ABSTRACT

This review paper is carried out in two centring. Firstly, a survey is done to know the work done on adaptive filters and secondly to know how and where the adaptive algorithms which led to the evolution of desired signal are being used in number of applications over the years and the effect of these algorithms on different areas over the years.

Keywords: Adaptive filters, Adaptive algorithms.

1. INTRODUCTION

Carrying out literature review is very significant in any research project as it clearly establishes the need of the work and the background development. It generates related queries regarding improvements in the study already done and allows unsolved problems to emerge and thus clearly define all boundaries regarding the development of the research project. Some recent papers have surveyed through this research, after review these papers aimed methodology is proposed.

2. USAGE AND WORK DONE ON ADAPTIVE FILTERS

Signal processing field has been made substantial contributions over the past thirty years. Due to the advances in digital circuit design, digital signal processing (DSP) systems have become attractive. Filtering application of DSP includes digital systems. A signal is processed by digital systems to control the information contained in the input signal. The Adaptive filters are acceptable in any unknown environment. The Adaptive filter is a powerful device for signal-processing and control applications in time variation environment of input statistics. To reduce the signal corruption stimulated by predictable and unpredictable noise adaptive filters are used. Some applications such as identification, inverse modelling, prediction and interference cancellation are essential to explicate the problem of acoustic echo & noise cancellation and related issue. Researchers have developed various algorithms for active interference cancellation to obtain adaptive filter mainly LMS, NLMS and RLS algorithm. NLMS has a better learning rate than LMS based adaptive filter. Thus the NLMS filter is in more existence. Rate of convergence, misadjustment, numerical robustness, computational requirements and stability are the performance measures of adaptive algorithm. The digital signal processing applications inflict significant constrains on area, power dissipation, cost and speed. Hence the design tool should be carefully opted. Generally ASIC, DSP and FPGA are mainly used to design of such application. The serial architecture of DSP restricts it to process high sampling rate applications but it can be used for extremely complex math-intensive tasks. The lack of flexibility and long design cycle makes ASIC complex and time consuming. The FPGA overcome the disadvantages of ASIC and DSP. Now a days in the field of signal processing, FPGA becomes the best selection for the design of signal processing system because of their tractability and prominent bandwidth, resulting from their collimate architecture.

3. USE AND EFFECT OF ADAPTIVE ALGORITHMS OVER THE YEARS

In several application of interference cancellation, with the help of adaptive algorithms the changes in signal characteristics could be rather fast. Because of simplicity in computation and implementation, LMS and NLMS adaptive filters are widely used in signal processing application. The RLS algorithm is the "ultimate" adaptive filtering algorithm since it is exhibiting the best convergence behaviour [1]. A new algorithm is developed by Chansarkar, M.M.Desai & U.B. in 1997; due to persistent and bounded data disruptions to be bounded this algorithm ensures the annealed bias in the weight vector. An approximate recursive implementation is called as the Robust Recursive Least Squares algorithm. The RLS algorithm is racy with respect to persistent bounded data disruptions. To exemplify the
efficacy of the RRLS algorithm simulation results are presented [2]. In 2001 P. Shristi, W.S. Lu & A. Antoniou proposed the new Variable-Step-Size (VSS-LMS) algorithm and investigated its performance through simulation. The obtained results show the superior performance achieved over the MVSS algorithm in the case of long adaptive filters. The proposed algorithm & a corresponding fixed-step-size FSS LMS algorithm are then adaptive applied to sub band adaptive echo cancellation. When compared with the FSS sub band LMS algorithm simulation result show that the proposed algorithm yields a lower steady-state misadjustment as well as a lower residual MSE. In comparison with the NLMS sub band algorithm, the proposed algorithm results in a slower convergence rate but also a lower steady-state misadjustment. It has been observed that improved system performance can only be achieved with a distinct step-size adaption for each individual sub band [3]. In digital signal processing such that channel estimation, interference cancellation, channel equalization, the adaptive algorithm is used. The LMS algorithm is one of the most important adaptive algorithms. Both the residual fault level & the convergence speed are decided by the step size in the LMS algorithm. The Variable Step-Size LMS algorithm is accommodated for obtaining both the residual fault level & highest speed of convergence. Various VS-LMS algorithms have been reviewed and a modified VS LMS algorithm proposed in 2007 by Li Yun [4]. Raj kumar Thenua & S.K. Agarwal implemented hardware of NLMS Algorithm for Adaptive Noise Cancellation in 2010. The adaptive NLMS algorithm is implemented on DSP processor in real-time environment to de-noise an ECG signal. A simulink model is produced & associated to TI TMS320C6713 digital signal processor which is embedded to simulink toolbox and realtime workshop to execute hardware adaptive noise cancellation. The system is examined for three degree of noise and shows better improvement in SNR [5]. Tracking speed and stability of adaptive gradient filtering algorithms represented by LMS are limited for non-stationary environment. In 2012 Harjeet Kaur, Dr. Rahul Malhotra & Anjali Pathi evaluated the noise cancellation simulation outcomes. According to these outcomes only after 20 iterative operations, this algorithm can be stabilized and provides stronger ability to boost SNR of weak signal as compared to LMS, NLMS, Variable size, sign LMS filter. All outcomes designate that tracking ability and convergence stability are superior to other algorithms [6]. Several techniques are used by P.Radhika, Monpur Ashwin & Chunduri.V.M.Naren Simha in 2014 for removal of unwanted entities from signals. The power line interference from all sensitive monitoring equipment’s can be removed by implementing several techniques with different error nonlinearity-based adaptive filters. The suggested implementation is best for applications such as biotelemetry. These systems employ simple addition, shift operations and attain considerable speed up over the other LMS-based realizations [7].

