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Implementation Of Least Mean Square Algorithm For Speech Enhancement Using Fpga

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Abstract: - The aim of this paper is to investigate the speech signal enhancement using Adaptive filters. Background noise elimination is very important in many applications like hearing aids, forensic applications, military, cellular environments etc. The adaptive filters has advantages like low processing delay and better tracking of the trajectory of non- stationary signals. If the evaluation of background noise is more slowly than the speech, i.e., noise signal is more stationary than the speech signal, we can easily estimate the noise during the pauses in speech. Any other way it is more difficult to estimate the noise which results in degradation of speech. In order to enhance the quality and intelligibility of speech, here we proposed a design based on LMS algorithm to remove noise from speech signal using VHDL based on commercially available FPGAs. In this design the LMS algorithm used as a noise canceller and the reference signal was adaptively filtered and subtracted from primary signal to obtain the estimated speech signal.

Keywords: - *LMS, Noise canceller, FPGA, VHDL and Speech Enhancement.*

I. INTRODUCTION

To model the relationship between two signals in real time in an iterative manner, An Adaptive filter is used which is a computational device. Adaptive filters are often realized either as a set of program instructions running on an arithmetical processing device such as a DSP chip or as set of logic operations implemented in a field programmable gate array (FPGA) or in a custom VLSI integrated circuits.
Adaptive filters, as a part of digital signal system, have been widely used in communication industry, as well as one of the applications such as adaptive noise cancellation.
However, implementation of adaptive filter takes a great deal and becomes a very important role in digital system design. Adaptive filters are usually implemented in DSP processor because they are capable of performing fast floating-point arithmetic operations. But now a day in signal processing, FPGA becomes an important competitor by providing a lot of facilities to designers in the market. In addition, FPGA is a form of programmable logic that can offer flexibility for repetitive reconfiguration. Because of slices organized as array of rows and columns in FPGA, a great deal of parallelism can be explored.
Although it is not efficient to use floating-point arithmetic in FPGA due to its need for an enormous area, it is sufficient to use fixed-point arithmetic for the adaptive filter to work well. The algorithm used to update the filter coefficient is the Least Mean Square (LMS) algorithm which is known for its simplification, low computational complication, and better performance in different running environments. In terms of the number of iterations required for convergence, LMS algorithm is seen to perform very well when compared to other algorithms used for implementing adaptive filters.

II. SPEECH ENHANCEMENT

Speech enhancement is concerned with improving some perceptual aspect of speech that has been degraded by additive noise. In most applications the aim of speech enhancement is to improve the quality and intelligibility of degraded speech. The improvement in quality is highly desirable as it can listener fatigue particularly in situations where the listener is exposed to high levels of noise for long period of time. The need

to enhance speech signals arises in many situations in which the speech signal originates from a noise location or is affected by noise location or is affected by noise over a communication channel.

Speech enhancement algorithms can therefore be used to improve the quality of speech at the receiving end. The solution to the general problem of speech enhancement depends largely on the application at hand, the characteristics of noise source or interference, the relationship of the noise to the clean signal.

Critical to the design of speech enhancement algorithms is knowledge of the range of speech and noise intensity levels in real-world scenarios. From that we estimate the range of signal-to-noise ratio [SNR] levels encountered in realistic environment. This is important since speech enhancement algorithms need to be effective in suppressing the noise and improving speech quality within that range of SNR level.

III. ADAPTIVE FILTERS

Adaptive filters learn the statistics of their operating environment and continually adjust their parameters accordingly. This phase presents the theory of the algorithms needed to train the filters. In practice, signals of interest often become contaminated by noise or other signals occupying the same band of frequency. When the signal of interest and the noise reside in separate frequency bands, traditional linear filters are able to extract the desired signal . However, when there is spectral overlap between the signal and noise, or the signal or interfering signal's statistical data change with time, fixed coefficient filters are inappropriate.

The goal of any filter is to extract useful information from noisy data. Whereas a normal fixed filter is designed in advance with knowledge of the statistics of both the signal and the unwanted noise, the adaptive filter regularly adjusts to a changing environment through the use of recursive algorithms. This is useful when either the statistics of the signals are not known beforehand of change with time.

The adaptive filter has advantages like lower processing delay and better tracking of the trajectory of non-stationary signals [2]. These are essential characteristics in applications such as noise estimation, echo cancellation, adaptive delay estimation and channel equalization, where low delay and fast racking of time-varying environments and time-varying processes are important objectives. The existence of a reference signal which is hidden in the fixed-filter approximation step, defines the performance norm. The general adaptive-filter structure is shown in Fig1.

