Implementation of Adaptive Noise Canceller using LMS Algorithm

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ABSTRACT
This paper describes the concept of adaptive noise cancelling, an alternative method of estimating signals corrupted by additive noise or interference. A desired signal corrupted by additive noise can often be recovered by an adaptive noise canceller using the least mean squares (LMS) algorithm. This Adaptive Noise Canceller is then useful for enhancing the S/N ratio of data collected from sensors (or sensor arrays) working in noisy environment, or dealing with potentially weak signals. The principle 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. In this paper, the aim is to reduce the noise in speech signal and improve the quality of speech signal as well as to improve the efficiency of the data transmission and adding more feature without doing major changes. FPGA implementation of ANC done using Xilinx ise 9.1 software and for synthesis use the Spartan 3 device.

General Terms
LMS Algorithm

Keywords
ANC, LMS, FPGA

1. INTRODUCTION
In this modern world, communication is the lifeline of the world. Surrounding is full by all kinds of signals in various forms. Some of the signals are natural, but most of the signals are man-made. Some signals are necessary (speech); some are pleasant (music), while many are unwanted or unnecessary in a given situation. In an engineering context, signals are carriers of information, both useful and unwanted. When the communication process is in progress number of error will introduce. There are various limitation and disadvantage in transmission and reception process so here this paper is trying to give the advance noise cancellation technique in which filter along with ANC (adaptive noise cancellation) is used to reduce the noise. This paper implements such system which gives the correct output without any error.

In common use, the word noise means any unwanted sound. But technically, Noise is unwanted electrical or electromagnetic energy that degrades the quality of signals and data. In both analog and digital electronics, noise is an unwanted perturbation to a wanted signal and can affect files and communications of all types, including text, programs, images, audio, and telemetry. Noise generated inside wireless receivers, known as internal noise, is less dependent on frequency. Engineers are more concerned about internal noise at high frequencies than at low frequencies, because the less external noise there is, the more significant the internal noise becomes.

An adaptive filter is a filter that self-adjusts its transfer function according to an optimization algorithm driven by an error signal. Because of the complexity of the optimization algorithms, most adaptive filters are digital filters. By way of contrast, a non-adaptive filter has a static transfer function. Adaptive filters are required for some applications because some parameters of the desired processing operation (for instance, the locations of reflective surfaces in a reverberant space) are not known in advance. The adaptive filter uses feedback in the form of an error signal to refine its transfer function to match the changing parameters. Generally speaking, the adaptive process involves the use of a cost function, which is a criterion for optimum performance of the filter, to feed an algorithm, which determines how to modify filter transfer function to minimize the cost on the next iteration.

2. Least mean square (LMS) algorithm

The Least Mean Square (LMS) algorithm, introduced by Widrow and Hoff in 1959 is an adaptive algorithm, which uses a gradient-based method of steepest decent. LMS algorithm uses the estimates of the gradient vector from the available data. LMS incorporates an iterative procedure that makes successive corrections to the weight vector in the direction of the negative of the gradient vector which eventually leads to the minimum mean square error. Compared to other algorithms LMS algorithm is relatively simple; it does not require correlation function calculation nor does it require matrix inversions.

The LMS algorithm is most commonly used adaptive algorithm because of its simplicity and a reasonable performance. It has a stable and robust performance against different signal conditions. However it may not have a really fast convergence speed compared other complicated algorithms like the Recursive Least Square (RLS). Usually traffic conditions are not static, the user and interferer locations and the signal environment are varying with time, in which case the weights will not have enough time to converge when adapted at an identical rate. That is,μ the step-size needs to be varied in accordance with the varying traffic conditions. The least mean square algorithm is a linear adaptive filtering algorithm that consists of two basic process.

1) Filtering process:

This involves (a) computing the output of a transversal filter produced by a set of tap inputs, and (b) generating an estimation error by comparing this output to a desired response.

2)Adaptive process:

This involves the automatic adjustment of the tap weights of the filter in accordance with the estimation error. Thus, the combination of these two processes working together constitutes a feedback loop around the LMS algorithm. The transversal filter, around which the LMS algorithm is built, this is responsible for performing the filtering process. The other one is mechanism for performing the adaptive control process on the tapweights of the transversal filter, hence the designation "adaptive weight control mechanism".

