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A Design Approach for Low Power Implementation of Adaptive Noise Canceller using LMS Algorithm

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***Abstract*— The basic theme of our paper is to implement a new idea of noise reduction in the real time implementation using the concepts of adaptive filters. This method uses a "Primary Input" containing corrupted signal and a "Reference Input" containing noise correlated signal in some unknown way of primary noise. The reference input is adaptively filtered and subtracted from the primary input to obtain the signal estimate. A desired signal corrupted by an additive noise can often be recovered by an adaptive noise canceller using the Least Mean Square (LMS) algorithm. This adaptive noise canceller is useful to improve the S/N ratio. This algorithm provide efficient performance with less computational complexity.**

Keywords*— *Adaptive noise cancellation (ANC), LMS Algorithm, S/N ratio, Error estimation.

I. INTRODUCTION

The removal of noise from signals is a major problem related to several areas of research in signal processing and communications. The introduction of noise between the transmitter and receiver corrupts and distorts the input signal, thus providing an inferior signal quality on the receiving end. The processes to remove this unwanted interference are common. The technique of adaptive filtering is one of the medium by which signal enhancement or noise reduction is accomplished [1]. The basic idea of an adaptive noise cancellation algorithm is to pass the corrupted signal through a filter that tends to suppress the noise while leaving the signal unchanged.

Adaptive filters are digital filters with an impulse response or transfer function that can be adjusted or changed over time to match desired system characteristics. It adapts, automatically, to changes in its input signals. An adaptive filter consists of two parts: one is a digital filter with adjustable coefficients and another is an adaptive algorithm which is used to adjust or modify the coefficients of the filter that have found widespread applications are the Least Mean Square (LMS), the Recursive Least Square (RLS), and the Kalman Filter algorithms.

IJERT

II. ADAPTIVE FILTER

A. ADAPTIVE NOISE CANCELLATION

The basic idea of an adaptive noise cancellation algorithm is to pass the noisy signal through a filter that tends to suppress the noise while leaving the signal unchanged [3]. This is an adaptive process, which means it cannot require a priori knowledge of signal or noise characteristics. In summary, to realize the adaptive noise cancellation, we use two inputs and an adaptive filter. One input is the signal corrupted by noise (Primary Input, which can be expressed as s(n) and x_1(n)).

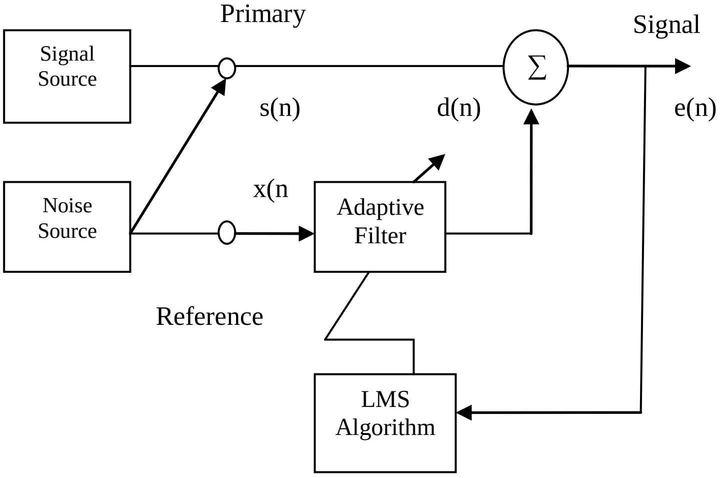

Figure 1: Adaptive Noise Cancellation

The other input contains noise related in some way to that in the main input but does not contain anything related to the signal (Noise Reference Input, expressed as x(n)) [5]. The noise reference input pass through the adaptive filter and output y(n) is produced as close a replica as possible of $x_1(n)$. The filter readjusts itself continuously to minimize the error between x_1 (n) and y (n) during this process.

