ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 2, Issue 4, July 2013

Area and Power Efficient LMS Algorithm for Denosing Speech Signal

Sujith Chatrad, Asha K S

Abstract— ***The Least Mean Square (LMS) algorithm is well known algorithm in many digital signal processing applications. It has established its popularity in several important areas of application. One such application is eliminating noise present in the speech signal. LMS algorithm, on the other hand, offers an excellent alternative, and its best characteristic are robustness and simplicity. Its popularity stems from its relatively low computational complexity, good numerical stability, simple structure and ease of implementation in terms of hardware. In this paper a adaptive filtering technique is used for noise cancellation in speech signal. LMS algorithm is implemented for adaption of the filter coefficients. The noise cancellation system is designed in MATLAB Simulink tool box and the simulation of MATLAB design is implemented and verified using XILINX 14.1 tool by model sim.***

Index Terms— **Least Mean Square Algorithm (LMS), Adaptive filter, Adaptive noise cancellation.**

I. INTRODUCTION

In all practical situations, the received speech waveform contains some form of noise component. The noise may be a result of addition of acoustically coupled background noise. Depending on the amount and type of noise, the quality of the received waveform can range from being slightly degraded to being annoying to listen to, and finally to being totally unintelligible. The most common problem in speech processing is the effect of interference noise in speech signals. Interference noise masks the speech signal and reduces its intelligibility. Interference noise can come from acoustical sources such as ventilation equipment, traffic, crowds and commonly, reverberation and echoes. It can also arise electronically from thermal noise, tape hiss or distortion products. If the sound system has unusually large peaks in its frequency response, the speech signal can even end up making itself. The problem of removing the unwanted noise component from a received signal has been the subject of numerous investigations. The pioneering work of Wiener and others give an optimum approach for deriving a filter that tends to suppress the noise while leaving the desired signal relatively unchanged [1] -[3]. The design of these filters requires that the signal and the noise be stationary and that the statistics of both signals be known a priori. In practice, these conditions are rarely met.

In this paper, we present a technique for optimum noise filtering for speech signals based upon the principles of least mean square (LMS) adaptive filtering [4]. This technique has the advantage of requiring no a prior knowledge of the detailed properties of the noise signal. The technique takes the advantage of the quasi-periodic nature of the speech, and preliminary results indicate that the technique improves the perceived speech quality of a signal corrupted by additive white noise. LMS is a hardware efficient algorithm because less multiplier is present in LMS, to save gate required implementing on FPGA. If multiplier is present, then cost and number of gates increases. The LMS algorithm has become a widely used approach to elementary function evaluation where the silicon area is a primary constraint.

II. PROPOSED SYSTEM

In this paper, we propose low computational complexity LMS algorithm for area-power efficient implementation of LMS. The proposed ANC system has comparable or less area complexity with other existing system LMS algorithms.

The proposed ANC design uses two sensors are based on the following key ideas:

1) The reference sensor measures the primary noise to be cancelled.
2) The error sensor monitors the performance of the ANC system.

The adaptive feedback Adaptive Noise Canceller (ANC) system uses only an error sensor and cancels only the predictable noise components of the primary noise. The block diagram for the proposed ANC system is shown in Fig 1. As shown in the Figure 1 below, an Adaptive Noise Canceller (ANC) has two inputs – primary and reference.

The primary input receives a signal s from the signal source that is corrupted by the presence of noise n uncorrelated with the signal. The reference input receives a noise n0 uncorrelated with the signal but correlated in some way with the noise n. The noise n0 passes through a filter to produce an output nˆ that is a close estimate of primary input noise n. This noise estimate is subtracted from the corrupted signal to produce an estimate of the signal at sˆ, the ANC system output. In noise canceling systems a practical objective is to produce a system output sˆ = s + n – nˆ that is a best fit in the least squares sense to the signal s. This objective is accomplished by feeding the system output back to the adaptive filter and adjusting the filter through an LMS adaptive algorithm to minimize total system output power. In other words the system output serves as the error signal for the adaptive process.

