Novateur Publication's
International Journal of Innovation in Engineering, Research and Technology [IJIERT]
ICITDCEME'15 Conference Proceedings
ISSN No - 2394-3696

DESIGN OF FPGA BASED ADAPTIVE FILTER BASED ON LMS ALGORITHM FOR ECHO CANCELLATION

Vaibhav S Kadam
Department of Electronics and Telecommunication Engineering.
SND College of Engineering & Research Center, Babhulgaon Yeola.

Prof. A.B Gaikwad
Department of Electronics and Telecommunication Engineering.
SND College of Engineering & Research Center, Babhulgaon Yeola.

ABSTRACT
Traditional telephony uses only a single receiver for speech acquisition. If the speaker is standing away from the telephone, the signal will be weak and there will be interference sources from room reverberation. In addition, there is acoustic echo coming from the loudspeaker, which further interferes with the signal of interest. This research investigated the combination of common solutions to these problems. Echo Cancellation is an occurrence in today's communication systems. The interferences caused by acoustic echo are distracting to the users and reduce the quality of the voice. Echo cancellation removed the echo coming from the loudspeaker. The main component of the system is the adaptive filter.
Filtering data in real-time requires dedicated hardware to meet demanding time requirements. If the statistics of the signals are not known, then adaptive filtering algorithms can be implemented to estimate the signals statistics iteratively. Modern field programmable gate arrays (FPGAs) include the resources needed to design efficient filtering structures. The present work deals with the designing and real-time implementation of adaptive echo cancellation on FPGA. Here LMS algorithm, which is one of the most popular algorithms to adjust the filter coefficient of an adaptive filter, is used. A model based design approach for the adaptive filter scheme is developed by using MATLAB, SIMULINK and SYSTEM GENERATOR which provides a virtual FPGA platform. The Filter is implemented using ALTERA Cyclone II FPGA board and the complete design cycle of Verilog Modeling, Coding, Simulation, Synthesis, Implementation; Testing on FPGA Target system is studied and practiced through the adaptive filter implementation. Simulation results of the conventional approach and model based design approach are compared for verification purpose.

KEYWORDS - FPGA, adaptive filter, LMS, echo cancellation, AEC, altera cyclone-II, DSP, system generator

INTRODUCTION
Acoustic echo cancellation (AEC) is an occurrence in today communication systems. The interferences caused by acoustic echo are distracting the users and reduce the quality of the voice. In order to attenuate the effects created by the echo, a Least Mean Square algorithm is used.AEC is one of the most popular applications for adaptive filters. AEC devices are needed for removing the echoes resulting from the acoustic coupling between the loudspeaker(s) and the microphone(s) in communication systems. The main component of the system is the adaptive filter, which generates at its output a replica of the echo that is further subtracted from the microphone signal. One of the most used algorithms for AEC is the Least Mean Square (LMS) algorithm, due to its simplicity and low computational complexity. The update for the coefficient requires a small number of multiplications and additions, which makes it suitable for programmable gate array design.
Employing adaptive filters overcomes the difficulties and drawbacks associated with the using of a conventional digital signal processor, or digital filters for processing data with inadequate statistics or when the statistical characteristics of the input data are known to change with time. The "adaptive filter" can be defined as a system which is trying to adjust its parameters so as to respond to some criteria with the aim of meeting some well-defined goal or target which depends upon the state of the system as well as its surroundings. Adaptive filter play an important role according to the following properties: first, it can work effectively in unknown environment; second, it is used to track the input signal of time-varying characteristics.