Where
s (n) - Source signal
d (n) - Primary signal
$x_1(n)$ - Noise signal
x(n) - Noise Reference input
y(n) - Output of Adaptive Filter
e(n) - System Output Signal

1.Filter output:

$$y[n] = w^T[n] \bullet x[n] \qquad \text{-------} \quad (1)$$

2.Estimation error or error signal:

$$e[n] = d[n] - y[n] \qquad \text{--------} \quad (2)$$

In the above equations, the tap inputs x(n), x(n-1), ……, x(n-M+1) form the elements of the reference signal x(n), where M-1 is the number of delay elements. d(n) denotes the primary input signal, e(n) denotes the error signal and constitutes the overall system output. w(n) denotes the tap weight at the nth iteration. In the system shown in Fig.1 the reference input is processed by an adaptive filter [4]. An adaptive filter differs from a fixed filter in that it automatically adjusts its own impulse response. Thus with the proper algorithm, the filter can operate under changing conditions and can readjust itself continuously to minimize the error signal. The error signal used in an adaptive process depends on the nature of the application.

B. LMS ADAPTIVE ALGORITHM

The Least Mean Square (LMS) algorithm, introduced by Widrow and Hoff in 1959 is an adaptive algorithm. The LMS algorithm is widely used in applications to adaptive filtering due to its computational simplicity, unbiased convergence in the mean to the Wiener solution, and the existence of a proof of convergence in a stationary environment. it uses a gradient-based method of steepest decent algorithm. The LMS algorithm consist of two basic process.

1. **Filtering process:** In filtering processes, calculation of output of FIR filter by convolving input and taps and calculation of estimated error by comparing the output to desired signal.

2. **Adaptation process:** In adaptation process filter adjust tap weights based on the estimation error.

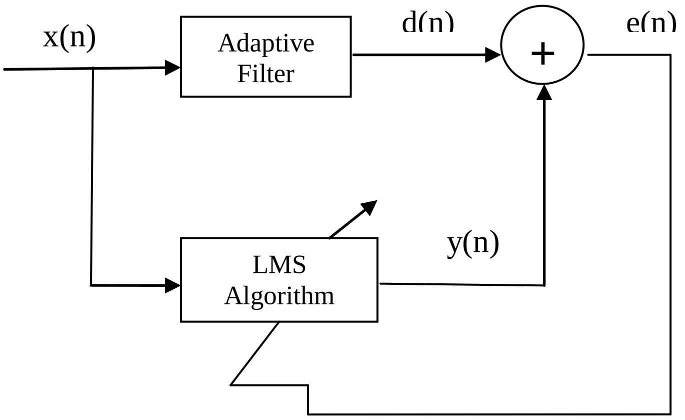

Figure 2: LMS Algorithm

The LMS algorithm approaches the minimum of a function to minimize error by taking the negative gradient of the function. The LMS algorithm is undoubtedly the most popular algorithm for adaptive signal processing [2]. The desired signal *d*(*n*) is tracked by adjusting the filter coefficients *w*(*n*). The coefficient vector updates equation for the LMS algorithm is given by

$$w[n+1] = w[n] + \mu \bullet x[n] \bullet e[n] \qquad \text{-------} \ (3)$$

Where µ is the step size of the LMS filter. The LMS algorithm is convergent in the mean square if and only ifnthe step-size parameter satisfy (0< µ < $2/l_{max)}$, l_{max} is the largest eigen value of the correlation matrix of the input data [8]. Larger values for step size Increases adaptation rate (faster adaptation) increases residual mean-squared error.

Some of the applications of the adaptive filters are as follows:

- Acoustic noise equalization
- Adaptive speech enhancement
- Channel equalization
- Adaptive line enchanter
- Noise cancellation.
- Signal & Channel prediction.

III. SIMULATION AND RESULTS

In this section we evaluate the performance of LMS algorithms in noise cancellation. Input signal is speech signal whereas Gaussian noise was used as noise signal. The LMS adaptive filter uses the reference signal and the desired signal, to automatically match the filter response [7]. As it converges to the correct filter model, the filtered noise is subtracted and the error signal should contain only the original signal.