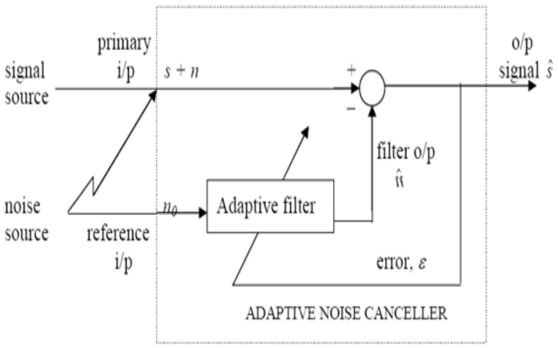

Fig 1: The Proposed Adaptive Noise Canceller System

Advantages:-

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

Applications:-

- This technique has recently gained much attention as a method to eliminate noise contained in useful signals.
- This technique has been applied in various communication and industrial appliances such as hands free phones, machineries and transformers.
- This technique has been implemented in biomedical signal and image processing, echo cancellation and speech enhancement.

III. LMS ALGORITHM

The best characteristics of LMS algorithm are robustness and simplicity. It has a simple structure and ease of implementation in terms of hardware [7]. The main advantage of LMS is that it does not require prior knowledge of signal statistics. The weight obtained by LMS is only estimation but it improves gradually with time as weights are adjusted and the filter learns the signal characteristics. LMS is simple, low computational complexity and has fast convergence rate.Consider the transversal filter with input x (n) i.e. vector of the M (filter length) most recent input samples at sampling point n.

x(n)=[x(n),x(n-1),....x(n -M+1)] ………...…………………………………………………………………….(1)

and W (n) i.e. weight coefficients as

W(n)=[W0(n),W1(n),Wm-1(n)] ………………………………………………………….……………..(2)

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 2, Issue 4, July 2013

At some discrete time n, the filter produces an output y (n) which is linear convolution sum given by

$$y(n) = \sum_{k=O}^{M-J} W(n)\, x(n-k)$$

Also can be represent in vector form as

y(n)=W^T(n)x(n) or x^T(n)W(n)……………………………………………………………………….………(3)

The error signal is difference of this output with the primary signal d (n) given by,

e(n)=d(n)y(n)……..………………………………………………………………………………………..(4)
.
To optimize the filter design, we choose to minimize the mean-square value of e (n). Thus the cost function is defined as the MSE denoted by J

J = E [e (n) e* (n)]
=E [|e (n)|^2] ………………………………………………………………………………………….....(5)

where E denotes the statistical expectation operator.

Applying the operator Δ to the cost function J, a gradient vector ΔJ obtain as

ΔJ (n) = P - RW (n
=x (n) d (n)-x (n) x^T (n) W (n) ……………………………………………………………………....(6)

where R is the autocorrelation matrix of x (n), and P is the cross correlation matrix of d (n) and x (n).
The LMS algorithm is based on steepest-descent method. To formulate steepest-descent method. Consider a cost function J (w) i.e. a continuously differentiable function of some unknown weight vector w.
To find an optimal solution W0 (initial guess) that satisfies the condition
J(W0)<=J(W)for all W……………………………………………………………………………...……..(7)
.
which is a mathematical statement of unconstrained optimization.

IV. SIMULATION AND RESULTS

The design for proposed ANC system is designed in MATLAB Simulink tool box and the Xilinx 14.1 development environment was used for implementation of MATLAB design. The design has been transferred to Verilog code and it's the hardware simulation done with the Xilinx ISE simulator and also implemented on Spartan-3 FPGA XC3s400pq208-5 board. Area and timing reports are given for particular target device. The power dissipation of the proposed system for different clock frequencies is estimated by Xilinx XPower tool. Table 1 shows the device utilization summary of ANC system using LMS algorithm.

