



Design and implemantation of LMS adaptive filter for acoustic echo cancellation system on FPGA using HDL coder

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Abstract: In telecommunication acoustic echo which is distorted and delayed version of original sound signal is a major problem. Adaptive filtering is an efficient technique to remove this echo from the received signal. Adaptive filters estimated the filter coefficient of unknown environment system which changes its filter coefficient dynamically. This paper presents the design of least mean square (LMS) adaptive filter to remove acoustic echo form wireless system and implementation of filter on Defense- Grade Vertex -6Q FPGA kit. The design of filter is done in MATLAB and by using HDL coder MATLAB design is converted to VHDL design. The design is evaluated in terms of SNR, filter order, FPGA resource utilization.

Keywords: LMS,, MATLAB, HDL coder, VHDL, FPGA, SNR

I. Introduction

Acoustic echo is a major problem in wireless systems. Acoustic echo is occur due to reflection of sound wave in environment resulting which the original signal has superposition of reflected attenuated time delayed signal which effect the sound quality of the original signal. To reduce the acoustic echo adaptive filters are used. LMS adaptive filters are prefer in acoustic echo cancellation system due to its simplicity, low computational cost and robustness[1].

Adaptive filter learn the statistics of their operating environment and change their filter coefficient accordingly. Adaptive filter changes its coefficient according to the step size and error signal which is difference of desired signal and output of the filter.

HDL coder is a tool available in MATLAB which convert MATLAB code to VHDL code. To implement the design VHDL design is loaded on field programmable gate array (FPGA) kit. FPGA is a flexible in terms of reprogram the same hardware in number of times and also good performance by enabling the parallel processing[2].

II. ECHO CANCELLATION SYSTEM

Acoustic echo is known as reflection of sound from surrounding environment. To remove the echo, an echo cancellation system is formed which is comprises of adaptive filter and that echo cancellation system is known as Adaptive Echo Cancellation system[3].

Let us assume the voice of speaker A as a near end signal $x(t)$, voice of speaker B as a far end signal $y(t)$ and the response of surrounding environment is $h(t)$. The speaker B hear the voice of near end signal but along with the voice of near end signal he hears his own voice with some delay and that reflection of signal is called acoustic echo, and we have a new reflective signal $v(t)$ which is convolution of $y(t)$ and $h(t)$.

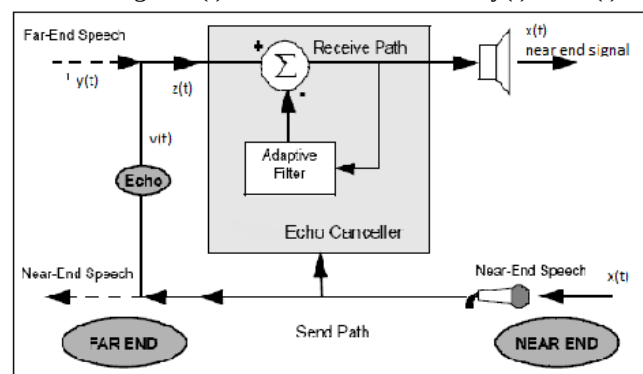


Figure 1: Echo Cancellation System

$$v(t) = h(t) * y(t) \quad (1)$$

then the received signal $z(t)$ is :

$$z(t) = v(t) + x(t) \quad (2)$$

Now adaptive filter changes its filter coefficients according to the surrounding environment and its impulse response is given by $h'(t)$. So the estimated reflected signal $v'(t)$ is

$$v'(t) = h'(t) * y(t) \quad (3)$$

So the error signal is

$$e(t) = v(t) + x(t) - v'(t) \quad (4)$$

$$\text{If } v(t) = v'(t)$$

$$\Rightarrow e(t) = x(t) \quad (5)$$

$\Rightarrow$ So the impulse response of surrounding channel and adaptive filter is same

$$\Rightarrow h(t) = h'(t) \quad (6)$$

Above equation shows that the predicted output is equal to the near end signal and now it is cleared that echo is removed from the system.

III. Adaptive Filter

In signal processing and communication the time varying noise from one point to another point is a major problem that problem occurs in a medium which is known as channel where noise and distortion comes into the picture according to the time varying[3,4] parameter of channel. This variation can be slow or fast. To remove this problem from the channel, adaptive filters are used because they are self predicted filters or they can change their properties according to the given input.