The LMS algorithm is a widely used algorithm for adaptive filtering. The algorithm is described by the following equations:

$$1) y(n) = \sum_{i=0}^{M-1} w_i(n) * x(n-1) ....(1)$$

2) e(n) = d(n) – y(n) ……………..(2)

3) wi(n+1) = wi(n) + 2ue(n)x(n-i) …………..(3)

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

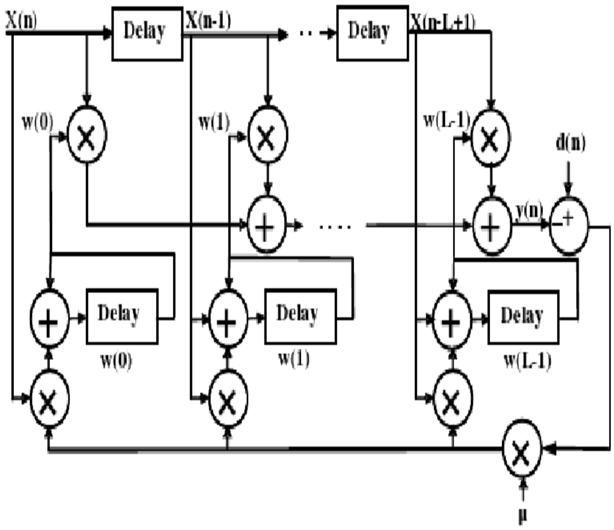

Figure 2.1 Flowchart of LMS algorithm

3. CONCEPT OF ADAPTIVE NOISE CANCELLER

As the name implies, ANC is a technique used to remove an unwanted noise from a received signal, the operation is controlled in an adaptive way in order to obtain improved signal-to noise ratio (SNR). ANC technique has been successfully applied to many applications, such as acoustic noise reduction, adaptive speech enhancement and channel equalization. As shown in Fig.3.1 an ANC is typically a dual-input, closed-loop adaptive feedback system. The two inputs are: the primary input signal d(n) i.e. the desired signal corrupted by the noise and the reference signal x(n) i.e. an interfering noise supposed to be uncorrelated with the desired signal but correlated with the noise affecting the desired signal in an unknown way.

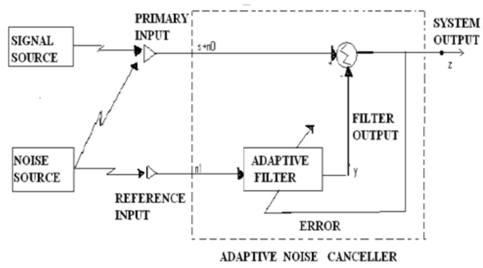

Figure 3.1 Adaptive Noise Canceller

4. LMS CORE IMPLEMENTATION

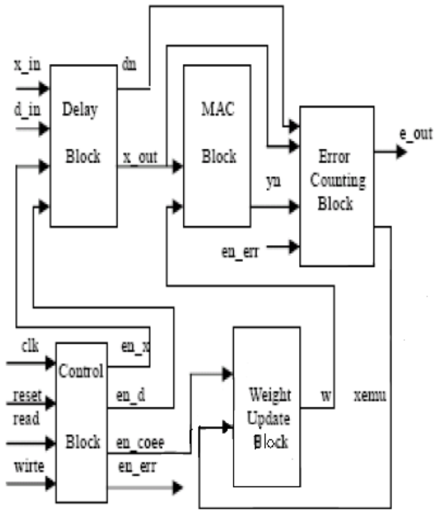

Figure 4.1 Block diagram of the LMS core

The LMS core is divided into five blocks-

1. Control block
2. Delay Block
3. Multiply Accumulator (MAC) Block
4. Error Counting Block
5. Weight Update Block

4.1 Control block

The **Control block** arranges the timing of the whole system. It produces four enable signals: en_x,en_d,en_coee, en_err, which enable the Delay Block, the Weight Update Block and the Error Counting Block separately. When read=1 all the enable signals get 1 that means it will read input and produce output. When write=1 all the enable signals get 0 that means it will not read any input just check output.

4.2 Delay block

The **Delay Block** receives the reference signal x_in and the primary input signal d_in under control of the enable signal en_x and en_d. And it produces the M tap delay signal x_out.

When enable signals get 1 output follows the input otherwise it will produce delay singal.

4.3 MAC block

The **Multiply Accumulator (MAC) Block** multiply the M_tap reference signal x_out with the M_tap weight w separately, and add them together, then we get yn. In MAC block as when clk=1 and reset=1 then filter output=0 and when clk=0 and reset=0 then filter output by multiplying reference input and weight and add them separately with previous filter output.

4.4 Error counting block

The **Error Counting Block** subtracts yn from dn and get the error signal e_out, which is also the output of the whole system. And it produces signal xemu as a feedback by multiplying e_out, x_out and the scaling factor u. When enable signal en_err get 1 it will generate error signal eout and feedback signal xemu.

4.5 Weight Update Block

The **Weight Update Block** updates the weight vector w(n) to w(n+1) that will be used in the next iteration. When enable signal en_coee get 1 next weight equal to current weight plus feedback signal otherwise next weight equal to zero.

5. DESIGN AND SIMULATION

The LMS algorithm satisfied three equations which are deriving by LMS core block.

5.1 SIMULATIONS RESULTS

The code is written in a hardware description language called VHDL and it is simulated in Xilinx ISE 9.1i. The simulation results obtained practically are found to be equivalent to the results obtained by mathematical calculations.

