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Implementation of Adaptive Filter using FPGA

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Abstract: This paper presents a design and implementation of least mean square (LMS) adaptive filter for noise cancellation and signal enhancement application using Field Programmable Gate Array. Audio signal quality is enhanced by removing the noise from the signal. Adaptive filter is used for cancellation of noise from the signal that is undesired. Adaptive filters the LMS algorithm is perform very well in terms of the number of iterations required for convergence. FPGA implementation of adaptive filter algorithm is studied in this project work.

Keywords: Adaptive filter; Least Mean Square algorithm; FPGA; Noise cancellation.

I. INTRODUCTION

Many digital signal processing applications requires linear filters and adaptive techniques in signal processing and analysis. Adaptive filters are used in various real-time applications such as system identification, prediction, echo cancellation and noise cancellation,. Field -programmable gate arrays (FPGAs) are also most widely used for applications where timing requirements are strict. Thus implementation of filter in real-time is needed. Speech enhancement aims to improve speech quality by using various algorithms. Noise reduction is a key-point of speech enhancement systems.

Adaptive filters are digital filters. They have capacity of self adjustment. Adaptive filters are used when noise occurs in same band as the signal, when noise band is unknown or signal varies over time. There are many types of adaptive filters employing different schemes to adjust the filter weights based on many different criteria. The distorted or degraded audio signals are called noisy speech signals. The presence of background noise in speech significantly reduces the intelligibility of speech. Noise reduction algorithms are used to suppress these background noises and also to improve the quality and intelligibility of audio signal. Removing various types of noise is difficult due to the random nature of the noise. An adaptive filter is a computational device that iteratively models the relationship between the input signal and output signal of a filter. An adaptive filter self-adjusts the filter coefficients according to an adaptive algorithm. Coefficients/weights are calculated in loop form. Using these coefficients real input signal is updated by removing noise.

FPGA based signal processing is based on hardware logic. In FPGA, applications can run in parallel; filtering, correlation and much other application can run simultaneously. Because of this it require less time. And used in many applications.

II. DESIGN OF ADAPTIVE FILTER

Adaptive filters are usually used for information processing in variable and noisy environments. By using Adaptive noise cancellation Signal to Noise Ratio (SNR) is improved by removing noise from the received signal. Here two input signals are applied simultaneously to the adaptive filter; real input and desired input. It automatically adjusts the weights of the filter with respect to error calculated.

A. LMS Filter

The coefficient/weight wn is calculated from the LMS algorithm,

$$\mathrm{Wn+1 = Wn + m * e(n)x(n)}$$

In the above equation,

Wn= Currently used coefficient

Wn+1= Coefficient obtained from LMS algorithm

m= Speed control of coefficient change

e(n)= Error value updated

x(n)= Noise signal coefficient

$$\mathrm{e(n) = d(n) - y(n)}$$

LMS algorithm determines amount of corrections which are applied to filter by using step size. This is adapted from one iteration to the next. A too small step size increases the time for the filter to converge on a set of coefficients or too large causes the adaptive filter to diverge. Here the resulting filter might not be stable. So use of smaller step size is better for convergence.

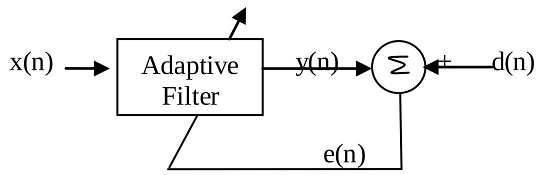

Fig1: Adaptive Filter Block diagram

In Fig1. x(n) is a real input signal with noise which is applied to Adaptive filter and d(n) is desired signal. Error signal is the difference of filter output and desired signal. This filter output is calculated by using updated weights.

An error difference e(n) is minimized by adjusting weights of filter using adaptive algorithm iteratively. The LMS algorithm is an adaptive algorithm among others which adjusts the coefficients of FIR filters iteratively.

The main idea behind LMS filter is to approach the optimized filter weights. It is done by updating the filter weights in a manner to converge to the optimum filter weight. Initially algorithm starts by assuming small weights as zero (in most cases) and, at each iteration, gradient of mean square error and weights are updated.