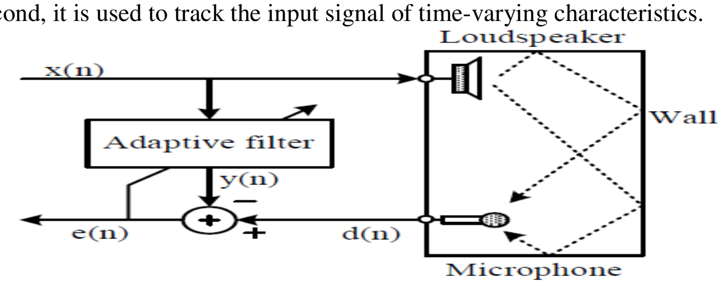


Figure 1: Basic schematic diagram of an AEC

IJIERT-ICITDCEME15

The problem of acoustic echoes in telecommunication systems arises when a loudspeaker and microphone are positioned such that the microphone picks up the acoustic waves emitted from the loudspeaker, and there is an electrical path from the microphone back to the loudspeaker. Depending on the turnaround time of the system, this is perceived as reverberation, or for longer delays, as an annoying echo. The situation appears in telecommunication systems and sound amplification applications if not proper steps have been taken to prevent it. A related problem is cancellation of line echoes in the Public Switched Telephone Network (PSTN). Line echoes appear in long telephone wires, due to long wave propagation time and impedance mismatch. The line echo problem is easier to solve, since wires are stationary, and the echo path changes little over time, as opposed to the acoustic counterpart.

BACKGROUND ESSENTIALS

In order to understand the content presented in this paper it is first necessary to provide some background information of digital signal theory.

a. ADAPTIVE FILTER

The basic concept of an adaptive filter is shown in Fig.2. The objective is to filter the input signal, x(n), with an adaptive filter in such a manner that it matches the desired signal, d(n). The desired signal, d(n), is subtracted from the filtered signal, y(n), to generate an error signal. The error signal drives an adaptive algorithm which generates the filter coefficients in a manner which minimizes the error signal. The least-mean-square (LMS) or recursive-least-squares (RLS) algorithms are two of the most popular. Adaptive filters are widely used in communications to perform such functions as equalization, echo cancellation, noise cancellation, and speech compression.

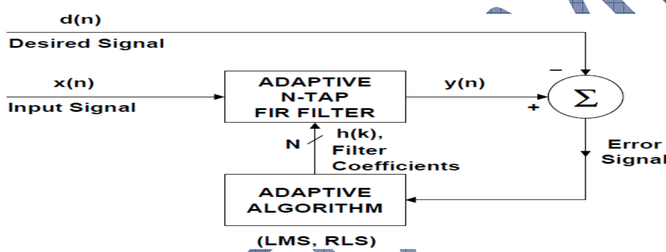


Figure 2: Adaptive Filter

b. LMS ALGORITHM

LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms LMS algorithm is relatively simple; it does not require correlation function calculation nor does it require matrix inversions.

c. LMS ALGORITHM STEPS

The LMS is well suited for a number of applications, including adaptive echo and noise cancellation, equalization, and prediction. Other variants of the LMS algorithm have been employed, such as the sign-error LMS, the sign-data LMS, and the sign-sign LMS.

1. For the sign-error LMS algorithm,

$$w_k(n+1) = w_k(n) + \beta\ \mathrm{sgn}[e(n)]x(n-k)$$

Where, sgn is the signum function,

$$\mathrm{sgn}(u) = \begin{cases} 1 & \text{if } u \geqslant 0 \\ -1 & \text{if } u < 0 \end{cases}$$

2. For the sign-data LMS algorithm,

$$w_k(n+1) = w_k(n) + \beta e(n)\ \mathrm{sgn}[x(n-k)]$$

3. For the sign-sign LMS algorithm,

$$w_k(n+1) = w_k(n) + \beta\ \mathrm{sgn}[e(n)]\ \mathrm{sgn}[x(n-k)]$$

Which reduces to

IJERT-ICITDCEME15

$$w_k(n+1) = \begin{cases} w_k(n)+\beta & \text{if } \operatorname{sgn}[e(n)] = \operatorname{sgn}[x(n-k)] \\ w_k(n)-\beta & \text{otherwise} \end{cases}$$

This is more concise from a mathematical view point because no multiplication operation is required for this algorithm.
The LMS algorithm has been quite useful in adaptive equalizers, telephone cancellers and so forth

