IOSR Journal of Electronics and Communication Engineering (IOSR-JECE)
e-ISSN: 2278-2834,p- ISSN: 2278-8735.Volume 8, Issue 2 (Nov. - Dec. 2013), PP 58-63
www.iosrjournals.org

Area Optimized Adaptive Noise Cancellation System Using FPGA for Ultrasonic NDE Applications

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Abstract: *This paper describes a method for noise removal from ultrasonic signals used in NDE (Non Destructive Evaluation) applications. The method used is FPGA based adaptive noise cancellation. Indeed adaptive filtering is one of the core technologies in digital signal processing. We can find numerous application areas in science as well as in industry. Adaptive filtering techniques are used in a wide range of applications, including system identification, adaptive equalization, adaptive noise cancellation and echo cancellation. Characteristics of signals that are generated by systems of the above application are not known a priori. Under this condition, a significant improvement in performance can be achieved by using adaptive rather than fixed filters. An adaptive filter is a self-designing filter that uses a recursive algorithm (known as adaptation algorithm or adaptive filtering algorithm) to design itself. Here the algorithm used is normalized least mean square algorithm, which has good convergence and stability.*
Keywords*; Adaptive noise cancellation, NLMS algorithm, NDE applications, ultrasonic grain noise.*

I. Introduction

A filter that adjusts its transfer function according to an optimization algorithm is called an adaptive filter. This optimization algorithm will be driven by an error signal. Since most of the optimization algorithms are highly complex majority of the adaptive filters are digital filters. On the other hand non adaptive filters will be having static transfer functions. Adaptive filters are well suited in operations where some of the process parameters are not known in advance. An error signal is fed in to adaptive filter as feedback signal to adjust the transfer function according to the changing parameters. Different applications like interference cancellation, predication, inverse modeling uses adaptive filter. This project describes field programmable gate array based adaptive noise cancellation for adaptive filtering in ultrasonic NDE applications. Backscattered noise from microstructures inside material can be efficiently reduced by adaptive filter.

In radar, sonar, medical ultrasound and industrial ultrasonic applications the detected signal always contains backscattered echoes from ground, sea, scatters or microstructures inside material [1-4]. There echoes are high overlapped and greatly dependent on the complex physical properties of the propagation path. It is challenging to extract valuable information from detected signals especially in the applications of target localization, flaw detection, object recognition, etc. In ultrasonic nondestructive evaluation (NDE), the backscattered noise is primarily from crystallites (i.e., grains). They have irregular boundaries, size and random orientation. The signal representation of flaw or defect may be not identified due to the high density scattering grains. Much research effort has been made to suppress grain noise and process ultrasonic data. Split spectrum processing (SSP),discrete wavelet transform (DWT), discrete Hadamard transform (DHT), discrete cosine transform (DCT), and chirplet transform have been utilized for ultrasonic signal processing [5-11].

It can be seen that a signal processing technique adaptive to ultrasonic data is highly sought-after. Adaptive filter is widely used in different applications for interference cancellation, prediction, inverse modeling, and identification of Least mean square (LMS) and Recursive Least Square (RLS) are commonly adopted to implement adaptive filters [16-17]. In [18], normalized LMS is explored for ultrasonic backscattered signal. In [19-20], ultrasonic NDE signals and medical ultrasound images are deconvoluted using adaptive filter. This project uses Normalized least mean square (NLMS) algorithm to implement adaptive filtering. The Normalized least mean square algorithm has a good convergence and also it is having a good stability. So NLMS algorithm is best suited in this scenario.

II. Adaptive Filtering For Ultrasonic Signal

A. General Block Diagram

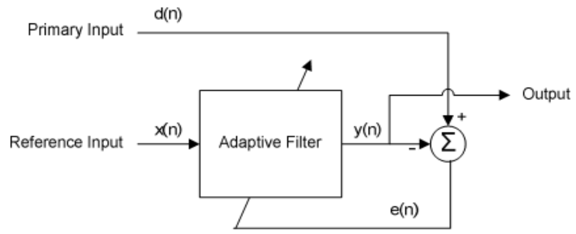

Fig.1.General block diagram of adaptive filtering system

General Block diagram of adaptive filtering system is shown in Fig 1 where d(n) denotes a primary input of the system, x(n) the adaptive filter input signal, y(n) the adaptive filter output signal, and the error signal, e(n) is the difference between d (n) and y(n).