Table I: Hardware Device Utilization Summary

S. NO	Logic utilization	Used	Available	Utilization
1	Number of slice Flip Flops	114	7168	1%
2	Number of slice	201	3584	5%

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 2, Issue 4, July 2013

3	Number of 4 input LUTs	335	7168	4%
4	Number of bonded IOBs	49	141	34%
5	Number of MULT18X18s	10	16	62%

The Figure2 and 3 shows the original and recovered signal.

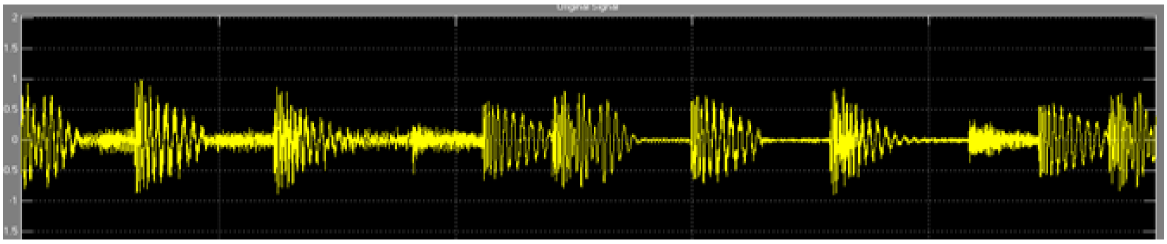

Fig2:- Original Signal

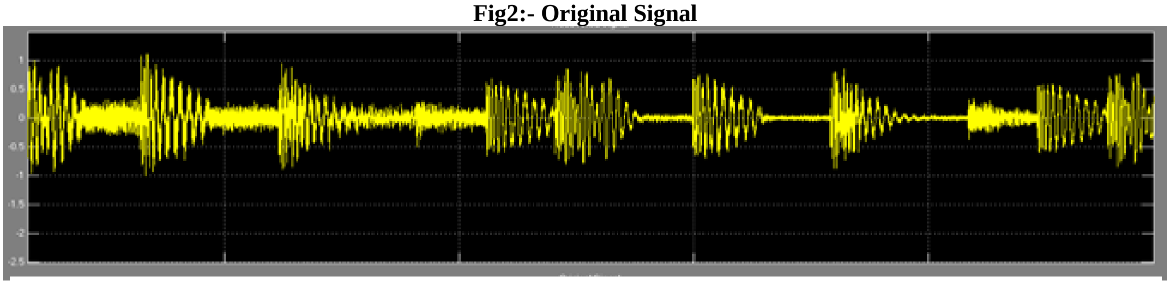

Fig3:- Recovered Signal

The Figure4 shows the error signal.

