RESEARCH ARTICLE OPEN ACCESS

An Adaptive Filtering System Configurations and Architecture on Reconfigurable Platform

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ABSTRACT
This paper proposed the implementation of adaptive filters on reconfigurable platform. Adaptive filter is an essential part of digital signal systems, have been widely used and its implementation takes a great deal, there is no dedicated IC for adaptive filter. When FPGA implemented in such area, provides a lot of facilities to the designers and also offer a better solution for filtering the data. Adaptive signal processing evolved from the techniques developed to enable the adaptive control of time- varying systems. Filtering data in real-time requires dedicated hardware to meet demanding time requirements and provide the highest processing performance, but is inflexible for changes. When a design demands the use of a DSP, design adaptability is crucial, then FPGA may offer a better solution. Reconfigurable hardware devices offer both the flexibility of computer software, and the ability to construct custom high performance computing circuits. Adaptive filter implemented using Field Programmable Gate Arrays (FPGAs) due to some of their attractive advantages include flexibility and programmability, availability of tens to hundreds of hardware multipliers available on a chip.
Keywords - Adaptive Filter, Adaptive Filter Architecture, Adaptive LMS Algorithm, Adaptive Filter on reconfigurable platform, Architecture of Adaptive Filter on FPGA

I. INTRODUCTION

Adaptive filters on the other hand, may be designed to adjust to their environment such that the filter may adapt to noise so as to produce the desired result. Adaptive filters learn the statistics of their operating environment and continually adjust their parameters accordingly. The adaptive filter architecture is taken a Gaussian noise, but in many practical real situation it is seen that, using Gaussian noise is not sufficient, because Gaussian noise has a fixed shape “Bell curve “ but many situation can’t predict the noise shape, So impulsive noise consideration is very important. Impulsive noise generally has no fixed shape it varying large amplitude spikes which is overlapped so many samples. So it is very difficult to detect or cancel it. The impulsive noise is generally destructive types of signal distortion. So in this paper impulsive noise reduction is also main aim of implementation. In all adaptive filter architecture try to minimize error i.e. minimization of difference between the desired output and the real one for all the input vectors. Now a days, the use of Field programmable gate arrays (FPGAs) is growing.

An adaptive filter is essentially a digital filter with self-adjusting characteristics. It adapts, automatically, to change in its input signals. Adaptive filters are the central topic in the sub-area of DSP known as adaptive signal processing. This paper describes key aspects of this important topic based on the LMS (least mean square) and RSL (recursive least squares) algorithms which are two of the most widely used algorithms in adaptive signal processing. The contamination of a signal of interest by other unwanted, often larger signals or noise is a problem often encountered in many applications. Where the signal and noise occupy fixed and separate frequency bands, conventional linear filters with fixed coefficients are normally used to extract the signal. However, there are many instances when it is necessary for the filter characteristics to be variable, adapted to changing signal characteristics, or to be altered intelligently. In such cases, the coefficients of the filter must vary and cannot be specified in advance. [8] E.C. Ifeachor and B. W. Jervis. Where there is a spectral overlap between the signal and noise or if the band occupied by the noise unknown or varies with time. An adaptive filter has the property that its frequency response is adjustable or modifiable automatically to improve its performance in accordance with some criterion, of their self adjusting performance and in-built flexibility. In speech processing noise cancellation and echo cancellation are very much important. In this paper shows the architecture of adaptive filter which is very applicable in the above mentioned applications.

II. ADAPTIVE FILTER

An adaptive filter is a computational device that attempts to model the relationship between two signals in real time in an iterative manner. The fundamental operations of an adaptive filter can be characterized independently of the specific physical realization that it takes. [6] Douglas, S.C.

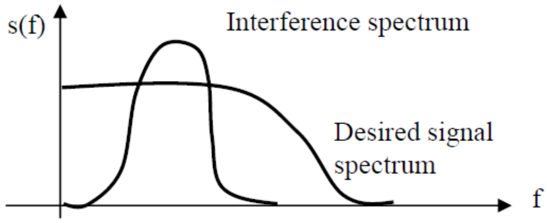

Fig.1 Spectral overalap between signal and noise

In summary we use adaptive filters:

1 When it is necessary for the filter characteristics to be variable, adapted to changing conditions;
2. When there is spectral overlap between the signal and noise.
3. İf the band occupied by the noise is unknown or varies with time.

The use of conventional filters in the above cases would lead to unacceptable distortion of the desired signal. There are many other situations, apart from noise reduction, when the use of adaptive filters is appropriate.