IV. LMS Algorithm

LMS algorithm is based on the Widrow and Hoff equation[4] in which a simplification of gradient vector computation is made possible by appropriately modifying the objective function. LMS algorithm is the most widely used algorithm of adaptive filters in the field of speech processing.

These types of filters are used to utilize the gradient vector of filter tap weights to converge the optimum solution of the wiener equation[4,5]. The weights of the filters are defined as:

$$w(n+1) = w(n) + 2\mu e(n)x(n) \quad (7)$$

$$e(n) = d(n) - y(n) \quad (8)$$

$$y(n) = w^T(n) * x(n) \quad (9)$$

where μ is called step size, $e(n)$ is error signal and $x(n)$ is the delayed input signal, $w(n)$ is known as the tap weights of the adaptive filter and $y(n)$ is output of adaptive filter. Step size is an important parameter in above equation which defines the convergence and stability of the algorithm[6].

V. MATLAB Simulation

MATLAB simulation of Echo Cancellation system for LMS algorithm is as follows:

The room impulse filter has 8 coefficients so the input near and far end signals must have 8 or more than 8 Samples. So the simulation is take place for 9 samples [6]. Because the sampling rate of wave is 8000 samples per second so 9 samples are able to tell behavior of wave in millisecond.

The MATLAB simulations are:

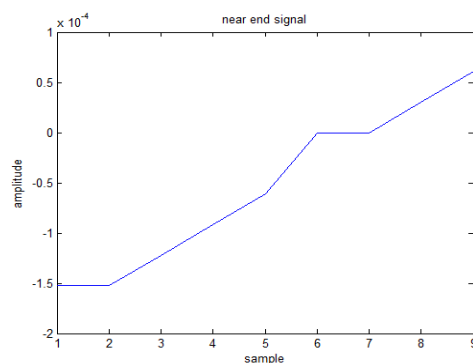


Figure 2: near end signal

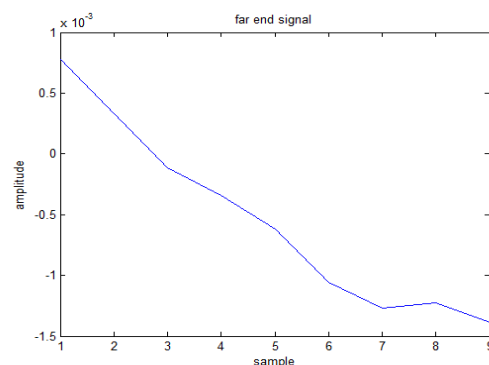


Figure 3: far end signal

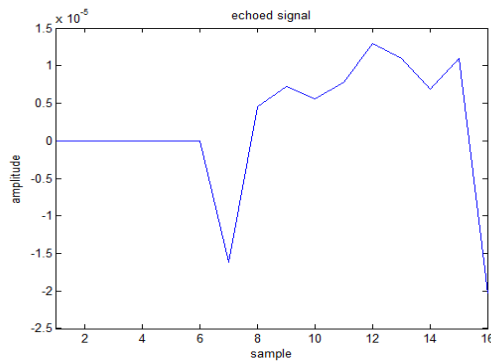


Figure 4: echo signal

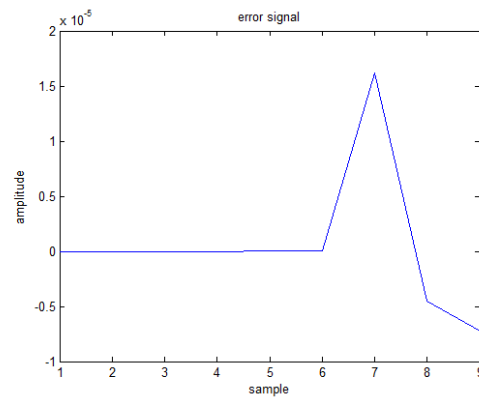


Figure 6: error signal

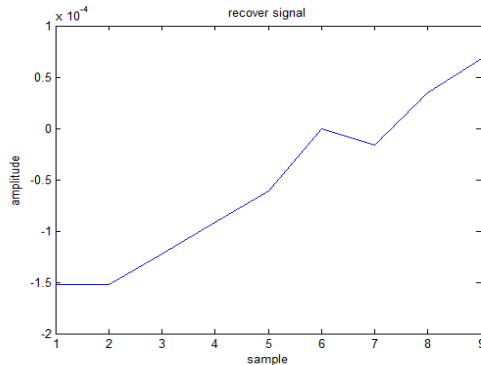


Figure 5: recover signal

VI. Observation

For different filter order and different no of sample observation is done and SNR and convergence rate is calculated, for step size .001 results are:

