DESIGN AND IMPLEMENTATION OF ADAPTIVE ECHO CANCELLER BASED LMS & NLMS ALGORITHM

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ABSTRACT

This paper proposes the adaptive echo cancellation using normalized least mean square (NLMS) algorithm. The NLMS algorithm is the normalized version of least mean square (LMS) algorithm. Todays technical scenario communication system, possesses additive noise, signal interference and echo etc. Due to this reason the error is generated at the time of transmission of data. Hence adaptive filter is a appropriate option to reduce the noise or channel effects. In this paper high speed of adder, substractor and Adaptive filter coefficients to design LMS and NLMS algorithm is realized. This paper focuses on the use of LMS and NLMS algorithms to reduce this unwanted echo, thus increasing communication quality. This work better highlights differences between algorithm performance than previously published work and sheds new light on algorithm behavior in terms of SNR. This work were created in the popular VHDL language. The NLMS algorithm establishes a better balance between simplicity and performance than least mean square algorithm.

Keywords: Adaptive Filter, NLMS Algorithm, LMS Algorithm, Vhdl Language

I. INTRODUCTION

An echo is a reflection of sound, arriving at the listener some time after the direct sound. Echo is the reflected copy of the voice heard some time later and delayed version of the original. The term echo cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. Echo cancellation is the process which removes unwanted echoes from the signal on a telephone line. It includes first recognizing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it can be removed by 'subtracting' it from the transmitted or received signal. Multiple reflections in acoustic enclosures and transmission delay affect the sound quality, which in the case of a teleconferencing system lead to a poor understanding of the conversation. In addition to improving subjective quality, this process increases the capacity achieved through silence suppression by preventing echo from traveling across a network. Hands-free phone is a basic and intrinsic application with small information terminals such as cell phones, smart phones, and tablet PCs. For hands-free communication, echo cancellation is common but still a difficult function. An echo canceller has an adaptive filter to emulate the echo path between the input of amplifier to drive a loudspeaker and the microphone. Even

with echo cancellers, suppression of echo is very difficult because the loudspeaker is small and close to the microphone, and the sound from the loudspeaker is very loud. Echo cancellation try to eliminate the echo from the transmitted audio signal with a special optimization algorithm. The algorithm produces copies of the received signal and checks for parts of the signal that reappear with some delay. This reappearing parts are then subtracted from the signal, the echo is removed. In this paper, the LMS and NLMS adaptive algorithms are described. The following sub-sections will present a brief description of adaptive filters focused on the cancellation of noise and echo. The second section if focused on briefly review of the adaptive filtering algorithm. The third section presents acoustic echo cancellation. The fourth section presents literature survey. The fifth section shows the methodology used. The last section presents the simulation and experimental results obtained from the adaptive cancellation of noise & it summarizes the main findings of the paper.

II. VARIOUS ADAPTIVE FILTERING ALGORITHMS

A filter which adapts itself to the input signal given to it. Adaptive Filtering dealing with adaptive filters and system design. The main function of adaptive filtering is the development of a filter capable of adjusting to the statistics of the signal. Usually, an adaptive filter takes the form of a FIR filter, with an adaptive algorithm that modifies the values of its coefficients. They are used in a wide range of applications including system identification, noise cancellation, signal prediction, echo cancellation and adaptive channel equalization, process control. The main configurations of adaptive filters are the adaptive cancellation of noise and echo. For this justification, the filter uses an adaptive algorithm to change the value of the filter coefficients, so that it acquires a better approximation of the signal after each iteration. The LMS, and its variant the NLMS are two of the adaptive algorithms widely in use. Also the normalized version of the LMS algorithm, called Normalized Least Mean Square (NLMS) algorithm is widely used. NLMS algorithm has been used more often in real time applications. The LMS algorithm have slow convergence and poor tracking as compare to the the NLMS algorithm. Both algorithms require a small number of multiplications and additions for the update of the coefficients.

2.1 LMS Algorithm [9]

The LMS algorithm adapts the filter tap weights so that e(n) is minimized in the mean-square sense. LMS algorithm is sensitive to variation in step size parameter. LMS algorithm requires number of itrations equal to dimensionality of the input. When the processes x(n) & d(n) are jointly stationary, this algorithm converges to a set of tap-weights which, on average, are equal to the Wiener-Hoff solution. It acts as a negative feedback to minimize error signal.