ECHO CANCELLER

Echo canceller algorithms are based on estimation of the acoustic path impulse response, and cancellation (removal) of the loudspeaker signal from the microphone signal. There are two requirements for the estimation of the acoustic path to work. First, it is essential that there is a signal emitted from the loudspeaker. Second, estimation cannot be performed while there is sound, such as a near end speaker signal, added in the acoustic path. When the near end Talker is speaking, estimation must be turned off. Assuming the far end talker is speaking, which can be easily detected by measuring the power of the loudspeaker signal, there are two cases. The first is that the near end talker is silent and there is no explicit noise within the enclosure. The second condition is the opposite, that is, there is sound that cannot be derived from the loudspeaker generated within the enclosure. Under such a condition as the latter, popularly denoted double-talk", it is not possible for the estimation algorithm to work properly. A double-talk detector has to be used to find these events. The design of double-talk detectors is not within the scope of this thesis.
Echo canceller systems can be built based on estimators and adaptive filters, theory of signal estimation and adaptive filters is presented. Several configurations are possible concerning both estimation and cancellation. The design space is vast and performance in terms of convergence rate and maximum echo suppression, as well as design cost in terms of silicon area or power consumption, depends strongly on how the algorithm is constructed. A number of estimation algorithms are theoretically applicable for acoustic echo cancellation purposes. However, in most cases, a simple and robust algorithm outperforms more sophisticated solution. Although the more sophisticated solution might have better performance in theory, the simple and robust algorithm is easier to get to work in an implementation. Furthermore, the simpler algorithm is easier to analyse, and is associated with a lower implementation cost. Therefore, the Least Mean Squares (LMS) algorithm has been selected as the estimation algorithm of choice throughout this work. Performance of the LMS is sufficient for a reasonable implementation cost while stability and convergence can be guaranteed.

DESIGN APPROCH

An ideal approach for embedded system design is model based design approach. There are many system model tools available from various vendors which facilitate the design of hardware & software before actual physical implementation of the system. MAT lab and SIMULINK tool is used for this paper.

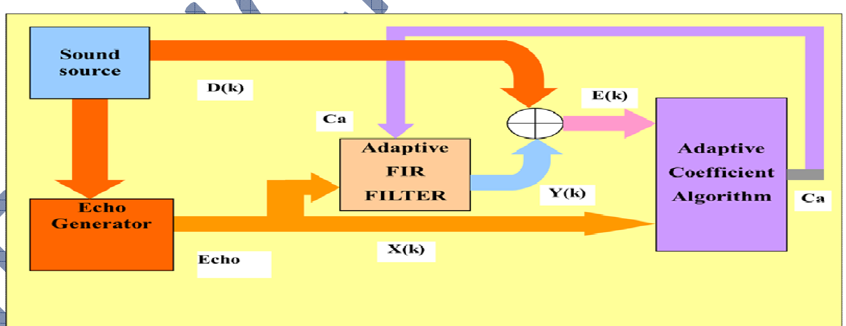


Figure 3: Top level block diagram for adaptive echo canceller.

The top level block diagram shown in "fig.3." comprises of a sound source module that is any audio data (input) and the echo generation module is any environment where echo is generated e.g. can be a vacant room. Here the adaptive FIR filter output Y(K) is compared with the desired signal D(K) to yield an error signal E(K), which is fed back to the adaptive LMS filter. The error signal is input to the adaptive algorithm which generates the filter coefficients in a manner which minimizes the error signal. These adaptive coefficients (Ca) are again used by the adaptive FIR filter to minimize the echoed signal.

a. MODEL BASED DESIGNING USING SYSTEM GENERATOR

System Generator is a system-level modelling tool that facilitates FPGA hardware design. It extends Simulink in many ways to provide a modelling environment that is well suited to hardware design. The tool provides high-level abstractions that are automatically compiled into an FPGA at the push of a button. The tool also provides access to underlying FPGA resources through low-level abstractions, allowing the construction of highly efficient FPGA designs. It also provides code generation facility.

ISSN No - 2394-3696

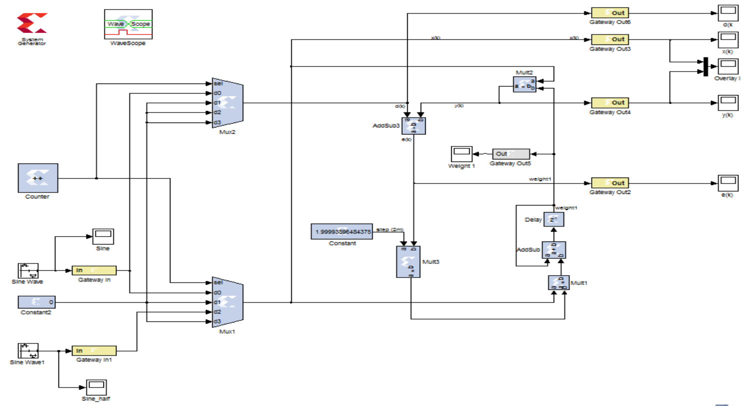


Figure 4 : System Generator Model for single order adaptive Echo Canceller for Internal Signal Pattern