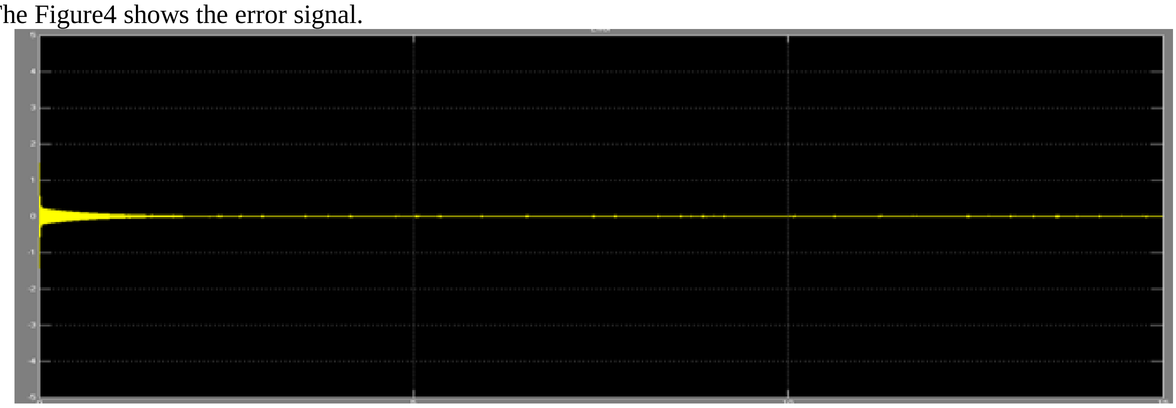

Fig4:- Error Signal

The RTL view of Proposed ANC system is shown in Fig5

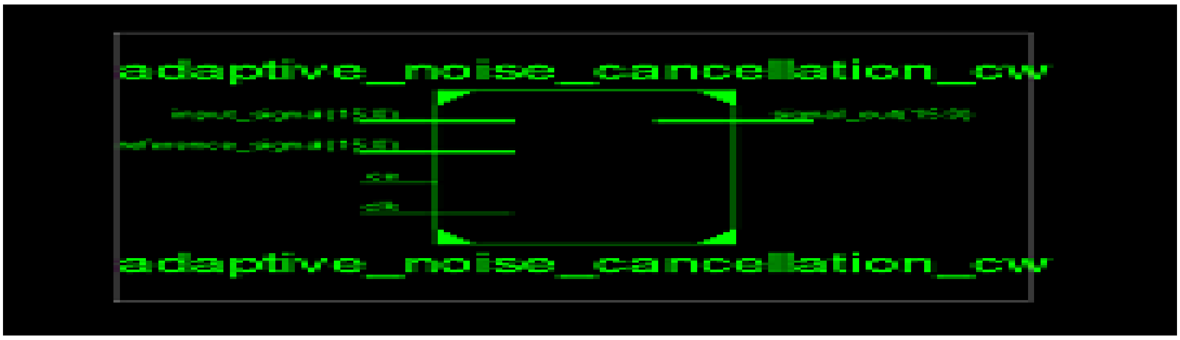

Fig5: RTL view of Proposed ANC System

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 2, Issue 4, July 2013

Timing reports include total time delay for output to appear after giving input. At speed grade of -12, design operates at maximum frequency of 137MH$_z$.The minimum period require is 7ns.The power dissipation of the proposed architecture for different clock frequencies is estimated by Xilinx XPower tool 0.157watts. And the simulation results showed in below Fig6.

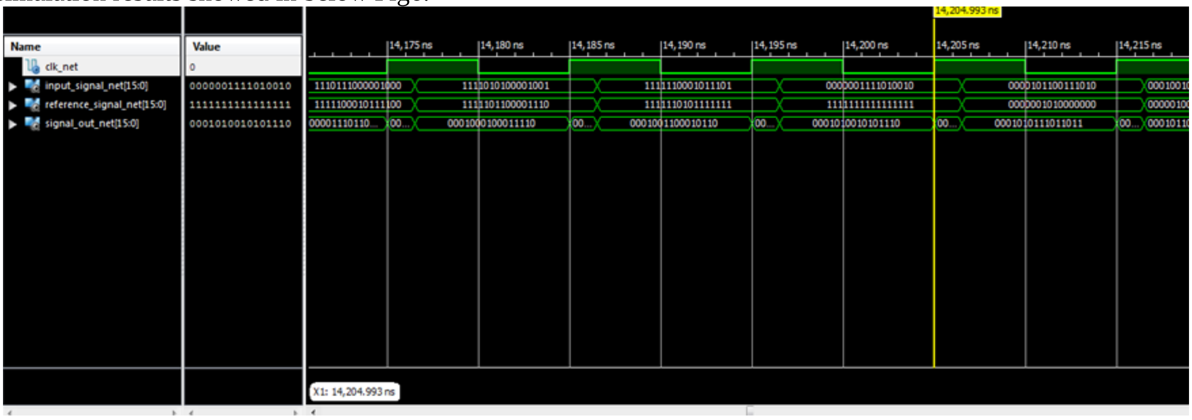

Fig6: Simulated result of proposed ANC system

V.CONCLUSION

The proposed ANC System provide a noise free from a speech signal using LMS algorithm technique.The LMS algorithm technique is suggested to reduce the number of coefficients for low computational complexity. In this work, the adaptive noise cancellation process has successfully implemented for three coefficients using Spartan 3 to reduce the area, multipliers and also proposed ANC system has less area and power consumption of 0.157 watts on Xilinx Sparten3 device.

