications. LMS algorithm achieve higher signal to noise ratio up to 17dB.

V. REFERENCES

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