B. Adaptive Filter Operation

The filter output, y(n), can be written as,

$y(n)=\sum_{i=0}^{L-1} w(n)x(n-i)=W^T(n)X(n)$ - (1)

where the vector

$W(n)=[\, w_0(n), w_1(n)\,, w_2(n)], \dots w_{l-1}(n)]^T$ - (2)

denotes the coefficients of the time varying L tap adaptive filter.

$X(n)=[x(n),x(n-1)],x(n-2)],\dots,x(n-L+1)]^T$ - (3)

denotes the filter input vector and $[.]^T$ denotes the transpose operation.

The filter coefficients, W(n), are adjusted to minimize the mean square error function of the system, J(n), given by

$J(n) = E[e^2(n)]$ - (4)

We have to set the partial derivative of J(n) with respect to the filter coefficients, W(n) to zero.

- (5)

The solution of equation (4) can be written as

$W_{opt}(n) = R_{xx}^{-1}(n)\, r_{dX}(n)$ - (6)

The computation cost of equation (6) is enormous due to the matrix inversion calculation for every new data sample. Therefore, an iterative strategy is commonly used to update the filter coefficients, W (n) .

III. Normalised Least Mean Square Algorithm

In many adaptive filter algorithms Normalized least mean square algorithm (NLMS) is used as the adaptation algorithm. The objective of the alternative LMS-based algorithms is either to reduce computational complexity or convergence time. The normalized LMS, (NLMS), algorithm utilizes a variable convergence factor that minimizes the instantaneous error. Such a convergence factor usually reduces the convergence time but increases the misadjustment. In order to improve the convergence rate the updating equation of the conventional LMS algorithm can be employed variable convergence factor.

Apparently, the convergence rate of the NLMS algorithm is directly proportional to the NLMS adaptation constant, i.e. the NLMS algorithm is independent of the input signal power. So by choosing appropriate values the convergence rates of the algorithms can be optimized. The NLMS algorithm converges more quickly than the LMS algorithm.

A. Advantages of NLMS algorithm

The Main advantages of NLMS adaptive filter are robust behavior when implemented in finite-precision hardware, well understood convergence behavior and computational simplicity for most situations, which make it as different from other adaptive algorithms. The simple well behaved NLMS adaptive filter algorithm is commonly used in applications where a system has to adapt to its environment. Architectures are examined in terms of the following criteria: speed, power consumption and FPGA resource usage. Modern FPGAs contain many resources that support DSP applications. These resources are implemented in the FPGA fabric and optimized for high performance and low power consumption.

B. Overview of the Algorithm

NLMS algorithm can be used to implement a very efficient adaptive filter. The filter coefficients vector is updated as [17-18].

$W(n+1) = W(n) + \eta\, e(n)\, X(n)$ - (7)

Here η is the learning factor to whose purpose is to control the convergence rate of the filter and is updated as,

$\eta = 2/(\partial + p(n))$ - (8)

Where δ is a small value to limit the learning rate and p (n) is the estimated signal power

The primary input, d(n) , and the reference input, x (n) of the adaptive filter can be defined as

$d(n) = s(n) + g_1(n)$ - (9)

$x(n) = g_2(n)$ - (10)

That means d(n) is the signal corrupted by a noise, $g_1(n)$ and $g_2(n)$ is a reference noise input which is always available to the system.

C. Input and Output of NLMS Algorithm

The above block diagram (figure 2) shows the inputs and outputs of the NLMS algorithm. It has four inputs and two outputs. Inputs are x_in (input data to adaptive filter),d_in (desired input), clk(clock) and adpt_enable. Outputs are error_out (difference between output of filter(y-out) and desired input (d_in) and final out.

