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FPGA IMPLEMENTATION OF NOISE CANCELLATION USING ADAPTIVE ALGORITHMS

RLS ADAPTIVE FILTER

Ms.Preethi Vadivel[1], Mrs.R.Ramya[2], Mr.R.Poomurugan.[3]

[1,2,3]Department of ECE, Gnanamani College of Technology Namakkal, India

Abstract —This paper describes the concept of adaptive noise cancelling. The noise cancellation using the Recursive Least Squares (RLS) to remove the noise from an input signal. The RLS adaptive filter uses the reference signal on the Input port and the desired signal on the desired port to automatically match the filter response in the Noise Filter block. The filtered noise should be completely subtracted from the "noisy signal" of the input Sine wave & noise input signal, and the "Error Signal" should contain only the original signal. Finally, the functions of field programmable gate array based system structure for adaptive noise canceller based on RLS algorithm are synthesized, simulated, and implemented on Xilinx XC3s200 field programmable gate array using Xilinx ISE tool.

Keywords – **RLS Algorithm; Adaptive Filter; Noise Canceller; Error estimation**

I. INTRODUCTION

Noise problems in the environment have gained attention due to the tremendous growth of technology that has led to noisy engines, heavy machinery, audio devices and other noise sources. The problem of controlling the noise level has become the focus of a vast amount of research over the years. If accurate information of the signals to be processed is available, the designer can easily choose the most appropriate algorithm to process the signal. When dealing with signals whose statistical properties are unknown, fixed algorithms do not process these signals efficiently. The solution is to use an adaptive filter that automatically changes its characteristics by optimizing the internal parameters. The adaptive filtering algorithms are essential in many statistical signal processing applications.

The adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, allowing the filter to adapt to changes in the input signal characteristics. Because of their self adjusting performance and in- built flexibility, adaptive filters are used in many diverse applications such as echo cancellation, radar signal processing, navigation systems, and equalization of communication channels and in biomedical signal enhancement .

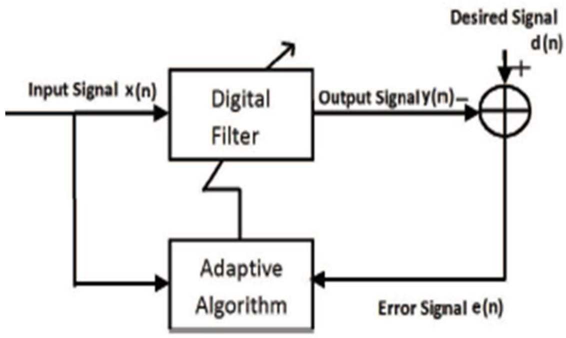

Fig 1. Block diagram of Adaptive Filter

The common adaptive algorithms that have found widespread application are the Least Mean Squares (LMS) and the Recursive Least Squares (RLS).Performance of adaptive filters using RLS algorithm has fast convergence speed and better mean square error.

II. RLS ALGORITHM

The Recursive Least Squares (RLS) algorithm is based on the well-known least squares method . The least-squares method is a mathematical procedure for finding the best fitting curve to a given set of data points. This is done by minimizing the sum of the squares of the offsets of the points from the curve. The RLS algorithm recursively solves the least squares problem. The forgetting factor λ is a positive constant less than unity, which is roughly a measure of the memory of the algorithm and the regularization parameter's value is determined by the signal-to-noise ratio (SNR) of the signals. The w(n) represents the adaptive filter's weight vector and the M-by-M matrix P is referred to as the inverse correlation matrix. This gain vector is multiplied by the a priori estimation error e(n) and added to the weight vector to update the weights. Once the weights have been updated the inverse correlation matrix is recalculated, and the training resumes with the new input values.

A summary of the RLS algorithm follows:

Output signal,

$$y(n) = u^T(n).w(n) \qquad (1)$$

Where ,u(n) is the filter input vector

$$u(n) = [x(n)x(n-1) \ldots . x(n-N+1)]^T \qquad (2)$$

w(n) is the filter coefficient vector.

$$w(n) = [w_0(n)w_1(n) \ldots . w_{N-1}(n)]^T \qquad (3)$$

Error signal,

$$e(n) = d(n) - y(n) \qquad (4)$$

Updates the filter coefficients by using the following equation:

$$w(n+1) = w(n) + e(n).k(n) \qquad (5)$$

where w(n) is the filter coefficients vector and k(n) is the gain vector. K(n) is defined by the following equation:

$$k(n) = \frac{P(n).u(n)}{\lambda + u^T(n).P(n).u(n)} \qquad (6)$$

where λ is the forgetting factor and $P(n)$ is the inverse correlation matrix of the input signal.

RLS algorithm uses the following equation to update this inverse correlation matrix.

$$P(n+1) = \lambda^{-1}P(n) - \lambda^{-1}k(n).u^T(n).P(n) \qquad (7)$$

III. ADAPTIVE NOISE CANCELLATION

The primary aim of an adaptive noise cancellation is to allow the noisy signal through a filter which suppresses the noise without disturbing the desired signal.

