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Least Mean Square (LMS) Adaptive Line Enhancer (ALE) Design for Modern Communication

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Abstract: The (LMS) Least Mean Square adaptive filter is a well behaved signal processing algorithm is used in applications where a system to adapt to its environment. Blocks of the system including the direct, transposed and hybrid forms are examined in terms of the following parameters:, power , speed consumption and FPGA resource usage. The hybrid and transposed forms, are derived from the delayed LMS, allow for high speeds without significant increases in area or power. Results for both these adaptations are independent of filter length with the maximum speed of the 16 tap transposed form being over 4 times higher than the speed of a 16 tap direct form implementation. For the FPGA implementation, the transposed form is optimal, as area and power are not significantly higher than values found for the direct form, despite the higher maximum frequency. The new one algorithm given good results even when noise is greater than signal which is considered a good achievement in the field. It eliminatess the drawbacks of LMS algorithm introduced by widrow and others. The (ALE) adaptive line enhancer design method updates the FIR filter coefficients in terms of the exact value of gradient and a variable step size. At high numbers of taps, the maximum frequency of the transposed form is not degraded, despite the input data paths driving an more no. of multipliers.

I. INTRODUCTION

A. Least Mean Square

The LMS algorithm which uses an instantaneous estimate of the gradient vector of a cost function is an approximation of the steepest descent algorithm. Based on sample values of the tap-input vector and an error signal the gradient is estimated. The algorithm iterates each coeffcient in the filter, moving it in the direction of the approximated gradient. For the LMS algorithm it is necessary to have a reference signal d[n] representing the desired filter output. The difference between the reference signal and the actual output of the transversal filter is the error signal .Considering as the step size(μ). The algorithm at each iteration requires that x(n),d(n) and w(n) are known. As the step size decreases, the convergence speed to the optimal values is slower. This also implies that, the LMS algorithm is a stochastic gradient algorithm if the input signal is a stochastic process.

B. Block LMS

In this technique the filter coefficients are held constant overeach block of the input signal. The filter output y(n) and errorsignal e(n) are calculated using filter coefficients of that block. Then, the filter coefficients are updated at the end of each block using an average of the L gradient estimates over that block.

C. Normalised LMS

The main drawback of the "pure" LMS algorithm is that it is sensitive to the scaling of its input. This makes it very hard to choose a learning rate μ that guarantees stability of the algorithm. The Normalised least mean squares (NLMS) filter is a variant of the LMS algorithm that solves this problem by normalising with the power of the input.

D. Adaptive Algorithms

Adaptive algorithms are used to update the weights, (coefficients) of the ADF and reduce the error signal, e(k), of the ADF according to some performance measures. These measures are convergence rate, minimum mean square error (MSE),computational complexity, stability, robustness, and filter length (number of weights).
There are many categories of adaptive-algorithms:

1) Least-Mean-Square (LMS).

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2) Recursive-Least-Squares (RLS) .
3) Kalman-Filter(KF).

The LMS algorithm is the most efficient algorithm in terms of memory storage and calculations. In addition, the LMS algorithm has been the least suffering in terms of the numerical stability problem when compared with the problem inherent in RLS,and KF algorithms. This makes LMS algorithm is one of the first choices in several applications. The RLS algorithm is the good in terms of convergence properties. The RLS algorithm was proposed in order to provide superior performance compared to those of the LMS algorithm and its variants, with few parameters to be predefined, especially in highly correlated environments. In the RLS algorithm, an estimate of

the autocorrelation matrix is used to de-correlate the current input data. Also, the quality of the steady-state solution keeps on improving over time, eventually leading to an optimal solution (Haykin, 2002).

II. ADAPTIVE LINE ENCHANCER

(ALE) Adaptive line enhancer used in many signal processing fields for its capability of tracking a signal of interest. The vector h(n) is the Mx1 column vector of filter coefficient at time k , in such a way that the output of signal, y(k) , is good estimate of the desired signal, d(k). The main advantage of it is that it does not require any reference signal to eliminate the noise signal. Fig. 1, show the adaptive filter setup, where s(k) , d(k) and e(k) are the input, the desired and the output error signals, respectively. This filter is an adaptive filter whose tap weights are controlled by an adaptive algorithm. Thus ALE refers to the case where a noisy signal, x(k),consisting of a sinusoidal component and the requirement is to remove the signal noise part of the signal. As a result, the predictor can only make a prediction about the sinusoidal component and when adapted to minimize the instantaneous squared error output, e (k), the line enhancer will be a filter optimize (the Wiener solution) or tuned to the sinusoidal component.