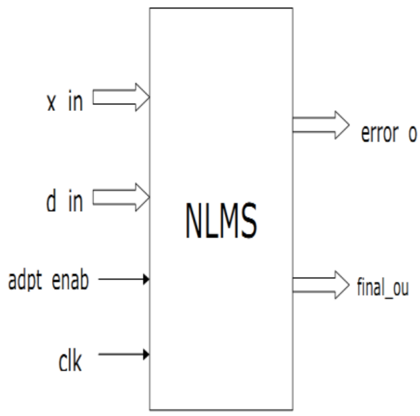

Fig.2. input and output of NLMS algorithm

IV. Design And Simulation Of Fpga Based Adaptive Noise Cancellation System.

Field Programmable Gate Array (FPGA) has been widely used in application such as communication, industrial automation, motor control, medical imaging etc. It is reconfigurable in term of digital logic, input/output block, and routing resource. In addition, algorithms running on FPGA could have a real-time performance with signal processing in parallel on hardware. Xilinx system generator is used to implement and test the adaptive noise cancellation system. In the design process first a standard FIR filer (6 taps) is designed using system generator. Then it is used as a subsystem to develop the entire adaptive noise cancellation system.

A. Standard 6-Tap FIR Filter

The output of the filter is verified using a sine wave which is added with another sine wave. The latter signal is considered as the noise. The Standard 6- Tap FIR Filter is simulated and results are shown in Fig 3, 4a, 4b and 4c.

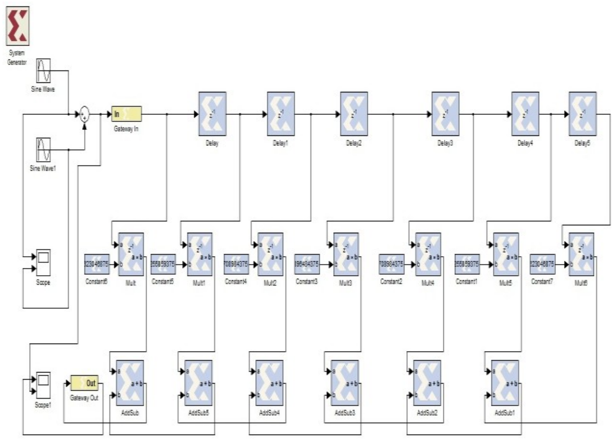

Fig.3. Standard 6-tap FIR filter

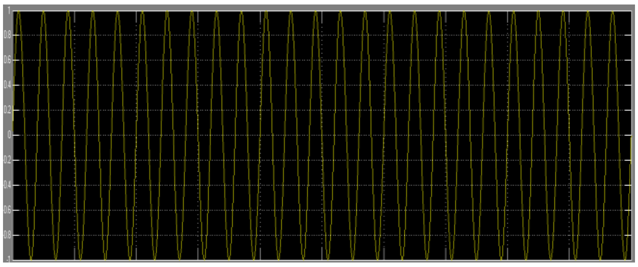

Fig.4a - input signal

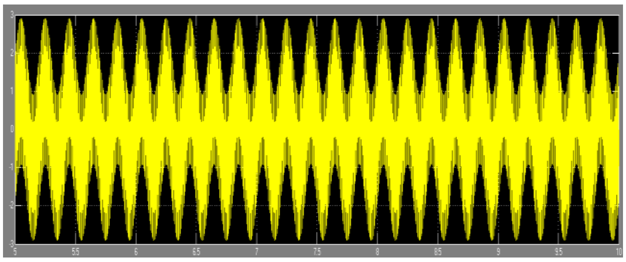

Fig 4b. Corrupted signal

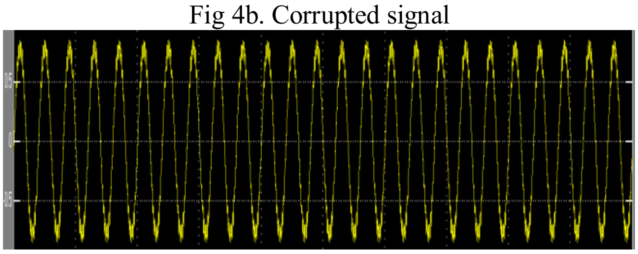

Fig 4c. FIR filter output

B. Adaptive Noise Cancellation System

The adaptive noise cancellation system is developed in Xilinx system generator using the above fir filter as a subsystem. This design is very much area efficient than the other existing systems. The filter coefficient updation part is designed according to the NLMS algorithm explained above. The coefficients are updated using equation (7). The block diagram is given below.