The LMS algorithm aims to minimize a cost function, $V(w(n))$, at each time step n, by a suitable choice of the weight vector $w(n)$. The strategy is to update the parameter estimate proportional to the instantaneous gradient value, $dV(w(n))/dw(n)$, so that:

$$w(n+1) = w(n) - \mu\, dV(w(n))/dw(n) \qquad (2)$$

where μ is a small positive step size and the minus sign ensures that the parameter estimates descend the error surface. The cost function, $V(w(n))$, which minimizes the mean-square error, results in the following recursive parameter update equation

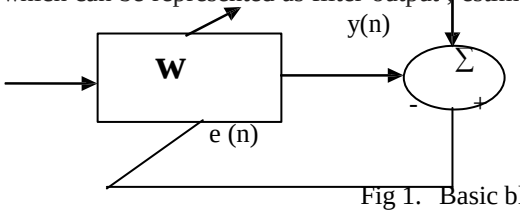
$w(n+1) = w(n) - \mu x(n)\,(y(n) - y_{est}(n))$ (3) The recursive relation for updating the tap weight

Equation

(1) may be rewritten as

$$w(n+1) = w(n) - \mu x(n)(y(n) - xT(n)w(n)) \qquad (4)$$

which can be represented as filter output , estimation error and tap weight adaptation .x(n) d(n)



$$y_{est}(n) = wT(n)x(n) \qquad (5)$$

$$e(n) = y(n) - y_{est}(n) \qquad (6)$$

$$w(n+1) = w(n) + \mu x(n)e(n) \qquad (7)$$

Fig 1. Basic block diagram of Adaptive filter

The LMS algorithm requires only 2N + 1 multiplications and 2N additions per iteration for an N tap weight vector. Therefore it has a relatively simple structure and the

The basic configuration of an adaptive filter, operating in the discrete-time domain k, is illustrated in Figure 1. In such a scheme, the input signal is denoted by x(n), the reference signal d(n) represents the desired output signal (that usually includes some noise component), y(n) is the output of the adaptive filter, and the error signal is defined as

$e(n) = d(n) - y(n)$ (1) The error signal is used by the adaptation algorithm to update the adaptive filter coefficient vector w(n) according to some performance criterion. In general, the whole adaptation process aims at minimizing some metric of the error signal, forcing the adaptive filter output signal to approximate the reference signal in a statistical sense.

IV. LMS ALGORITHM

The LMS algorithm is a stochastic gradient algorithm, which uses a fixed step-size parameter to control the updates to the tap weights of a transversal filter, (Widrow and Hoff 1960). The algorithm aims to minimize the mean-square error, the error being the difference in $y(n)$ and $y_{est}(n)$. The dependence of the mean-square error on the unknown tap weights may be viewed as a multidimensional paraboloid referred to as the error surface .The surface has a uniquely defined minimum point defining the tap weights for the optimum Wiener solution. However, in the non-stationary environment this error surface is continuously changing, thus the LMS algorithm needs to be able to track the bottom of the surface. hardware is directly proportional to the number of weights.

The LMS algorithm is by far the most widely used algorithm in adaptive filtering for many reasons. The main features that attracted the use of the LMS algorithm are low computational complication, proof of convergence in static environment, unbiased convergence in the mean to the Wiener solution, and stable behaviour when implemented with finite-precision arithmetic. The convergence analysis of the LMS presented here utilizes the independence assumption.

V. FPGA IMPLEMENTATION

FPGA Implementation of an Adaptive Filter architecture using LMS algorithm. Adaptive filter is commonly used for a wide range of applications such as Echo cancellation, Prediction, Noise cancellation, Adaptive interference cancelling, system identification, Digital communication Receiver, Radar signal processing, Equalizations of communication channels, Biomedical signal enhancements, Navigational systems, Adaptive antenna systems beam forming etc..

In this paper, adaptive filter architecture implemented mainly application for noise cancellation. In the noise cancellation the architecture is taken in both sequential as well as parallel. For implementing Adaptive filter architecture, LMS algorithm is used because of low computational complexity, simplicity and gives its better performance in different running environments.

In all adaptive filter architecture try to minimize error i.e. minimization of different between the desired output and the real one for all the input vectors. Nowadays, the use of Field programmable gate arrays (FPGAs) is growing. Field programmable gate arrays (FPGAs) are widely used in many areas such as audio and video, Digital signal processing, Image signal Processing, Digital communication systems, mobile communication system and many other embedded applications, because of their high performance, parallel processing ability and flexibility.

There are usually two ways to implement the LMS algorithm, hardware implementation and software implementation [6], [7]. The hardware implementation of the algorithm in an FPGA has good real-time ability, but requires large resources.

VI. HARDWARE IMPLEMENTATION

In this work, the ANC is implemented in the Xilinx virtex-5 kit (Genesys) board. The Genesys circuit board is a complete, ready-to-use digital circuit development platform based on a Xilinx Virtex 5 LX50T. The large on-board collection of high-end peripherals, including Gbit Ethernet, HDMI Video, 64-bit DDR2 memory array, and audio and USB ports make the Genesys board an ideal host for complete digital systems, including embedded processor designs based on Xilinx's MicroBlaze. Genesys is compatible with all Xilinx CAD tools, including ChipScope, EDK, and the free WebPack[9].