1. The output of the FIR filter, y(n) is

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n)$$

2. The value of the error estimation is

$$e(n) = d(n) - y(n)$$

3. The tap weights of the FIR vector are updated in preparation for the next iteration

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu e(n)\mathbf{x}(n)$$

The main reason for the LMS algorithms popularity in adaptive filtering is its computational simplicity, making it easier to implement than all other commonly used adaptive algorithms. The only disadvantage is its weak convergence. For each iteration the LMS algorithm requires 2N additions and 2N+1 multiplication. Where n= 0,1,2,....n.

1.2 NLMS Algorithm [9]

As the NLMS is an extension of the standard LMS algorithm, the NLMS algorithms practical implementation is very similar to that of the LMS algorithm. One of the drawback of LMS algorithm is selection of step size parameter. In order to solve this difficulty we can use the NLMS algorithm. Here the step size parameter is normalized.So the NLMS algorithm is time varying step size algorithm. The NLMS algorithm is used in system in order to improve voice quality. Each iteration of the NLMS algorithm requires these steps in the following order

1. The output of the adaptive filter is calculated

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = \mathbf{w}^T(n)\mathbf{x}(n)$$

2. An error signal is calculated as the difference between the desired signal and the filter output

$$e(n) = d(n) - y(n)$$

3. The step size value for the input vector is calculated

$$\mu(n) = \frac{1}{\mathbf{x}^T(n)\mathbf{x}(n)}$$

4. The filter tap weights are updated in preparation for the next iteration

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

Each iteration of the NLMS algorithm requires 3N+1 multiplications, this is only N more than the standard LMS algorithm. This is an acceptable increase considering the gains in stability and echo attenuation achieve[10]. Where n= 0,1,2,....n.

Adaptive filters are systems with four terminals as shown in Fig.1, where x is the input signal, d is the desired signal, y is output signal of filter and e is the error signal [4]. Adaptive filters design technique may be digital, analog or mixed. Every technique presents advantages and disadvantages, for example, analog adaptive filters are very fast, but offset avoids getting the least error [4]. Digital filters are slow but precise, because is necessary the use of a lot of components, due to floating point operations. Mixed design (analog and digital), offers a good compromise between precision and speed, but VLSI design is more complicated because is necessary to separate analog and digital components inside the chip .The LMS algorithm is one of the most used algorithms because it is easy and stable. The only disadvantage is its weak convergence [3]. The length of the FIR filter is 64 and the size of the step is 0.001.

Two inputs are required:

- A reference noise that should be related with the noise that exists in distorted the input signal. This means that the noise comes from the same source.
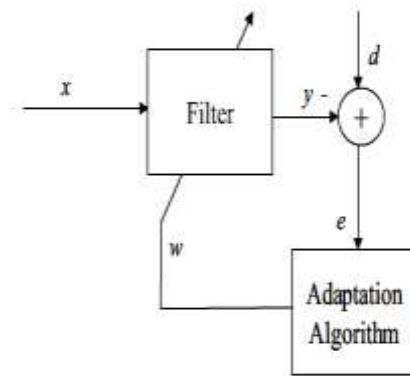- An error signal already calculated.

Figure 1 Adaptive Filter

III. ACOUSTIC ECHO CANCELLATION (AEC)

The AECs is a system identification application as shown in Fig.2, which uses adaptive filters to obtain a copy of the acoustic transfer function (the response of an enclosure to an acoustic impulse). Acoustic echo occur when an audio signal is reverberated in a real environment, resulting in the original intended signal plus attenuated and delay time. The signal applied to the loudspeaker(s) x(n) propagates through multiple acoustic paths and it is picked up by the microphone(s). This signal is used as the desired signal d(n) in the system identification process. The output of the adaptive filter y(n) is determined by convolving the samples x(n) with the adaptive filter coefficients w(n).The filter is altered iteratively to minimize the error signale(n). The coefficient update can be carried out with various adaptive algorithms.
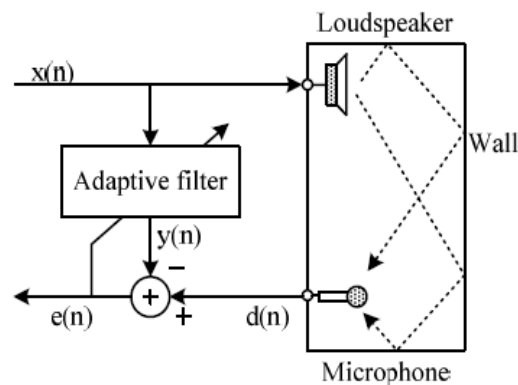

Figure 2 Acoustic Echo Cancellation

Public addressing systems are affected by acoustic feedback that may lead to the saturation of the system. In order to improve the sound quality and prevent audio feedback the acoustic echo cancellers (AECs) are deployed to remove the undesired echoes resulting from the acoustic coupling between the loudspeaker(s) and the microphone(s).

