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REVIEW ON DESIGN OF EFFICIENT LMS ADAPTIVE FILTER FOR CHANNEL EQUALIZATION

Pravin S. Wakode , Prof. B. J. Chilke , Er. S. S. Kemekar ,[3]Mrs.Yogita Pagrut

[1] M-tech 4th sem Electronics Engg. S.D.C.O.E. selukate, (Wardha),[2]Dept. of Electronics Engg. S. D. C.E. Selukate, (Wardha), Inox Air Product Ltd., Wardha, 4Asst. Prof. MSOP,Akola

[1]parvwak23@gmail.com, [2] b_kuchewar@rediffmail.com

ABSTRACT :

Modern Wireless Communication System impose the need of Adaptive Channel Equalizer with better rate of convergence. Least Mean Square(LMS) algorithm is one of the best adaptive algorithm in the case. In Wireless communication, equalization is an effective technology to change channel characteristics and reduce the Inter-Symbol Interference (ISI). Based on analyzing the LMS algorithm, the application of this algorithm in Channel equalizer is studied. In this paper we discussed about behaviour of the LMS algorithm in design of the Adaptive Filter for the application of Channel equalization.

Keyword : ***Least Mean Square (LMS), Intersymbol Interference (ISI), Equalization, Mean Square Error (MSE).***

I. INTRODUCTION:

In this modern era wireless communication is heart of the world. We know that in a communication system every signal carries useful as well as unwanted information. For transmitting and receiving such signals we required an efficient system which may compensate the various unwanted noise. The usual method to remove the noise is to pass the signal through filter that tends to suppress the noise without affecting the information signal. Filters are of two types; it may be fixed or adaptive. Fixed filters are the static filters which are unable to update their filter coefficient according to the output desired. On the other hand adaptive filters can vary their coefficient according to the require output signal. LMS adaptive filter is the most commonly used adaptive filter not because of its simplicity but also its satisfactory convergence performance. It was invented in 1960 by Stanford University professor Bernard Widrow and his first Ph.D. student, Ted Hoff. Adaptive digital filters find various applications in Digital Signal Processing eg. Channel equalizer, Echo cancellation, Noise cancellation, Channel estimation etc.

To implement the LMS algorithm, one has to update the filter weights during each sampling period using the estimated error, which equals the difference between the current filter output and the desired response. Since all weights are updated concurrently in every cycle to compute the output, direct-form realization of the FIR filter is a natural candidate for implementation. However, the direct-form LMS adaptive filter is often believed to have a long critical path due to an inner product computation to obtain the filter output. This is mainly based on the assumption that an arithmetic operation starts only after the complete input operand words are available. Since this critical-path estimate is quite high, it could exceed the sample period required in many practical situations, and calls for a reduction of critical-path delay by pipelined implementation. But, the conventional LMS algorithm does not support pipelined implementation. The implementation of traditional LMS is difficult to satisfy the real time demand and high speed of digital signal processing. What's worse, it badly increases the system cost with the improvement of the sampling rate, capacity of single processing unit and system frequency.

II. TRAINING ADAPTIVE FILTER :

An adaptive filter is a time-varying filter which must constantly be retuned. The adaptive algorithm is controlled by error signal. This error signal is derived by comparing the output of the equalizer with some signal which is either an exact scaled replica of the transmitted signal or which represents a known property of the transmitted signal. The adaptive algorithm uses error signal to minimize a cost function and updates the filter weights in a manner that iteratively reduces the cost function. For example, the least mean square algorithm searches for the optimum or near optimum filter weights by performing the following iterative operation:

New weights = Previous weights + (constant) x (Previous Error) x (Current Input vector)

Where

Previous error = (Previous desired output – Previous actual output)

And the constant may be adjusted by the algorithm to control the variation between filter weights on successive iterations. This process repeated rapidly in a programming loop while the filter attempts to converge. Upon reaching convergence, the adaptive algorithm freezes the filter weights until the error signal exceeds an acceptable level or until a new training sequence is sent. Base on classical equalization theory, the most common cost function is the mean square error (MSE) between the desired signal and the output of the equalizer. By detecting the training sequence, the adaptive algorithm in the receiver is able to compute and minimize the cost function by driving the tap weights until the next training sequence is sent.

III. LITERATURE REVIEW:

Mihir Narayan Mohanty, Biswajit Mishra, et.al has worked on the FPGA implementation of the constrained LMS algorithm. The use of adaptive systems in a wide variety of applications is one of the factors to update the current research. For building and realizing the adaptive filters, Digital Signal Processors and Application specific Integrated Circuits (ASICs) are the common and popular medium. For such adaptive systems Least Mean Square (LMS) algorithm has occupied an important space for its simplicity and robustness. Practically step size parameter as a constraint, is a key parameter of LMS algorithm. Simultaneously Field Programmable Gate Array (FPGA) has become attractive for the purpose of realization. In this paper, an approach has been proposed to realize the constrained LMS (CLMS) in FPGA. Variable step size parameter is considered as the constraint. Simulation results for system identification are provided to support the theoretical analysis. The result shows the efficacy of the approach [1].