In the system generator model of adaptive echo cancellation design. There are two main stages of this model, they are echo generation module and echo cancellation. Here the mux1 is used to generate the echo by adding the sine signal and the sine half signal. Single order LMS algorithm module is attached to the echo generation module where mux2 is used to cancel out the echoed signal and display the desired signal. The step size of the LMS algorithm is variable and can be adjusted to minimize the error e(k) and to get more accurate output y(k). The System Generator Wave Scope block used in the model provides a powerful and easy-to-use waveform viewer for analysing and debugging System Generator designs. Here it allows observing the time-changing values of single order LMS after the conclusion of the simulation. The signals are viewed in analog format.

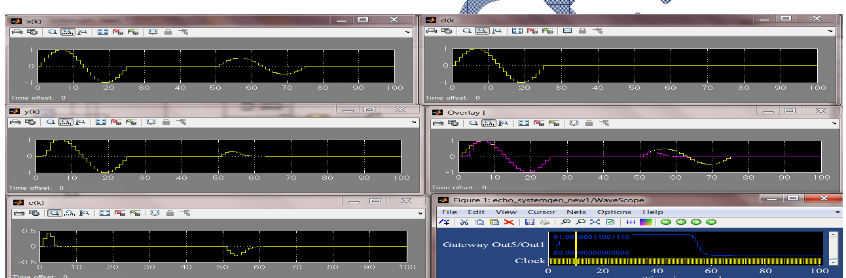


Figure 5: Observation of System generator model for echo cancellation

REAL TIME EXTERNAL SIGNAL ECHO CANCELLATION (HUMAN VOICE)

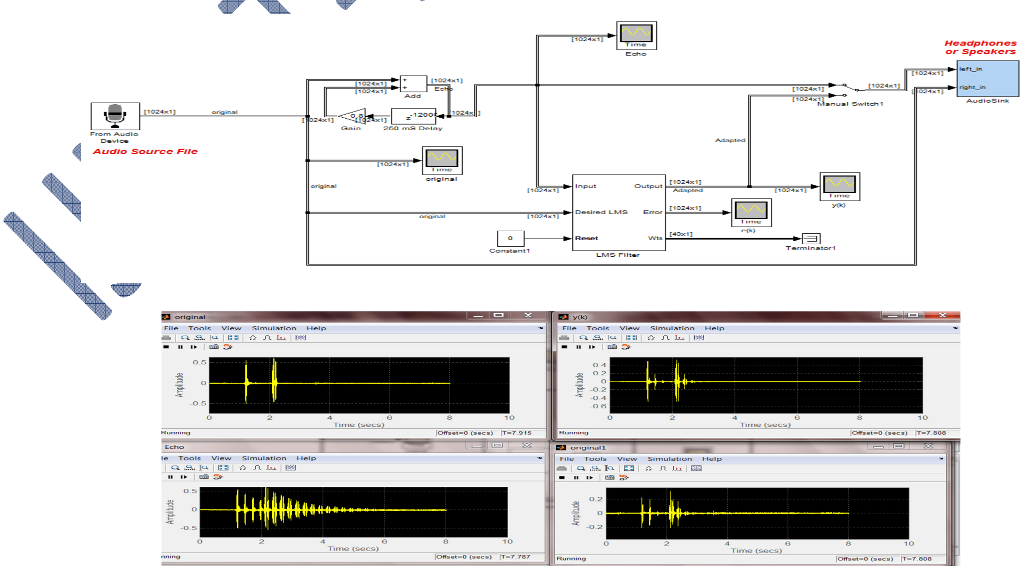


Figure 6: Model & Scope output of Real Time External echo cancellation.