2.1 System Architecture

A low cost and high performance programmable digital finite impulse response (FIR) filter follows the adaptive algorithm used for the development of the system. The architecture employs the computation sharing algorithm to reduce the computation complexity. The algorithm used to update the filter coefficient is the Least Mean Square (LMS) algorithm which is known for its simplification, low computational complexity, and better performance in different running environments [2] Lan-Da Van,Wu-Shiung Feng.

Fig.2. shows a block diagram of how an adaptive filter can be formulated in an equalizer setting. In this case, the linear filter having weight *w* is adapting to produce an output sequence which is identical to a known output d[n]. *T*he linear filter as being of FIR type, with *p* coefficients, The adaptation algorithm calculates the update based on knowledge of the input, and on an error signal e. Because of the ability of adaptive filter to perform well in unknown environments and track statistical time variations, adaptive filters have been employed in a wide range of fields.

In most adaptive systems, the digital filter is realized using a transversal or finite impulse response (FIR) structure. Other forms are sometimes used, because of its simplicity and guaranteed stability.

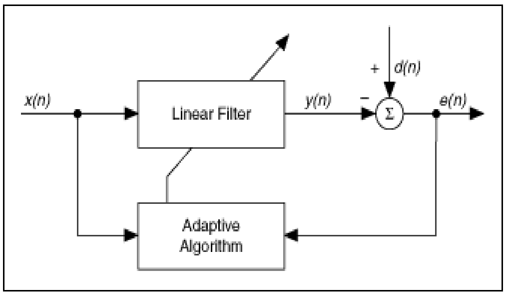

Fig.2 Adaptive Filter Block

For the *N* – point filter, the output is given by,

$$y_k = \sum_{i=0}^{N-1} W_k(i) X_{k-i}$$

where $w(i), i = 0,1,......., k$ are the adjustable filter coefficients (or weights), and $x_k(i)$ and y_k are the input and output of the filter. Figure 2 illustrates the signal-input single-output system. In a multiple-input single-output system, the x_k may be simultaneous inputs from *N* different signal sources.

2.2 Adaptive Filtering System Configurations

There are four major types of adaptive filtering configurations; adaptive system identification, adaptive noise cancellation, adaptive linear prediction, and adaptive inverse system. All of the above systems are similar in the implementation of the algorithm, but different in system configuration. All four systems have the same general parts; an input x(n), a desired result d(n), an output y(n), an adaptive transfer function w(n), and an error signal e(n) which is the difference between the desired output u(n) and the actual output y(n). In addition to these parts, the system identification and the inverse system configurations have an unknown linear system u(n) that can receive an input and give a linear output to the given input shown in the Fig. 4.

Fig. 3 shows the transversal filter with the time dependent component here the filter coefficient are variable and are adapted by an adaptive algorithm. In vector form x[n], x[n-1], x[n-2],…., x[n-N+1], the tap input vector at time n.
$c_0[n]$, $c_1[n]$, $c_2[n-2]$, ….., $c_{N-1}[n]$, the coefficient vector at time n.

The key aim of the adaptive filter is to minimize the error signal e_k . The success of this minimization will clearly depends on the nature of input signals, length of adaptive filter, and the implementation of adaptive algorithm .

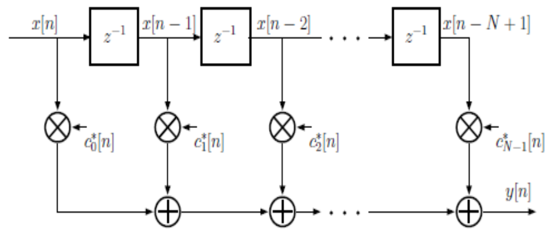

Fig. 3 Transversal filter with time dependent component

In a transversal filter of length N, as depicted in fig. 3, at each time n the output sample y[n] is computed by a weighted sum of the current and delayed input samples x[n], x[n − 1], . . .

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.

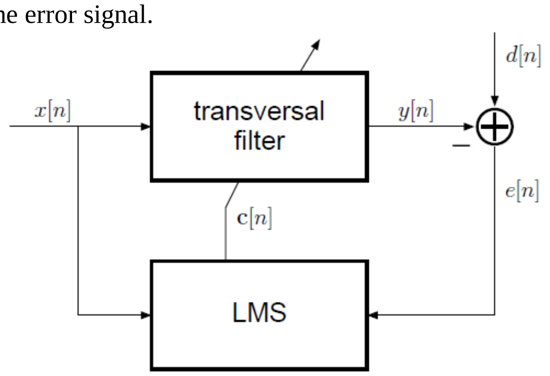

Fig. 4 Adaptive filter system using LMS algorithm

The adaptive filter could be FIR (non-recursive), IIR (recursive) or even a non-linear filter. Most adaptive filters are FIR for reasons of algorithm stability and mathematical tractability. In the last few years however, adaptive IIR filters have become increasingly used in stable forms and in a number of real world applications (notably active noise control, and ADPCM techniques). Current research has highlighted a number of useful non-linear adaptive filters such as Volterra filters, and some forms of simple artificial neural networks.