Xianglei Dong, Huiyong Li, Yu Wang have proposed the High-speed FPGA Implementation of an Improved LMS Algorithm. The FPGA implementation of a new parallel processing method was studied by introducing the parallel processing method into the delayed least mean square (DLMS) algorithm. The parallel delayed least mean square (PDLMS) algorithm has the faster data throughput and higher convergence rate than the DLMS algorithm. The hardware implementation of PDLMS is realized by hardware description language, while the simulation structure is presented. The results show that the PDLMS algorithm has certain superiority according to DLMS [2].

Pramod Kumar Meher and Sang Yoon Park presented Area-Delay-Power Efficient Fixed-Point LMS Adaptive Filter architecture with low adaptation-delay. For achieving lower adaptation-delay and area-delay-power efficient implementation, they use a novel partial product generator and proposed a strategy for optimized balanced pipelining across the time-consuming combinational blocks of the structure. From synthesis results, they found that the proposed design offers nearly 17% less area-delay product (ADP) and nearly 14% less energy-delay product (EDP) than the best of the existing systolic structures, on average, for filter lengths N = 8, 16, and 32. They proposed an efficient fixed-point implementation scheme of the proposed architecture, and derived the expression for steady-state error. They were shown that the steady-state mean squared error obtained from the analytical result matches with the simulation result. Moreover, They have proposed a bit-level pruning of the proposed architecture, which provides nearly 20% saving in ADP and 9% saving in EDP over the proposed structure before pruning without noticeable degradation of steady-state-error performance [3].

Pramod Kumar Meher and Sang Yoon Park in later research have performed a precise analysis of the critical path of the least-mean-square (LMS) adaptive filter for deriving its architectures for high-speed and low-complexity implementation. It was shown that the direct-form LMS adaptive filter had nearly the same critical path as its transpose-form counterpart, but provides much faster convergence and lower register complexity. From the critical-path evaluation, it was further shown that no pipelining is required for implementing a direct-form LMS adaptive filter for most practical cases, and can be realized with a very small adaptation delay in cases where a very high sampling rate is required. Based on these findings, this paper proposed three structures of the LMS adaptive filter: (i) Design 1 having no adaptation delays, (ii) Design 2 with only one adaptation delay, and (iii) Design 3 with two adaptation delays. Design 1 involves the minimum area and the minimum energy per sample (EPS). The best of existing direct-form structures requires 80.4% more area and 41.9% more EPS compared to Design 1. Designs 2 and 3 involve slightly more EPS than the Design 1 but offer nearly twice and thrice the MUF at a cost of 55.0% and 60.6% more area, respectively. [4]

Shibalik Mohapatra and Asutosh Kar implemented a feedback FXLMS algorithm for noise cancellation. Here, an improved feedback FXLMS algorithm has been used to cancel noise. The original signal was passed through a primary filter having transfer function P(z) to give the desired input d (n) . The secondary path transfer function S(z) exists but was unknown in most practical applications. Thus it had to be estimated and used for noise cancellation. A S(z) is the estimated secondary path transfer function which was generated by offline modeling of the system [5]

Zhang Bo, Tian Xiuwei has worked on the design of novel implementations of adaptive FIR filter based on multiplier-free structure. The implementation scheme based on the distributed algorithm uses access to a range of look-up table (LUT) to replace the traditional method of multiply-accumulate operations. The update of the adaptive filter weights was based on the quantization error least mean square algorithm (QE-LMS). To reduce the time, the update process of the LUT which stored filter weights is realized by a novel LUT update using a matched auxiliary LUT. The filter was described with VHDL, and realized on Cyclone series chip. The system simulation and timing analysis show that the proposed method could implement FIR filters with the smaller resource usage and higher speed [6].

Yongbo Bai Zili Chen et.al has discussed FPFA implementation of a channel equalizer based on LMS algorithm. Equalization is an effective technology to change channel characteristics and to reduce the intersymbol interference (ISI). Based on analyzing the LMS algorithm, the application of this algorithm in channel equalizer was studied. A channel equalizer based on LMS algorithm was implemented using Xilinx system generator for DSP develop software. This proposed implementation was a novel way to develop FPGA. The simulation results show the equalizer can eliminate ISI; the performance of the implemented equalizer is dependable [7].

III. CONCLUSION:

LMS, PDLMS and DLMS all provides stability and satisfactory convergence performance but having different design complexity and critical path. In that case there is lot of scope to design a structure with less complexity as well as less critical path.

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