IV. LITERATURE SURVEY

The paper [2] presents a solution to noise or echo cancellation and a hardware real-time implementation of the LMS algorithm. The overall performance of the designed adaptive filter compared with other implemented systems using the same filter is good, and with further improvements the results will improve. The LMS

algorithm provides good numerical stability and its hardware requirements are low. On the other hand, the NLMS algorithm is one of the most implemented adaptive algorithms in actual industrial applications.

This paper [3] proposes an FPGA implementation of an Adaptive Noise Canceller using the LMS algorithm. In this paper, in order to show the performance of FPGA in digital signal processing applications, implement an Adaptive Noise Canceller on an FPGA and use the LMS algorithm as the adaptive filtering algorithm of the Adaptive Noise Canceller. The performance of the LMS algorithm implemented by hardware is comprehensively analyzed in terms of convergence performance, truncation effect and tracking ability.

In paper [4] an echo canceller is presented, using an adaptive filter with a modified LMS algorithm, where this modification is achieved coding error on conventional LMS algorithm.

V. METHODOLOGY USED

The architecture of adaptive filter is already implemented with LMS Algorithm for Adaptive echo cancellation. Also the comparison between LMS & NLMS algorithm is being done. The NLMS algorithm outperforms the LMS algorithm, in terms of Mean Square Error. It has a better convergence than LMS algorithm. The NLMS adaptive filtering algorithm is expressed by its simplicity in implementation and its stability. These advantages recommend the NLMS algorithm as a good choice for real time implementation. Hence the NLMS algorithm can be implemented for Adaptive echo cancellation in VHDL. Also the comparison of LMS and NLMS algorithm for echo cancellation along with signal to noise ratio has been done. Fig.3 shows the architecture used. The system architecture consist of main four modules such as 16 bit adder, 16 bit substractor, multiplier and delay.
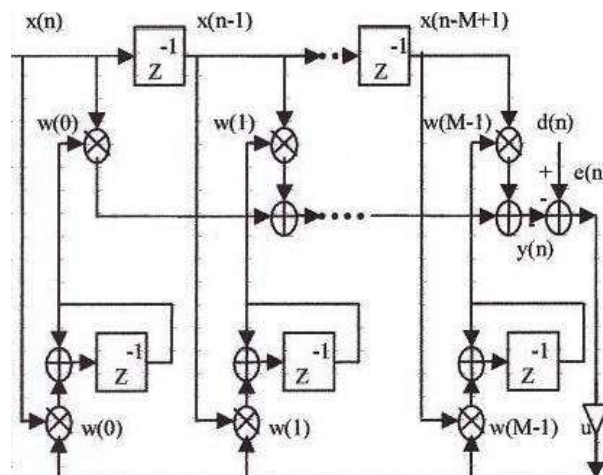

Figure 3 System Architecture

Where, $x(n), x(n-1), \ldots, x(n-M+1)$ is the tap inputs

$x(n)$ is the reference signal

$M-1$ is the number of delay elements.

$d(n)$ denotes the primary input signal

$e(n)$ denotes the error signal and

$wi(n)$ denotes the tap weight at the nth iteration..