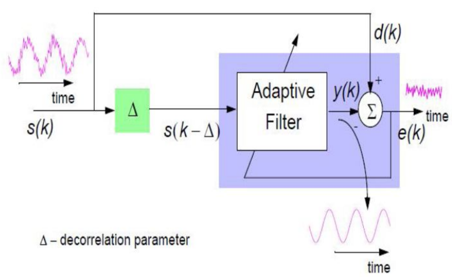

Figure.1-Basic Adaptive line enchancer

A. *Concept of Adaptive Filters as Adaptive Line Enhancer*

In this section it will be described how it is possible to get a narrowband signal (s(k)) which is contaminated by a wideband noise (n(k)). Usually, the n(k) has larger power than the power of s(k) which results in negative decibels (dB) of signal to noise ratio (SNR). The configuration of an ALE is shown in the Figure 2. The input signal of ALE is assumed to be b(k) = s(k)+n(k). The input signal of the ADF is the delayed version of b(k). The amount of delay must be chosen such that n(k) is de-correlated and s(k) is correlated. The ADF produces an estimate of s(k) which is denoted byˆ. The ADF is called linear prediction filter because it predicts the recent sample of s(k) from its previous samples and at the same time weakens the wideband signal. The error signal at the output of ALE can be written ask) = b(k)-y(k) = s(k) + n(k) -ˆ1.1 It is clear that error signal is a wideband noise and the output of the ADF is narrowband signal. Because of its ability to enhance sinusoidal signal in the presence of noise, this scheme is therefore called adaptive line enhancer.

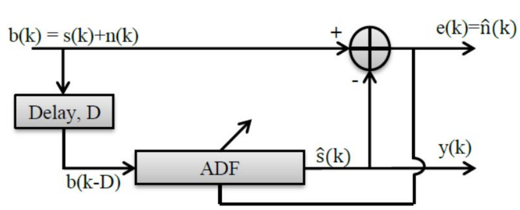

Figure 2: The Structure of the ALE

III. ADAPTIVE NOISE CANCELLATION

A. Basic Principles of Adaptive Noise Cancellation

Adaptive noise cancelation (ANC) is an alternative technique of estimating signals corrupted by additive noise. The ANC is greatly useful in many applications which are listed below.

B. Cancelling the Below Items

1) 60Hz Interference in Electrocardiography.
2) Donor ECG in Heart Transplant Electrocardiography
3) The Maternal ECG in Fetal Electrocardiograph.
4) Noise in Speech Signals
5) Antenna Sidelobe Interference.
6) Periodic-Interference without an External Reference Source.
7) Adaptive Self Tuning Filter.

The ANC configuration and its two input signals are b(k) and x(k) are shown in Figure 2.1. The signal b(k) is assumed to b(k) = s(k) + n(k). The signal s(k) is uncorrelated with x(k) and n(k) , and n(k) is correlated with x(k). The signal x(k) , which is a measure of the contaminating signal, is processed by the ADF to produce an estimateˆ. The estimate of signalˆcan be obtained by subtractingˆ from b(k) as follows.