IV. MATLAB SIMULATION

The simulation is done in real time, in MATLAB Simulink using the model shown in figure. The simulation model is tested in real-time environment, by acquiring the corrupted signal and the noise signal.

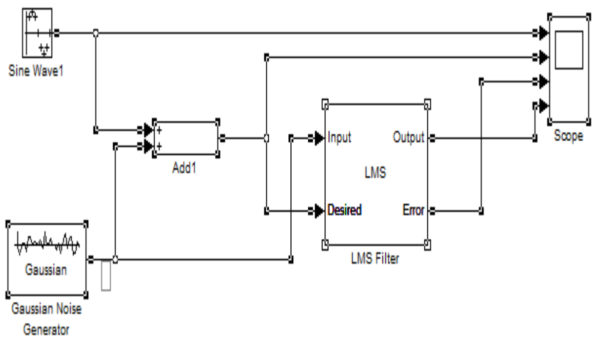

Figure 3: The MATLAB simulink model of the adaptive noise cancellation scheme

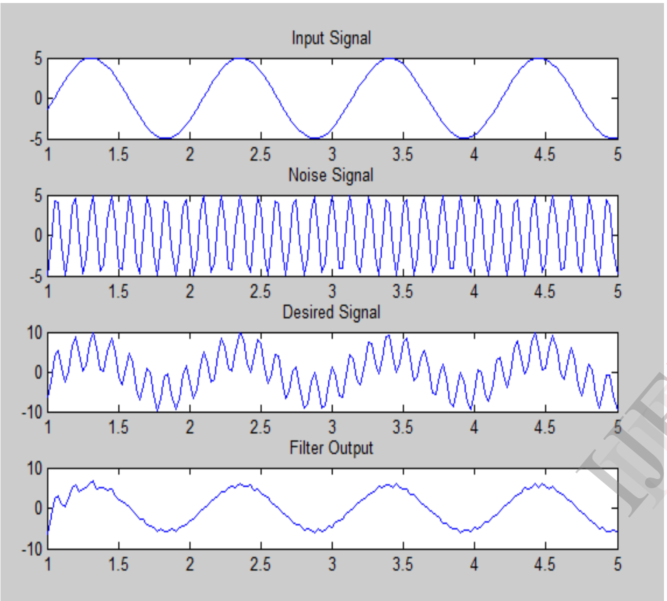

Figure 4: Filtered signal obtained using MATLAB

System inputs are analog signal and Gaussian noise signal. The system outputs are the sinusoidal signal after filtering [6]. Comparisons are worked out in the form of figures, which show the input, desired and error signals. By using manual switch LMS Adaptive filter Step size parameter is changed between high and low constant values. If the step size parameter is at higher constant value the response is fast but showing less accurate and if step size factor is at lower constant value the response may be slow but it is showing more exact performance.