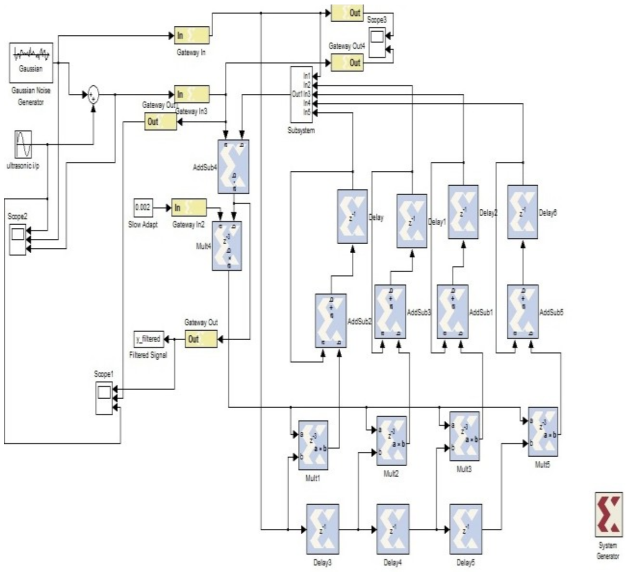

Fig.5. adaptive noise cancellation system

The system is tested by giving an ultrasonic signal corrupted by additive white Gaussian noise. The original signal, noise and the system output are given below.

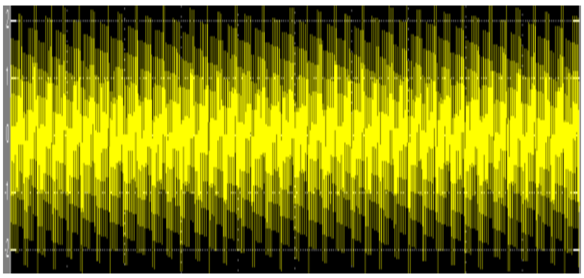

Fig.6a - input ultrasonic signal

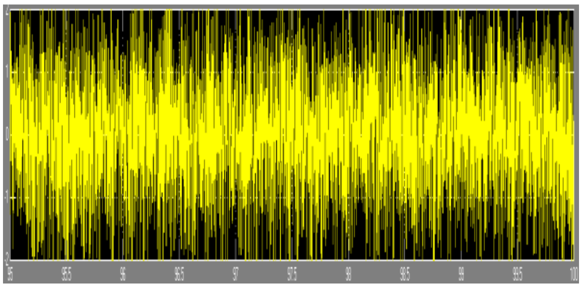

Fig 6b - input corrupted with AWG noise

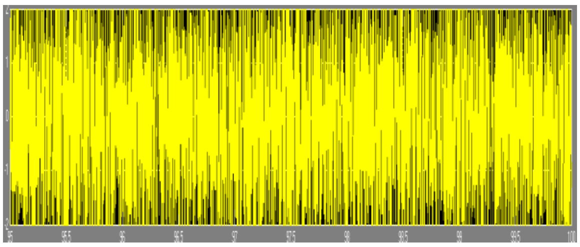

Fig 6c - adaptive noise cancellation system output

V. Conclusion

FPGA based adaptive noise cancellation system is an efficient method for removing grain noise from ultrasonic signals and restoring original signal. This fact is proved from the above simulation results. Further Xilinx system generator can generate the HDL code for different FPGA's. The above method is concentrating on ultrasonic signals used in NDE applications, so timing constraints won't be a problem. The choice of NLMS algorithm is proved to be a good one from the above simulation results.

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