TO STUDY LMS & NLMS ALGORITHM FOR ADAPTIVE ECHO CANCELLATION

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ABSTRACT

Communication Systems développent increases considerably obtaining more troubles as additive noise, signal interference and echo, therefore errors in data transmission are generated, but adaptive filter is an option to reduce these channel effects. This paper represents the adaptive echo cancellation using normalized least mean square (NLMS) algorithm. The NLMS algorithm is the normalized version of LMS algorithm. The normalized least mean square algorithm demonstrates a better balance between simplicity and performance than least mean square algorithm.

Keywords: Adaptive Filtre, NLMS Algorithme, VHDl Language

I. INTRODUCTION

Echo is the repetition of a waveform due to reflection from points where the characteristics of the medium through which the wave propagates changes. 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. In addition to improving subjective quality, this process increases the capacity achieved through silence suppression by preventing echo from traveling across a network. Echo cancellation involves 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. Hands-free phone is a basic and essential 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 remove the echo from the transmitted audio signal with a special algorithm. The algorithm creates 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.

II. ADAPTIVE FILTERING ALGORITHMS

Adaptive Filtering dealing with adaptive filters and system design.They are used in a wide range of applications including system identification, noise cancellation, signal prediction, echo cancellation and adaptive channel equalization. The main configurations of adaptive filters are the adaptive cancellation of noise and the adaptive cancellation of echo. For this purpose, 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 (Least Mean Square), and its variant the NLMS (Normalized LMS) are two of the adaptive algorithms widely in use. One of the most popular adaptive algorithms available in the literature is the Least Mean Square (LMS). The main reason is the simplicity in implementation. Also widely used is the normalized version of the LMS algorithm, called Normalized Least Mean Square (NLMS) algorithm. 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 normalized least-mean-square (NLMS) algorithm. Both algorithms require a small number of multiplications and additions for the update of the coefficients, which make them suitable for digital design.

2.1 LMS Algorithm

The LMS algorithm changes (adapts) the filter tap weights so that e(n) is minimized in the mean-square sense. 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-Hopf solution.

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. For each iteration the LMS algorithm requires 2N additions and 2N+1 multiplication.

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2.2 NLMS Algorithm

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. 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].

Communication systems development increases considerably obtaining more troubles as additive noise, signal interference and echo, therefore errors in data transmission are generated, but adaptive filter is an option to reduce these channel effects. Adaptive filters are systems with four terminals as shown in Figure 1, where x is the input signal, d is the wish signal, y is output signal filter and e is the output filter error [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]. 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.

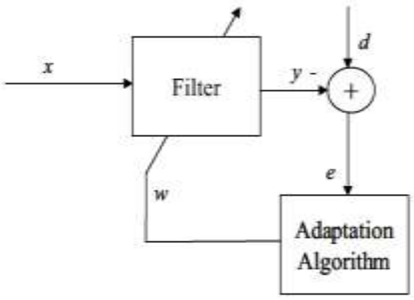

Figure1. Adaptive filter

III. ACOUSTIC ECHO CANCELLATION

The AECs is a system identification application Figure2, which uses adaptive filters to obtain a copy of the acoustic transfer function (the response of an enclosure to an acoustic impulse). 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 obtained by convolving the samples x(n) with the adaptive filter coefficients w(n).The filter is altered iteratively to minimize the error signal(n). The coefficient update can be carried out with many algorithms. One of the most popular adaptive algorithms available in the literature is the Least Mean Square (LMS).

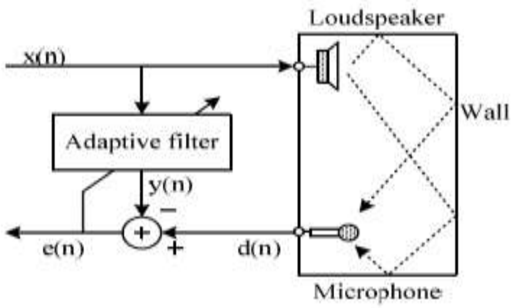

Figure 2. Acoustic Echo Cancellation

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. 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. PREVIOUS WORK

[2] The paper presents a solution to noise / 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 telecom/industrial applications.

[3] This paper proposes an FPGA implementation of an Adaptive Noise Canceller using the Least Mean Square (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.

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

V. PROPOSED WORK

Communication systems development increases considerably obtaining more troubles as additive noise, signal interference and echo, therefore errors in data transmission are generated, but adaptive filter is an option to reduce these channel effects. The LMS algorithm is one of the most used algorithms because it is easy and stable. The only disadvantage is its weak convergence. The NLMS algorithm is the normalized version of LMS algorithm. So, it has a better convergence. 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. Thus we have to use the NLMS algorithm as the adaptive filtering algorithm of the adaptive echo canceller.

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 MSE. It has a better convergence. 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.

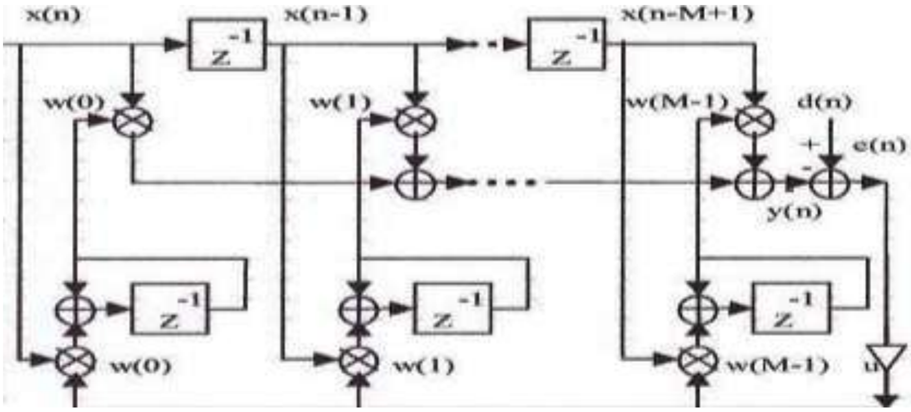

Figure 3. Proposed Architecture

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**i(n) denotes the tap weight at the nth iteration.

VI. CONCLUSION

In this, the proposed work will results the NLMS algorithm has better convergence. When we applied input signal then some echo is created. So our main aim to cancel this echo based on LMS & NLMS algorithm which will meet the following specifications:

- Convergence Performance
- Peak signal to noise ratio
- Truncation Effect
- Tracking Ability

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