2.3 ADAPTIVE ALGORITHM

Adaptive algorithms are used to adjust the coefficients of the digital filter such that the error signal *e* is minimized according to some criterion. For example in the least squares sense. Common algorithms that have found widespread application are the least mean square (LMS) , the recursive least squares (RLS), and the Kalman filter algorithms. In terms of computation and storage requirements, the LMS algorithm is the most efficient. Further, it does not suffer from the numerical instability problem inherent in the other algorithms. For these reasons, the LMS has wide spread.

2.4. LMS ADAPTIVE ALGORITHM

LMS algorithm is a stochastic gradient algorithm. Steepest decent uses a deterministic gradient in recursive combinations. LMS does not require measurement of correlation and cross correlation function and matrix inversion.
The LMS algorithm is based on the steepest descent algorithm.

$$W_{k+1} = W_k - \mu \nabla_k$$

where *Wk* is the filter weight vector at the *k*th sampling instant, μ controls stability and rate of convergence and ∇_K is the true gradient of the error-performance surface, derive the windrow-Hopf LMS algorithm for adaptive noise cancelling, stating any reasonable assumptions made. The gradient vector, ∇, the cross-correlation between the primary and secondary inputs, **P**, and the auto correlation of the primary input, **R**, are related as

$$\nabla = -2P + 2RW$$

In the LMS algorithm, instantaneous estimates are used for ∇. Thus

$$\begin{aligned} \nabla &= -2P_k + 2R_k W_k \\ &= -2P_k + 2R_k W_k \\ &= -2X_k Y_k + 2X_k X_k^T W_k \\ &= -2X_k(Y_k - X_k^T W_k) \\ &= -2e_k X_k \end{aligned}$$

Where,

$$e_k = y_k - X_k^T W_k$$

From the steepest descent algorithm we have the basic Windrow-Holf LMS algorithm is given as

$$W_{k+1} = W_k + 2\mu e_k X_K$$

where,

$$e_k = y_k - W_k^T X_k$$

The Weight Adaptation is given as:

$$W_{k+1} = W_k + 2\mu X_k e_k$$

The key feature of the LMS algorithm is its simplicity.
Where, $W_k(i)$ - Vector of updated filter coefficient
$X_k(i)$ – Vector of latest input samples
e_k – error sample
y_k – contaminated sample.
instead of calculating the gradient at every time step in steepest-descent, the LMS algorithm uses a rough approximation to the gradient. The error at the output of the filter can be expressed as which is simply the desired output minus the actual filter output and can be expressed as,

$$e_k = y_k - W_k(i) * X_k(i)$$

which is simply the desired output minus the actual filter output Using this definition for the error an approximation of the gradient is found by,

$$\hat{\nabla} = -2e_{\mathbf{k}}\, \mathbf{x}_{\mathbf{k}}$$

Substituting this expression for the gradient into the weight update equation from the method of steepest-descent gives

$$W_k(i+1) = W_k(i) + 2\mu\, e_k\, X_k(i)$$

$$0 < \mu < \frac{2}{MS_{\max}},$$

which is the Widrow-Hoff LMS algorithm.

where M is the number of filter taps and Smax is the maximum value of the power spectral density of the tap inputs X_K. The relatively good performance of the LMS algorithm given its simplicity has caused it to be the most widely implemented in practice. For an N-tap filter, the number of operations has been reduced to 2*N multiplications and N additions per coefficient update. When the filter taps are increased, this improves the convergent performance of LMS algorithm, but every tap (in structure of LMS adaptive filter) costs two more multipliers and two more adders however, this will increase the area needed and decrease the maximum frequency of the design. So, balance is required between the convergent performance and the amount of hardware used effectively. The high-speed capability and register rich architecture of the FPGA is ideal for implementing LMS. This is suitable for real-time applications, and is the reason for the popularity of the LMS algorithm.