TOP LEVEL BLOCK DIAGRAM OF INTERNAL WORKING OF FPGA

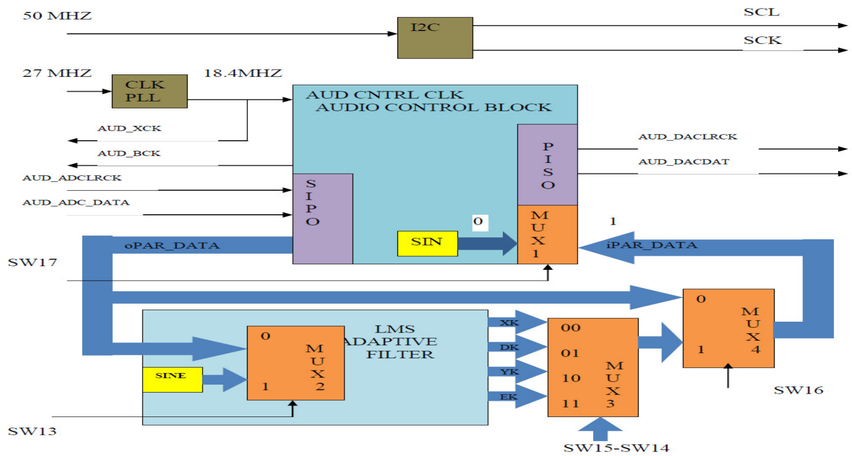


Figure 7: Single Channel test Setup of FPGA for internal signal

The FPGA implementation section shown in "fig.7" comprises of central cyclone-II FPGA interfaced with oscilloscope. Here the FPGA uses 27 MHz clock input and 18.432 MHz clock output. Inside the ROM of FPGA a sine lookup table is there which is required for generating internal sine wave. This sine look up table is given as input to the MUX through LMS adaptive filter. Inside FPGA block MUX are designed using Verilog program and adapted output is synchronized to sampling clock (48 KHz).The output is finally displayed on oscilloscope.
The Verilog program is designed in such a way that when switch13 is ON ("1") then internal sine wave is generated. Here switch15 and switch14 are used to display different adaptive signal output on oscilloscope. When both switch 15 and switch 14 are in OFF ("0") condition then unknown signal X(k) is displayed. When switch15 is OFF & switch 14 is ON then desired signal D(k) is displayed. When switch15 is ON and switch 14 is OFF then output signal Y(K) is displayed and when both switch15 & switch14 are in ON ("1") condition then the error signal E(K) is displayed on the oscilloscope.

TESTING & OBSERVATION

When the hardware set-up is ready according to the design then different and observation are taken.

SWITCHES ASSIGNED FOR SPECIFIC TESTING ARE AS FOLLOWS:

Table I

SWITCH	OFF (0)	ON (1)
	Signal	Signal
13	External	Internal
17	Internal	External
16	Internal	Adapted

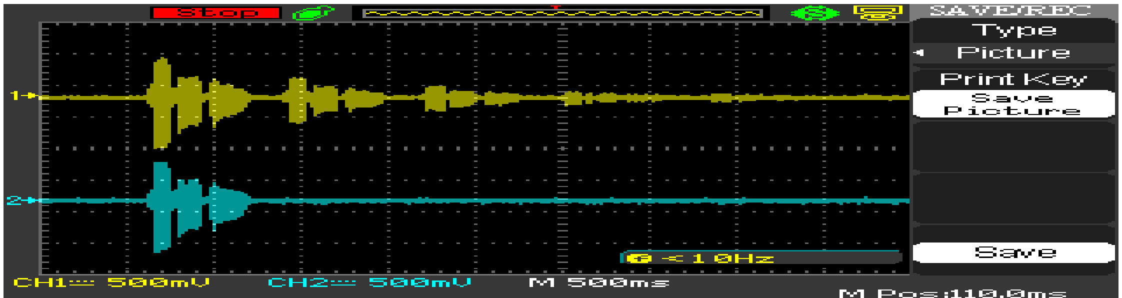
Table II

SWITCH 15	SWITCH 14	SINGLE CHANNEL	MULTI CHANNEL	
		SIGNAL	LEFT CHANNEL SIGNAL	RIGHT CHANNEL SIGNAL
0	0	X(K)	X(K)	D(K)
0	1	D(K)	Y(K)	E(K)
1	0	Y(K)	X(K)	Y(K)
1	1	E(K)	D(K)	Y(K)