5.2 LMS core block

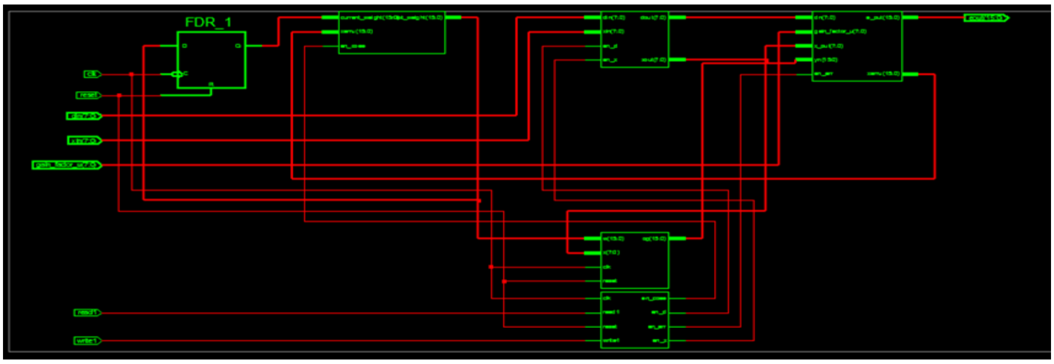

Figure 5.1 RTL schematic of LMS core block

There with the help of above five blocks we have successfully design the three equation of LMS algorithm.

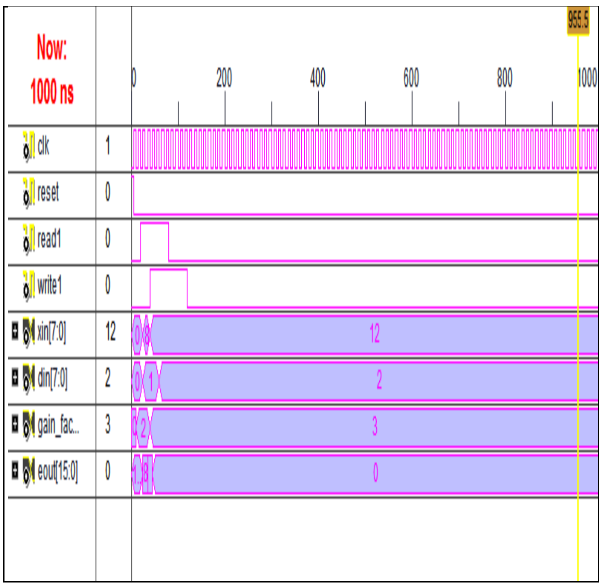

Figure 5.2 Simulation result forLMS core block

6. Comparison of adaptive algorithms

The performance of these adaptive algorithms is highly dependent on their filter order and signal condition. Figure show the Mean squared error (MSE) performance of the LMS algorithm and the RLS algorithm with different filter orders.

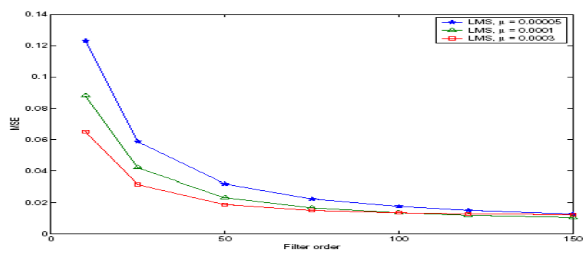

Figure 6.1 MSE performance of LMS algorithm with different filter orders

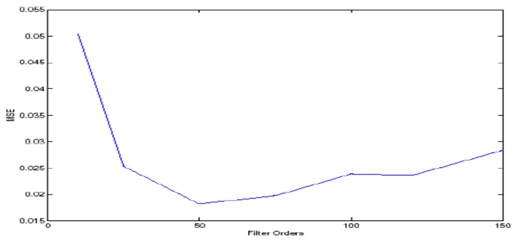

Figure 6.2 MSE performance of RLS algorithm with different filter orders

The LMS algorithm performs much better than the RLS algorithm in the high filter order region. As we increase the order of filter the MSE performance stabilizes. As the RLS is highly sensitive to numerical instability, the filter order will severely affect the performance of the algorithm. RLS performance does not improve when the filter order increases. Hence, a careful selection of the filter order is needed for optimal performance. Thus we are using LMS algorithm in our project.

7. CONCLUSION

This paper uses a hardware description language called VHDL to design LMS algorithm for adaptive noise canceller. A desired signal corrupted by additive noise can often be recovered by adaptive noise canceller using the Least Mean Square (LMS) algorithm. In this paper strategies & implementation of Least Mean Square algorithm for adaptive noise canceller is described. The Least Mean Square algorithm was found to be the most efficient training algorithm for FPGA based adaptive filters. In terms of the high speed architecture , the direct-form approach is preferred for design. This also contains the characteristics of LMS-based algorithms, the FPGA family, and the design methodology in the research. The overall aim is to choose an algorithm that will offer good tracking ability, particularly where the signals are time–varying. A adaptive filter is designed with a direct-form FIR filter coded in VHDL and with the LMS algorithm written in VHDL code executing on the Xilinx ISE 9.1.

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