III. Systolic Array Architecture

This adaptive digital filter can be design on reconfigurable FPGA platform using Systolic array architecture, this paper shows the implementation of given filter architecture in systolic array form, which reduce the circuit scale into half without impairing the processing speed. The design digital FIR filter architecture employs the computation sharing algorithm to reduce the computation complexity. Systolic arrays represent a bottle neck architecture in which the adaptive algorithm calculations are performed in parallel. Implementing such adaptive filter architecture by arranging circuits (cells) that perform individual, calculations in a regular pattern and the data required for the calculation are feed to such cell in a pipelined manner. This parallel approach can improve the processing speed to a large extent and the uniform structure provides excellent expandability. Moreover, since the majority of the cells are only connected directly to adjacent cells and the data exchange is local, they are highly suited for implementation using Large Scale Integration circuits (LSI). Figure 5 shows the architecture used for implementing the RLS or LMS algorithm using systolic arrays, shows a case with four array elements. x1(i), x2(i), x3(i) and x4(i) represent the ith sample point of the signals of each element and y(i) represents the ith sample point of the reference signal used for identifying the desired signal. The systolic arrays are mainly configured using two types of cells, boundary cells and

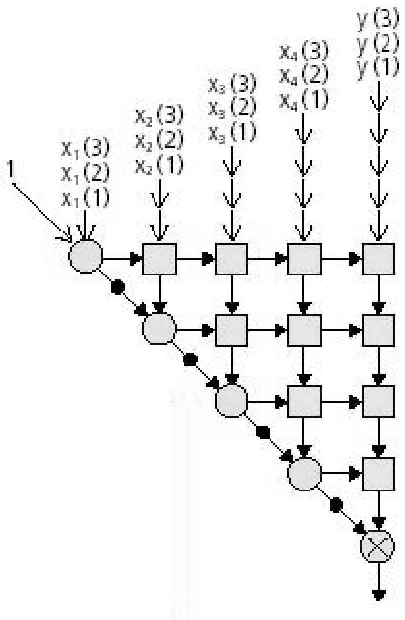

Fig. 5 Systolic array architecture

internal cells. All cells operate under the control of a single clock, and the data propagates through the cell arrays in synchronization with this clock, each data point is a complex number, and two multiplications of complex numbers are included in the internal cell processing.

IV. CONCLUSION

In this Paper an efficient N-Tap systolic Adaptive Filter can be implemented on reconfigurable platform i.e. Field Programmable Gate Array (FPGA). The high processing speed and low computational complexity adaptive filter architecture ideal for implementing adaptive LMS algorithm. This paper has been to development of algorithms and systolic architectures for proposed adaptive filtering system architecture.

V. ACKNOWLEDGEMENTS

We would like to acknowledge the Faculties of Electronics & Telecommunication Department, Sipna College of Engineering & Technology, Amravati for their support. I, Dipen B. Patel specially want to thank my guide Prof. Pritesh R. Gumble for their guidance and constant encouragement towards the work.

REFERENCES

Journal Papers

[1] M.S. Sutaoane, M.B. Mali and S.R. Deo, "VLSI Implementation Of Adaptive Filter On Reconfigurable Platform" *SPIT-IEEE Colloquium and International Conference*.

[2] Lan-Da Van,Wu-Shiung Feng on "An Efficient Systolic Architecture for the DLMS Adaptive Filter and Its Applications"*IEEE Transactions On Circuits & Systems.*

[3] Lei Wang and Naresh R. Shanbhag, "Low-Power Filtering via Adaptive Error-Cancellation" *IEEE Transactions On Signal Processing.*

[4] T. Rissa, R. Uusikartano, J. Niittylahti, "Adaptive FIR filter architectures for run-time reconfigurable FPGAs," *IEEE International Conference on Field-Programmable Technology*.

[5] Ankit Jairath, Sunil Kumar Shah, Amit jain, "Design & implementation of FPGA based digital filters" *IJARCET.*

Books

[6] Douglas, S.C. "Introduction to adaptive filters", *Digital signal processing handbook*.

[7] S. Haykin, Adaptive Filter Theory. *Upper Saddle River, NJ: Prentice Hall, 1996*.

[8] E.C.Ifeachor and B. W. Jervis, Digital Signal Processing, *A Practical Approach, Prentice Hall, 2002*.

[9] Paulo S. R. Diniz "Adaptive Filtering Algorithms and Practical Implementation" *Third Edition Kluwer Academic Publishers.*

[10] Vinay K. Ingle, John G. Proakis, "Digital Signal Processing using MATLAB" *The PWS publisher.*