The Virtex5-LX50T is optimized for high performance logic and offers: 7,200 slices, each containing four 6-input LUTs and eight flip-flops,1.7Mbits of fast block RAM, 12 digital clock managers, six phase-locked loops, 48 DSP slices and 500MHz+ clock speeds.

A. Simulation Results

A review of adaptive filters shows that the LMS algorithm is still a popular choice for its stable performance and high speed capability. The other advantage of the LMS over other adaptive algorithm is its high convergence rate. The high- speed capability and register rich architecture of the FPGA is ideal for implementing LMS. The proposed model is loaded
on XILINX FPGA board. It is implemented on virtex-5 LX50T. After synthesizing in Xilinx project navigator we got RTL schematic diagram of our proposed design which is shown in Figure4.

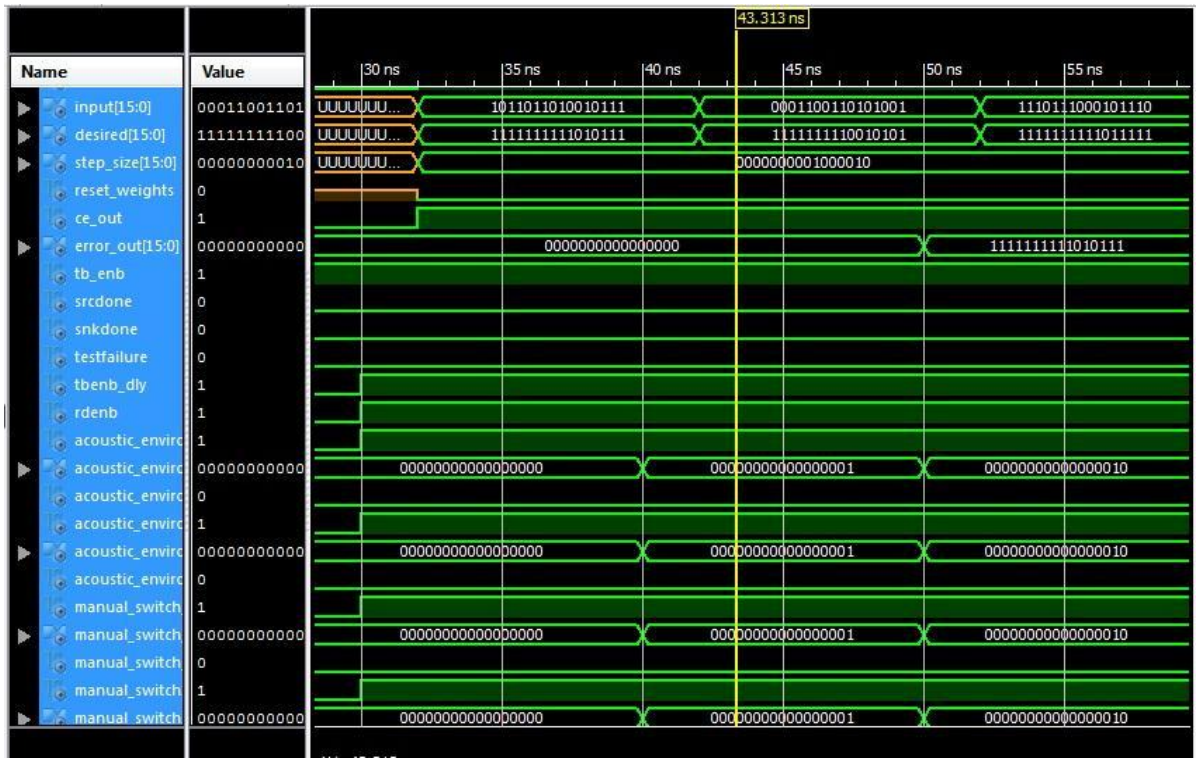


Fig 2. Xilinx ISE Simulation Results

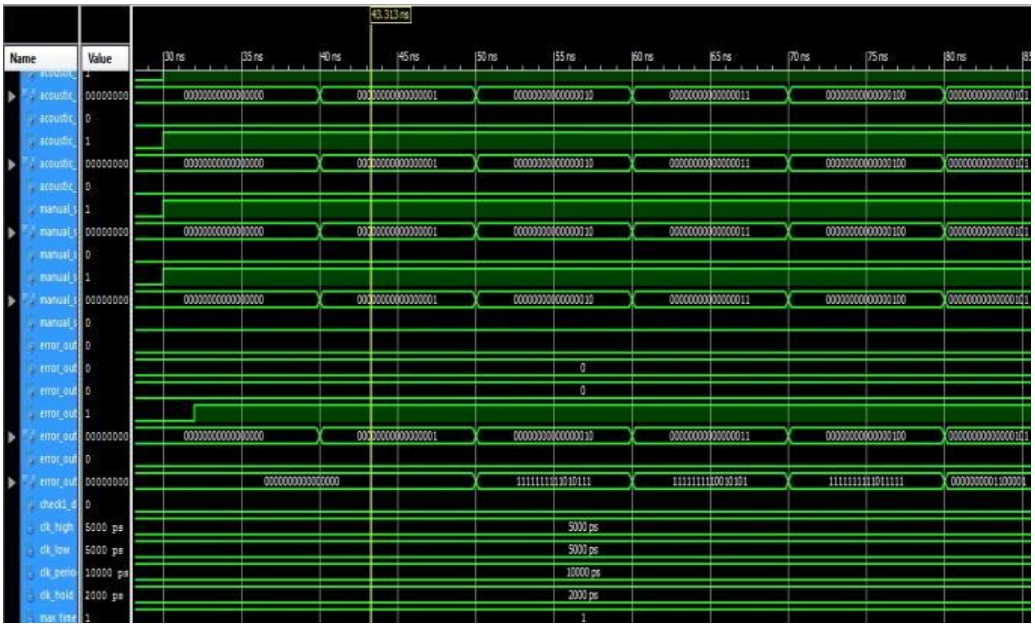


Fig 3. Simulation Wave Forms

The Xilinx ISE 13.4 software package is used in this project to implement the VHDL program. The system can be simulated to observe and verify its behaviour and the results were showed in fig. 2 and fig. 3. The design is used for the implementation on the FPGA chip.

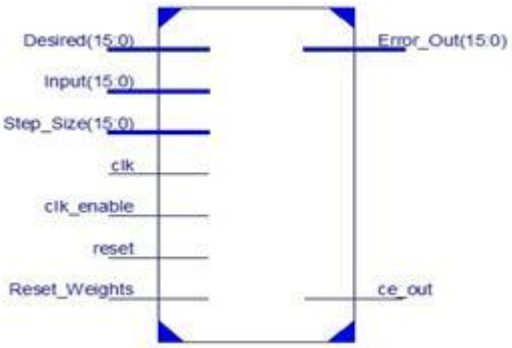


Fig 4. RTL schematic of LMS Filter

B. Chipscope Results