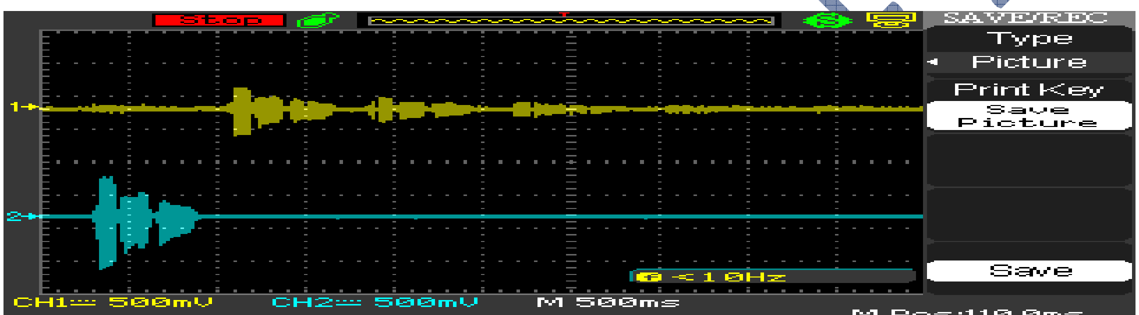
ISSN No - 2394-3696

EXTERNAL SIGNAL TESTING

sw13=0; sw17=1; Sw16=0 (LMS output), sw 15 & sw 14 '00' L=X(k), R=D(k)



sw 15 & sw 14 '01' L= Y(K) R=E(K)



sw 15 & sw 14 '10' L= X(K) R=Y(K)

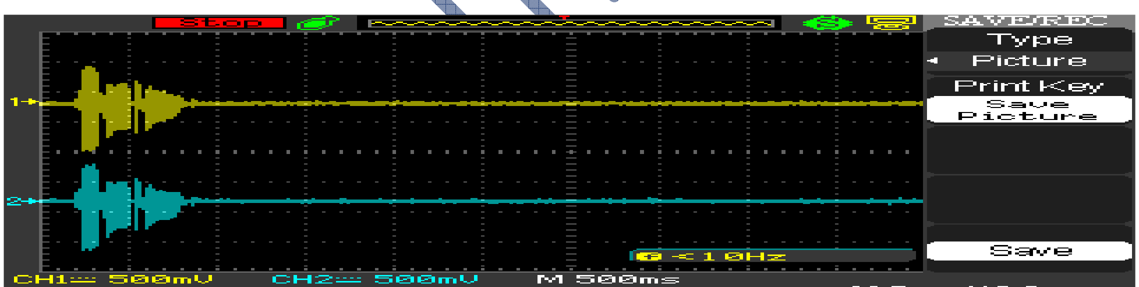


Table III. Fpga Hardware Resources Utilized

RESOURCE	USED		AVAILABLE	PERCENTAGE	
	Single Channel	Multi-Channel		Single Channel	Multi-channel
Logic elements	888	1,032	33216	3%	3%
Combinational Functions	816	919	33216	2%	3%
Dedicated Logic Registers	404	452	33216	1%	1%
Total Registers	404	452			
Total Pins	312	312	475	66%	66%
Embedded Multiplier 9-bit elements	14	14	70	20%	20%
Total PLLs	1	1	4	25%	25%

HARDWARE SETUP

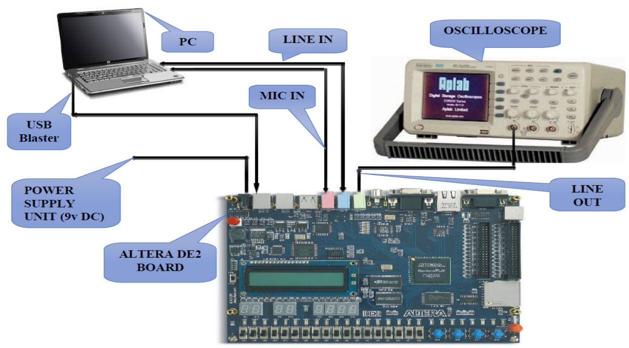


Figure 8: Hardware setup

For the hardware set up in "Fig.8," an USB blaster is connected to the laptop in which the required software is pre-installed. On successful compilation of the program a .SOF file will be generated. That files needs to be dumped on ALTERA CYCLONE-II board through this USB cable by the help of programmer option of the QUARTUS-II. A digital storage oscilloscope is taken to display the results. The use of function generator is in future scope for generating an external signal.