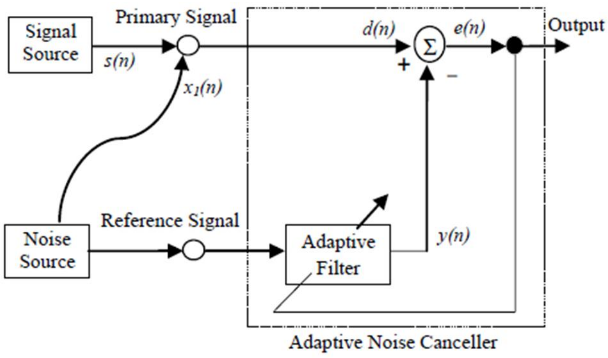

Fig 2. Adaptive Noise Cancellation System

S(n)- source signal
d(n)-primary signal
x1(n)-noise signal
x(n)-noise reference input
y(n)-output of adaptive filter
e(n)-system output signal

Fig. 2 shows the adaptive noise cancelling concept,the corrupted signal passes through a filter that tends to suppress the noise while leaving the signal unchanged. This process is an adaptive process, which means it cannot require a priori knowledge of signal or noise characteristics. 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) □ x_1(n)). 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)). 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. Then the output y(n) is subtracted from the primary input to produce the system output

$$e(n) = s(n) + x_1(n) - y(n) \qquad (8)$$

This is the denoised signal.

In the system shown in Fig. 1 the reference input is processed by an adaptive filter. 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. In noise cancelling systems the practical objective is to produce a system output e(n)=s(n)+ $x_1(n)$ –y(n) 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 RLS adaptive algorithm to minimize total system output power.

IV. RESULTS

A. Simulink Model

The following figure shows simulink model of RLS adaptive filter.

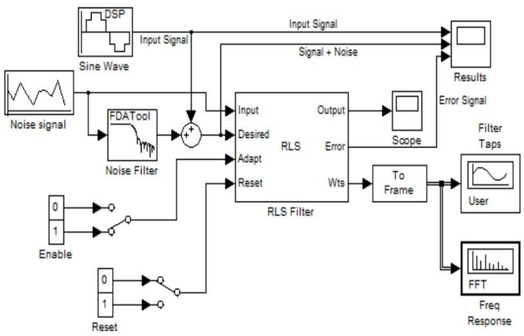

Fig 3. simulink model of RLS adaptive filter

System inputs are analog signal and Gaussian noise signal. The system outputs are the sinusoidal signal after filtering. Comparisons are worked out in the form of figures, which show the input, desired and error signals. By using RLS Adaptive filter Step size parameter is changed between high and low values. The response is fast and showing more accurate performance.

The following Figure shows the result of the simulated RLS filter. It shows the input signal,which is a sine wave. Then it shows the input signal with the noise signal. Lastly it shows the error signal.

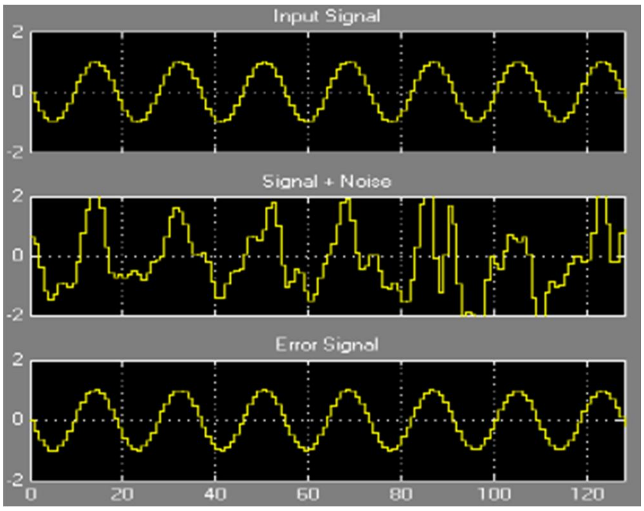

Fig 4. Result of RLS filter

B. Xilinx output

The following figure shows the Xilinx 12.1 development environment, for implementing the proposed Verilog designof the RLS algorithm. The design is written in Verilog and simulated using ISE simulator.

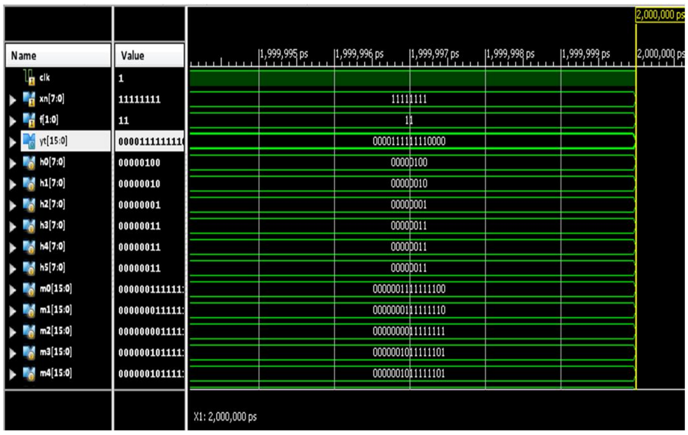

Fig 5. Xilinx output

V. CONCLUSION

In this paper, efficient adaptive noise canceller has been simulated using RLS algorithm. By this, the recursive least squares (RLS) algorithms have a faster convergence speed, the input signals are considered as deterministic, while for the LMS and other algorithms they are considered as stochastic.However,RLS algorithm requires more computational resourses and involves complicated mathematical operations.

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