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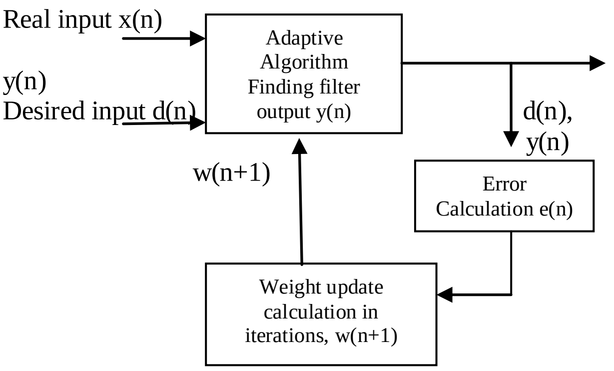

Fig2: Architecture of filter

If the gradient is positive, that means if the same weight is used for further iterations the error would keep increasing positively. So we need to reduce the weights. If the gradient is negative, then we need to increase the weights.
In the filtering process, first the Low pass filter removes the corrupted low frequency noises in signal. Now d(n) is compared with y(n) which produces e(n). These error coefficients are fed back to each LMS algorithms to update the coefficients. Then these steps are repeated up to error becomes negligible. The updated coefficients of LMS Algorithm is the Response of desired Filter

B. Implementation
Field Programmable Gate Arrays (FPGAs) has feature of reprogramming as many times to achieve the desired result. The fabrication process of FPGA can be quite expensive and very time consuming. FPGA gives more design flexibility, and reducing a developing time and cost. If in case design fails after testing on FPGA, then designer can simply redevelop the design and dump it again to the FPGA. Use of an FPGA would thus eliminate the loss in development time caused by a faulty initial design. This also gives the designer knowledge of whether or not the design works.
Here we have used System Generator software for implementation of the project.
System generator has advantage of,
i. Fast prototyping
ii. Fast fixed point Implementation
iii. Optimized design
iv. Model view better understanding
v. Easy to modify

III. RESULT

A. Simulation

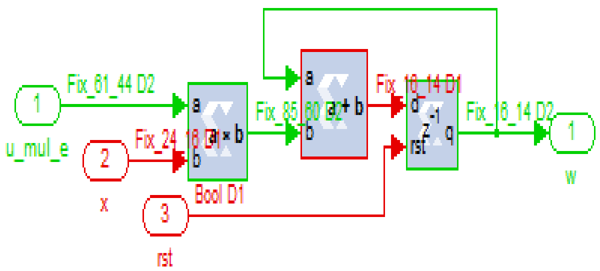

Fig3. Simulation of weight update

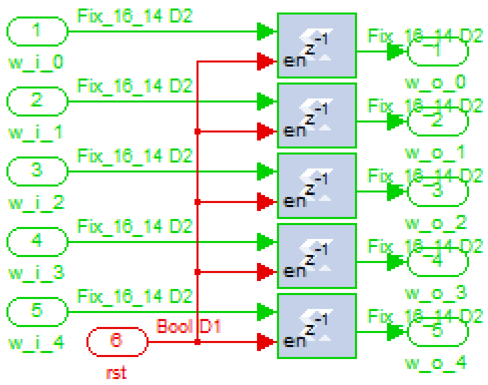

Fig4. Simulation of final weight

B. Output

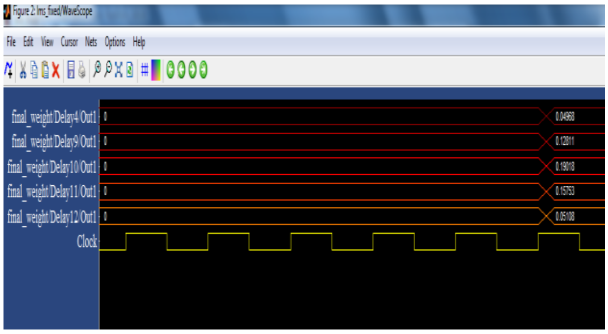
Fig5. Simulation output of final weight

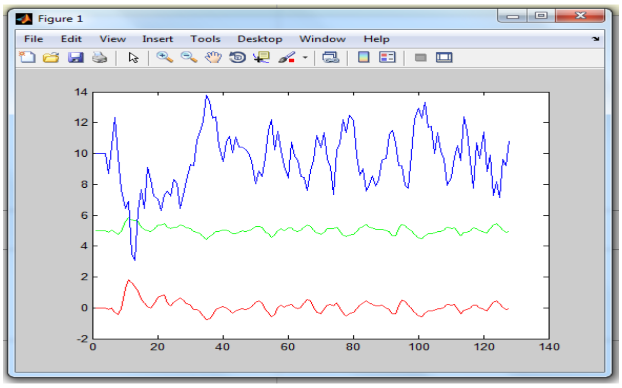

Fig6. Response of signals

From the Fig7, First signal in blue color is the real signal which is containing noise components. This signal is given as an input to the adaptive filter. Second signal in green color is the desired signal which is provided at the input for comparing with the real signal. And last signal in red color is the filtered output of the adaptive filter after applying algorithm.

IV. CONCLUSION

In signal processing system FPGA is best choice due to their greater flexibility and higher bandwidth, parallel architecture. The performance of this algorithm is evaluated in terms of various analytical parameters. From the results obtained, it is concluded that the algorithm reduces the noise from the real signal and desired signal is obtained.

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