CONCLUSION

The Present work gives a practical approach to learning methodology for DSP applications on FPGA platform which would form the basis of future advance work in the area of adaptive algorithm for a variety applications in control, computing and communication domains.

The adaptive echo cancellation system is successfully developed with the LMS algorithm. Conventional Simulink model results and FPGA testing results are analysed and verified which are approximately matching.

FUTURE SCOPE

Real hardware environment can be created by using an actual system with real echoes. A microphone and speaker with a box in between them could be used to create the echoes. It can further be extended to Multi-Order LMS adaptive filter to enhance the capability of adaptation and different algorithms can also be used for echo cancellation. This echo canceller can be developed in so many platforms not only conventionally but also with digital controllers. As the present age scenario is giving more emphasis on digital controllers for faster and better compensation.

REFERENCES

[1] Prof. Sashank Shekhar Pujari,Prangya Paramita Muduli,Amruta Panda,Rashmita, Badhai, Sofia Nayak and Yougajyoty Sahoo; "Design & implementation of fir filters using on-board adc-dac & fpga" ;ICICES, 2014; in press.

[2] nithin v.george n, ganapatipanda;"advances in active noise control: a survey with emphasis on recent nonlinear techniques",Pages- 363-377,elsevier 2012.

[3] Dr. Raaed Faleh Hassan,AliSubhiAbbood;"Design and implementation of fpga based adaptive filter";IJES ,Volume2, Issue 6 ,Pages74-80, 2013.

[4] j. malarmannan et.al;"fpga implementation of adaptive filter and its performance analysis"; international journal of engineering and technology (ijet), 2013

[5] kafaicedricyiu , yaolua, chunhok ho, wayneluk , jiaquanhuo , svennordholm; "reconfigurable fpga-based switching path frequency-domain echo canceller with applications to voice control device", pg-376-390, elsevier 2012.

[6] radhikachinaboina, d.s.ramkiran, habibulla khan, m.usha, b.t.p.madhav, k.phanisrinivas& g.v.ganesh;"adaptive algorithms for acoustic echo cancellation in speech processing" , ijrrasapril 2011; vol7 issue1.

[7] sanjay k. nagendra, vinaykumar.s.b;"echo cancellation in audio signal using lms algorithm"; national conference on recent trends in engineering & technology 2011.

[8] ioanahomana, irinamuresan, marina topa, cristiancontan;"fpga implementation of an acoustic echo canceller", volume 52, number 2, 2011actatechnicanapocensiselectronics and telecommunications.

[9] Rafid Ahmed Khalil; "Adaptive Filter Application in Echo Cancellation System and Implementation using FPGA"; Al-Rafidain Engineering ,Vol.16 No.5 Dec. 2008.

[10] S. S. Godbole1, P. M. Palsodkar2 and V.P. Raut; "FPGA Implementation of Adaptive LMS Filter"; Proceedings of SPIT-IEEE Colloquium and International Conference, Mumbai, India; Vol. 2,2 2 6.

[11] michaelhutson;"acoustic echo cancellation using digital signal processing"; 2003,online

IJIERT-ICITDCEME'15

[12] wee chong chew and dr. b. farhang-boroujeny;"fpga implementation of acoustic echo cancelling"; 1999 ieeetencon
[13] Diggikar, A.B. ; Ardhapurkar, S.S"Design and implementation of adaptive filtering algorithm for noise cancellation in speech signal on fpga"; ICCEET, 2012.
[14] Simon Haykin, Adaptive Filter Theory, Prentice-Hall, Inc.,1996.
[15] Windrow, B., and Stearns, S. D., Adaptive Signal Processing, Prentice Hall, 1985.
[16] Math Work Help Documentation.
[17] DE2 User manual.
[18] Audio Codec W8731 datasheet.

IJIERT-ICITDCEME'15