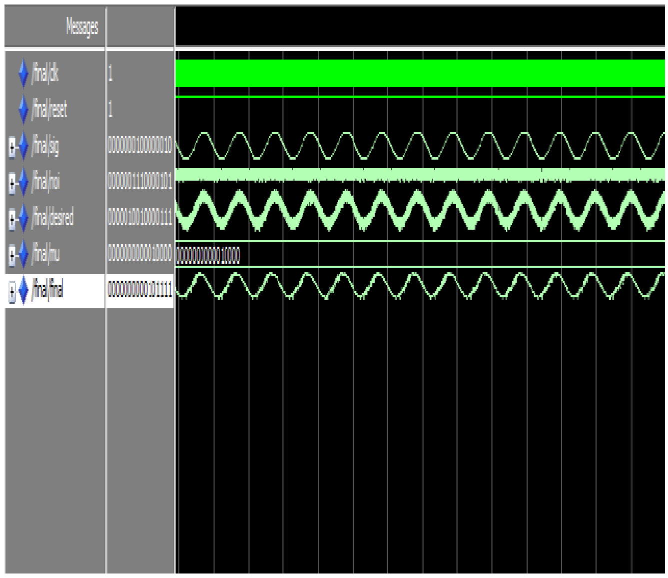

Figure 5: Filtered signal obtained using MODELSIM

The proposed design implemented on Xilinx – Spartan 3E based FPGA can work at maximum operating frequency of 25.384 MHz. The total power consumption of the proposed design based on 3s200pq208-5 FPGA device has been calculated and it can be observed from that proposed design has consumed 0.152W at 29.1° C. The parameter μ is varied. Various outputs are obtained for various step size i.e. μ = 0.002, 0.04, etc. Here we used μ = 0.0078.

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Table 1: comparative analysis based on Spartan 3E

Device Utilization Summary (Spartan 3E)			
Logic Utilization	Used	Available	Utilization
Number of Slices	842	1920	43%
Number of Slice Flip Flops	334	3840	8%
Number of input LUTs	1608	3840	41%
Number of bonded IOBs	18	14	12%
Number of MULT 18*18s	10	12	84%
Number of GCLKs	1	8	12%

V. CONCLUSION

This paper presents the real-time implementation of the adaptive noise cancellation scheme proposed in using MATLAB. The conclusion of above results is that, LMS filter is used for noise cancellation and complexity reduction. The principal advantages of the method are its adaptive capability, its low output noise, and its low signal distortion. The adaptive capability allows the processing of inputs whose properties are unknown. Output noise and signal distortion are generally lower than can be achieved with conventional optimal filter configurations. LMS algorithm converges more quickly, but at the expense of granularity – the LMS Filter Output is not as smooth. Step size is not too small it takes time to converge, and step size is not too large, filter response is not converging in case of large step size.

REFERENCES

[1] Ahmed Elhossini, Shawki Areibi and Robert Dony, "An FPGA Implementation of the LMS Adaptive Filter for Audio Processing", IEEE Symp. on Field-Programmable Custom Computing Machines (FCCM04), pp. 324- 335, Apr 2006.

[2] Bernard Widrow, John M. McCool, John Kaunitz, Charles Williams, Robert H. Hearn, Eugene Dong, "Adaptive Noise Cancelling: Principles and Applications", Proceedings of the IEEE, Vol. 63, pp. 12, Dec 1975.

[3] Dukel.B, Rizkalla.M.E, and Salama.P, "Implementation of Pipelined LMS Adaptive Filter for Low-Power Applications", Proc.45th IEEE Int. Midwest Symp.Circuits and Systems, Tulsa, Vol.2, pp.533-536, Jun 2009.

[4] Marcus Zeller, Cristian Contan, and Marina Topa1, "Excitation-Dependent Step size Control Of Adaptive Filters For Acoustic Echo Cancellation", Proc.20th European Signal Processing Conference, pp.604-608, Aug 2012.

[5] Rafael Ramos, Joaquin Del Rio, and Gerard Olivar, "FPGA-Based Implementation of an Adaptive Canceller for 60-Hz Interference in Electrocardiography", IEEE Trans. On Instrum. Meas.,Vol. 56, pp.2633-2640, Dec 2007.

[6] Ramakrishna.V and Anilkumar.T, "Low Power VLSI Implementation of Adaptive Noise Canceller Based on Least Mean Square Algorithm", IEEE transactions on circuits and systems, vol.6, July 2013.

[7] Tian Lan and Jinlin Zhang, "FPGA Implementation of an Adaptive Noise Canceller", Proc.InT. Information Processing (ISIP08), pp.553–558, May 2008.

[8] Ying He, Hong He, Li Li, Yi Wu, "The Applications and Simulation of Adaptive Filter in Noise Canceling", International Conference on computer Science and Software Engineering, pp.1-4, Jun 2008.

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