ChipScope provides you with a convenient software based interface for controlling the integrated logic analyzer, including setting in the triggering options and viewing the waveforms. Time required for simulating complex design for all possible test cases becomes prohibitively large and simulation approach fails. For rapid testing, such designs can be loaded on to the target FPGAs and tested by applying test inputs and directly observing their outputs[13].

The ChipScope Pro tools integrate key logic analyzer and other test and measurement hardware components with the target design inside the FPGA. Computer based software tool communicate with these hardware components and provide a designer robust logic analyzer solution. Waveforms are showed in Fig. 5.

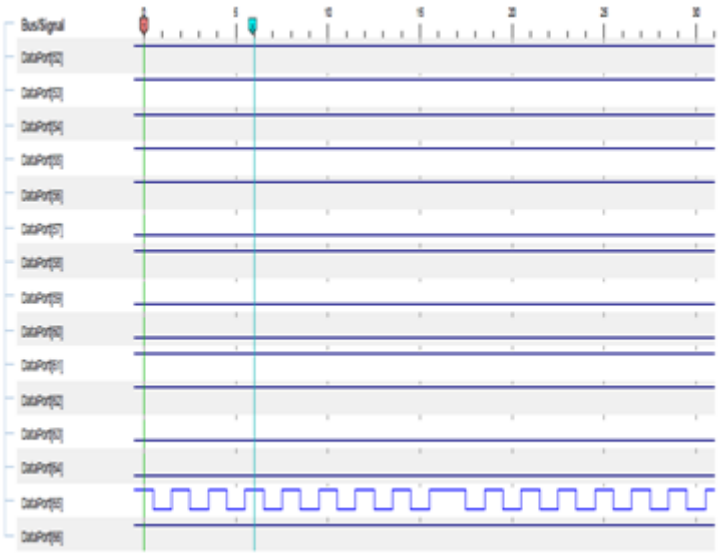


Fig 5.Chipscope waveform

C. Timing Summary

1. Speed Grade : -2
2. Minimum period : 23.452ns
(Maximum Frequency: 42.641MHz)
3. Minimum input arrival time before clock : 15.361ns
4. Maximum output required time after clock : 2.826ns
5. Maximum combinational path delay : 3.237ns

VII. CONCLUSION

Based on LMS algorithm an adaptive equalizer is designed and implemented using hardware description language VHDL. The design is simulated on Xilinx ISE 13.4 and a satisfactory result is obtained.

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