ACKNOWLEDGMENT

The satisfaction and euphoria that accompany the successful completion of any task would be but incomplete without mentioning the people who made it possible, whose selfless criticism arguments complements constructive feedback and inspiration crowned our efforts with success. I sincerely thank each of them for their valuable support. I express my deep sense of gratitude and indebtedness to our institution “SCHOOL OF ENGINEERING & TECHONOLGY, JAIN UNIVERSITY, which provided me the opportunity to fulfill our cherished goals. I extend my sincere thanks and regards to Dr. V. MURALIDHARA, Principal SET-JU for giving us this opportunity to carry out this work in the college. I sincerely acknowledge the encouragement and support given to us by our guide Mrs. ASHA K S, Assistant professor, Dept. of ECE, SET-JU in completion of my paper. I would most heartily like to thank the almighty, my family members and friends without whom this paper would be impossible.

REFERENCES

[1] N. Wiener, Extrapolation, Interpolation an Smoothing of Stationary Time Series, with Engineering Applications. New York: Wiley, 2006.

[2] R. Kalman, “On the general theory of control,” in Proc. 1st IFAC Congress. London: Butterworth’s, 2006.

[3] R. Kalman and R. Bucy, “New results in linear filtering and prediction theory,” Trans. ASME (ser. D, J. Basic Eng.), vol. 83, pp.95-107, Dec. 2006.

[4] J. M. Gorriz, Javier Ramirez, S. Cruces-Alvarez, Carlos G. Puntonet, Elmar W. Lang, and Deniz Erdogmus, “A Novel LMS Algorithm Applied to Adaptive Noise Cancellation” IEEE Signal Processing Letters, Vol. 16, No. 1, January 2009.

[5] V.R. Mustafa “Design and Implementation of Least Mean Square Adaptive Filter on Altera Cyclone II Field Programmable Gate Array for Active Noise Control”, IEEE Sym. on Industrial Electronics Applications (ISIEA), Kuala Lumpur, Malaysia, (2009), 479-484.

[6] H. Zheng-wei and X. Zhi-yuan, “Modification of Theoretical Fixed-Point LMS Algorithm for Implementation in Hardware” 2nd International Conference on Electrical Commerce and Security(ICECS), Vol. 2 USA, 174-178,2009.

[7] Julie E. Greenberg “Modified LMS Algorithms for Speech Processing with an Adaptive Noise Canceller” IEEE Transactions on speech and audio processing, Vol. 6, No. 4, July 2008.

ISSN: 2319-5967
ISO 9001:2008 Certified
International Journal of Engineering Science and Innovative Technology (IJESIT)
Volume 2, Issue 4, July 2013

AUTHOR BIOGRAPHY

Sujith Chatrad is a student in the Department of Electronics and Communication Engineering, School of Engineering, Jain University, Bangalore. He obtained his Bachelor degree in Electronics & Communication Engineering from S.T.J.I.T Institute of Technology, Ranebennur in 2009, Visvesvaraya Technological University, Belgaum and He is pursuing M.Tech (SP and VLSI) in Electronics and Communication Engineering, Jain University, Bangalore. His research interest includes Image Processing, VLSI, DSP, and Micro-Controller.

Asha K S is an Assistant Professor in the Department of Electronics and Communication Engineering, Sri Bhagawan Mahaveer Jain College of Engineering, Jain University, Bangalore. She obtained her Bachelor degree in Electronics and Communication Engineering from S.I.T Tumkur and Master Degree in Digital Electronics from S.S.I.T Tumkur, Karnataka, India.