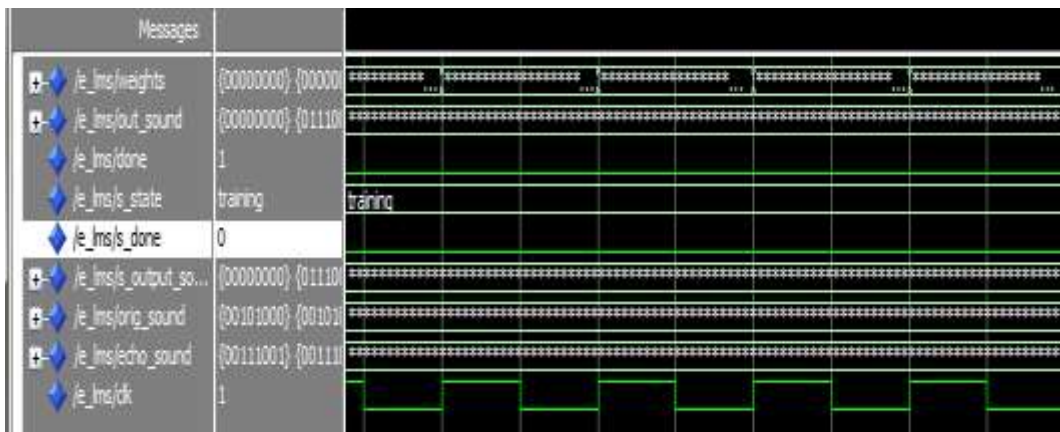## VI. RESULT AND CONCLUSIONS

6.1 VHDL Result for Signal 1

Figure 4 Training Signal 1 of LMS algorithm

Figure 5 Testing signal 1 of LMS algorithm

Figure 6 Training signal 1 of NLMS algorithm

Figure 7 Testing signal 1 of NLMS algorithm
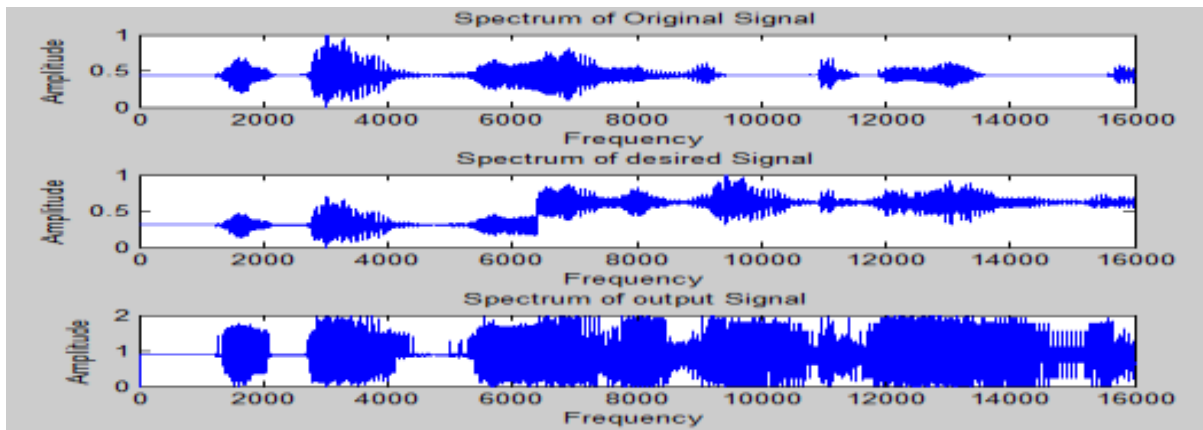
6.2 MATLAB Results for LMS & NLMS Algorithm