Filter length	Samples	SNR[dB]	Convergence (millisecond)
8	9	13.70	0.976
	16	34.02	1.134
	32	46.31	2.356
16	16	14.75	1.147
	32	43.57	2.578
	64	49.24	3.924

TABLE 1

VII. FPGA IMPLEMENTATION

FPGA contain an array of programmable logic blocks, and a hierarchy of reconfigurable interconnects that allow the blocks to be “wired together”. Logic blocks can be configured to perform complex combinational function, or simple logic gates. The FPGA configuration is generally specified using a hardware description language (HDL).

In acoustic echo cancellation system the hardware to the system has to be implemented and for that implementation Defense-Grade Vertex-6Q FPGA kit is used. For FPGA implementation the code should be in HDL but our source code is MATLAB code. This problem is resolves using HDL coder of MATLAB.

HDL CODER : HDL coder is a code converter present is MATLAB which is able to convert source code of MATLAB in Front end designing language or it can change .m file(MATLAB file) into .vhd (VHDL) or .v(Verilog) file[7]. In this paper MATLAB code is converted into VHDL code for the hardware implementation of algorithm. HDL coder changes floating point design of MATLAB to fixed point design for the hardware implementation of code in FPGA[8,9].

HDL coder also tells the source Utilization of the converted code and the result of source utilization report is:

Multipliers	353
Adders/Subtractors	416
Registers	0
RAMs	0
Multiplexers	0

Figure 7: resource report

The converted VHDL code[10] runs on Defense-grade vertex-6Q FPGA and target device is xq6v5x475t-1rf1759 used.

The outputs and synthesized report are as follows:

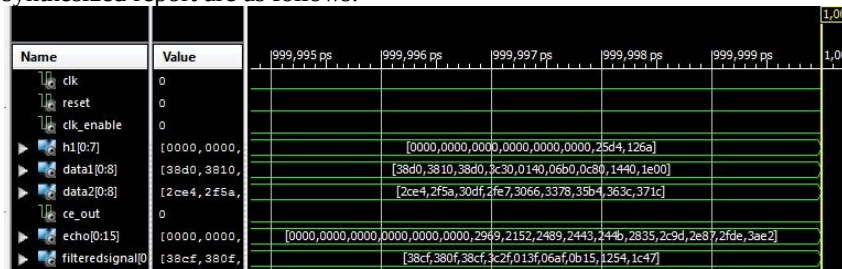


Figure 8: simulation result

lmsfunction_fixPt Project Status (05/02/2015 - 15:16:41)			
Project File:	sixteen_bit_cancellation.xise	Parser Errors:	No Errors
Module Name:	lmsfunction_fixPt	Implementation State:	Synthesized
Target Device:	xq6v5x475t-1rf1759	Errors:	No Errors
Product Version:	ISE 12.2	Warnings:	No Warnings
Design Goal:	Balanced	Routing Results:	
Design Strategy:	Xilinx Default (unlocked)	Timing Constraints:	
Environment:	System Settings	Final Timing Score:	

Device Utilization Summary (estimated values)			
Logic Utilization	Used	Available	Utilization
Number of Slice LUTs	2017	297600	0%
Number of fully used LUT-FF pairs	0	2017	0%
Number of bonded IOBs	716	840	85%
Number of DSP48Es	333	2016	16%

Figure 9: XILINIX synthesis report

VIII. Conclusion

In this paper design and implementation of acoustic echo cancellation system using LMS algorithm is done successfully. Acoustic echo cancellation system by LMS algorithm is design in MATLAB and for the different samples and filter order SNR and convergence rate of the design is checked (table 1) and conclude that as we increase the no of samples SNR is improved and convergence rate is also increases. The resource requirement for implemented LMS algorithm on Defense Grade Vertex-6Q is quite comprehensive (fig 9) and concluded that MATLAB design is directly implemented on FPGA by using HDL coder.

Reference

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