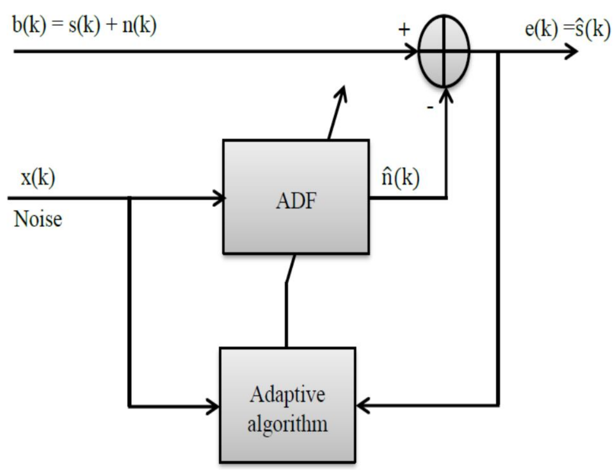

Figure 3: Adaptive Noise Canceller Scheme

IV. THE LEAST MEAN SQUARE ALGORITHM

The (LMS) least mean square algorithm is the simple, most powerful and famous adaptive algorithms which can be applied easily in real-time. The LMS algorithm which was developed by Widrow, trains its input correlation matrix and minimizes the MSE. The ratio of maximum to minimum eigen values has an important influence on the speed of convergence. If this ratio is small, it would speed up the rate of convergence, and if it is large, it would reduce the speed rate of convergence. The LMS algorithm makes use of the steepest descent algorithm in which the weights are updated by using the following equation Where is the weight, is the true gradient vector defined by , k is the kth sampling instant, and is the step size. It should be noted that the stability and the convergence rate depends on the value of step size. It is clear that in order to be able to compute, needs information about and as shown in equation.

V. IMPLEMENTATION

A variety of field programmable gate array architectures are available from various vendors. i evaluate our designs using the Xilinx spartan series as this provides good low power, high complexity performance suitable for data path manipulation. In spartan designs, the building block of each configurable logic block (CLB), which contributes to providing the functional elements for constructing logic within the FPGA, is the logic cell (LC). The arithmetic functions are to be implemented, dedicated carry logic provides fast arithmetic carry capabilities for high speed arithmetic functions.

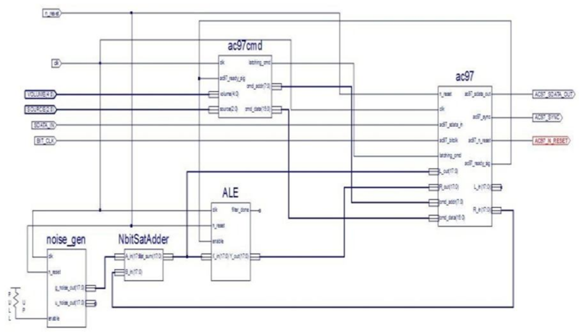

Figure 4: Implementation of the filter

Its highly regular structure makes it ideal for implementation on an FPGA. Without the presence of dedicated carry logic, it would be too slow due to the fact the output would have to ripple through the full *n* RCA's. With the spartan, losses in comparison with a much faster adder such as the carry look ahead adder (CLA) are minimal. This is also due to the highly up normal structure of the CLA which uses a generate term to create the carry. It requires 132 logic levels as opposed to logic levels in the case of a carry save adder. Implementing both these on a Spartan 3E XC3S500E, gives a maximum combinational path delay of 14.852ns for a 12 bit carry save adder as opposed to 15.153ns with a 12 bit carry look ahead adder.

VI. RESULTS

Our goal is to investigate the trade-off between these various possible architectures and compare them in terms of maximum speed, relative power and resource usage. Filters have been implemented using fixed point twos complement integer arithmetic. Input data was limited to integers in the range -5 to +5 ensuring no overflow. Verilog descriptions were synthesised to the target FPGA - a Virtex XCV1000E-8. Table 1 shows the achievable clock speeds for the de-sign. Maximum frequency remains relatively constant for the transposed form over 4, 8 and 16 taps, with a significant increase in speed over the direct form.

The critical path is only one adder and one multiplier, independent of the number of taps. In ASIC design as the number of taps is increased in the transposed form, the capacitance of the input bus would limit its speed, and the advantages of the hybrid form, would become more evident. In contrast the performance of the direct form implementation falls from almost 23MHz for a 4 tap filter to 11MHz for a 16 tap filter due to the extension in critical path.

The speed in the transposed form is achieved at a slight expense of area, which can be seen in Table 2. Table 3 indicates the estimated power for the designs considered. Their power was estimated using the Xilinx Virtex Power estimator Worksheet V.1.5. However these figures represent estimated data path power only and do not take into account interconnect power which is usually far more significant.

VII. CONCLUSION

In this project alternative implementations of the delayed LMS adaptive algorithm are investigated. As expected, CLB power for 16 taps Direct Transpose Hybrid CLB power 46mW 48mW 47mW the direct form implementation is slowest, with its maximum frequency falling as the critical path increases. In contrast the maximum frequency of the transposed and hybrid forms are approximately independent of the number of taps due to the fact their critical paths are restricted to a fixed number of multipliers and adders.The potential fan in capacitance limited speed of the transposed form, is not a factor for filters of the size we have investigated. The hybrid form performs quite well in terms of power consumption and area, with its speed remaining relatively constant over the full range of taps. Its modular structure gives it a distinct advantage in terms of speed over the direct form structure.

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