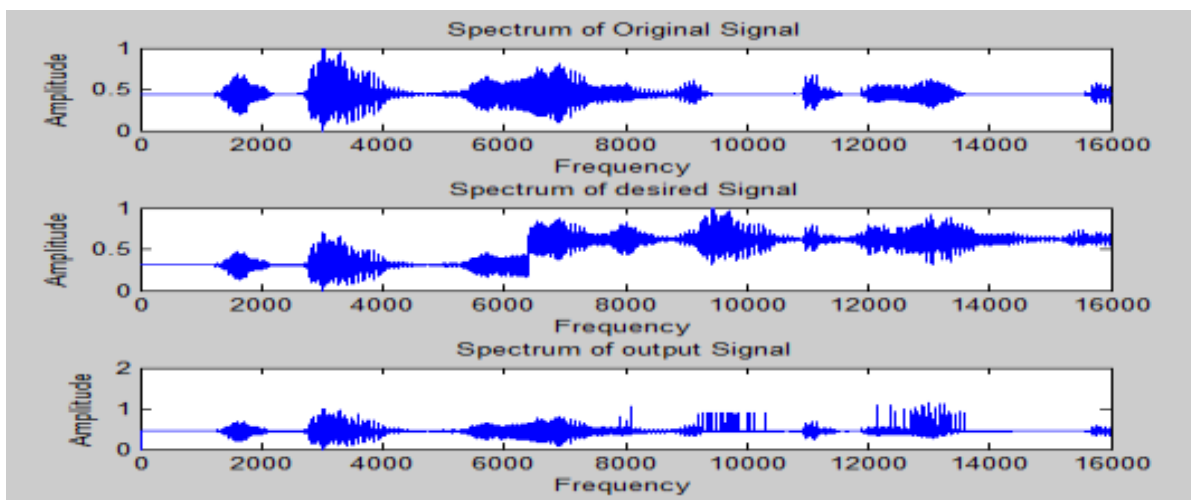**Figure 8 Signal 1 of LMS Algorithm**

Figure 9 Signal 1 of NLMS Algorithm

The PSNR of the LMS algorithm is 4.08 dB and the PSNR of the NLMS algorithm is 29.72 dB. Whereas the mean square error of the LMS algorithm is 0.39 and the mean square error of the NLMS algorithm is 0.00106.
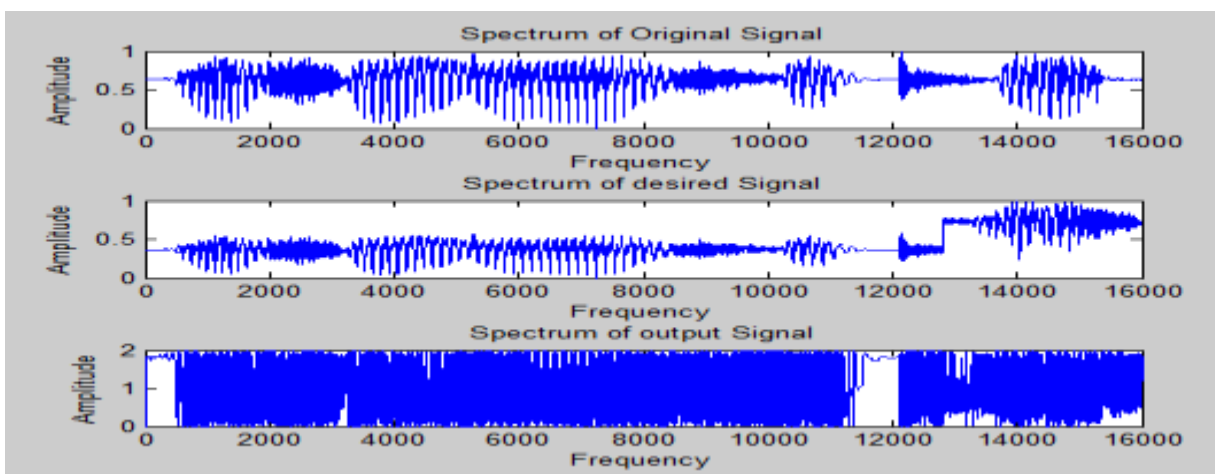

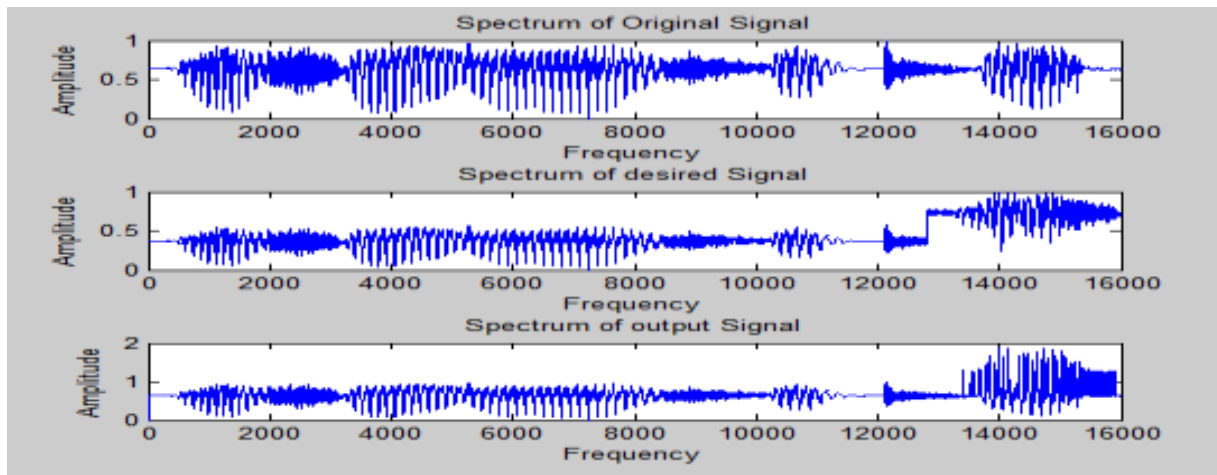**Figure 10 Signal 2 of LMS algorithm**

Figure 11 Signal 2 of NLMS Algorithm

The PSNR of the LMS algorithm is 2.88 dB and the PSNR of the NLMS algorithm is 22.17 dB. Whereas the mean square error of the LMS algorithm is 0.514 and the mean square error of the NLMS algorithm is 0.006.

The adaptive echo canceller based LMS and NLMS algorithm is successfully designed wherein comparison of LMS and NLMS algorithm has been done in terms of PSNR. The PSNR of the NLMS algorithm is 25.945 dB and the PSNR of the LMS algorithm is 3.48dB. So the PSNR is maximum it realize better sound quality.

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