4. FPGA MODEL FOR ADAPTIVE ALGORITHMS

In 2010 K. R. Rekha, Dr B. S. Nagabushan & Dr K.R. Nataraj proposed a VHDL implementation of a variable step size Least Mean Square (NLMS) adaptive algorithm. The good convergence and good stability of NLMS algorithm have made it preferable. Adaptive filtering comprises one of the kernel technologies in digital signal processing and are used extensively admitting system identification, adaptive equalization, adaptive noise cancellation, wireless communication and echo cancellation. As compared conventional LMS it has been proven that NLMS algorithm has good behaviour [8]. A novel pure-hardware design of NLMS-based adaptive FIR filter core which is highly efficient in FPGA area/resource utilization and speed is proposed by Omid Sharifi-Tehrani in 2011.Unlike HW/SW co-design and other pure-hardware methods, the required area/resource is reduced while keeping the speed in an appropriate level by taking advantage of Finite State Machine (FSM) and using internal block-rams (BRAM). This model because of being completely general (device independent), gives the ability of implementation on different FPGA brands and thus, is suitable for embedded systems, system-on-programmable-chip (SoPC) and network-on-chip (NoC) applications [9]. The LMS algorithm is used to modify the filter coefficient. This recognized for its low computational complexity, simplification & improved performance. The LMS algorithm is well suited if the number of iterations involved for convergence and it is achieved by a sufficient choice of bit length to represent the filter’s coefficients. Asit Kumar Subudhi presented a low cost and prominent performance programmable digital finite impulse response filter in 2011. The structure uses the computation sharing algorithm to reduce the computation complexity [10]. Hesam Ariyadoost in 2011 implemented the adaptive digital LMS and DLMS FIR filters on FPGA chips for distinctive interference cancellation applications and compared the behaviour these adaptive algorithms in terms of the filter critical path time and chip area utilization. The direct FIR architecture is conceived for filter designing and the VHDL is used for algorithm modelling. The synthesize tool QUARTUS-II present that the DLMS algorithm has a faster pipeline architecture than LMS algorithm in the cost of using more chip area due to use of extra registers[11]. In 2011 Ioana Homana proposed a RTL description of two long-familiar adaptive algorithms used in acoustic echo cancellation: the LMS and NLMS. The RTL descriptions of these algorithms are established on their FSM model and were designed in the popular VHDL language. ModelSim simulation outcomes entirely with plots obtained in MATLAB prove the effective behaviour [12].
Now days FPGA systems are replacing dedicated PDSP systems due to their greater tractability and eminent bandwidth, resulting from their parallel architecture. The FPGA platform is well suited for the complex real time audio processing. An adaptive noise cancellation process has been implemented. The pertinency of a FPGA system for speech processing is explained in this paper by A. B. Digikar and S. S. Ardhapurkar Kartheek in 2012. LMS algorithm is used in many signals processing environment and is carried out for adaption of the filter coefficients. VHDL module is used to implement the cancellation system and simulation of VHDL design of adaptive filter. VLSI design is analised on the basis of SNR and MSE [13]. Due to simplicity the NLMS algorithm is most popular in signal processing. The optimal step size NLMS algorithm is able to solve the conflicts of fast convergence and low excess mean square error associated with a fixed step size NLMS. To control the step size and the theoretical performance analysis of the steady state nature, L. Bharani and P. Radhika deduced a new nonparametric algorithm in 2013. The simulation outcomes in MATLAB show that offered algorithm performs better in Fast convergence rate, low error rate in noise cancellation [14]. Electrical engineering program efforts to introduce software/hardware design concepts and tools in senior-level and senior-design courses. Dr. Wagdy H Mahmoud and Dr. Nian Zhang provided details of laboratory exercises and a senior project to implement adaptive filters using variations of the least mean square (LMS) and the recursive least squares (RLS) algorithms and the use of adaptive filters designed using these algorithms in the design of adaptive noise cancellation system. The adaptive noise cancellation system on an FPGA board was also implemented in 2013[15]. In 2013 B. V and Manoj kumar introduced the applicability of a FPGA for speech processing in which adaptive filtering technique is used for interference cancellation. For the complex real time audio processing the FPGA platform is well suited. Here to evaluate the input signal Spartan -3 FPGA XC3s400pq208-5 board is used. The resources can be minimized when tested with different signals [16]. In 2013 Shashikala Prakash implemented the LMS algorithm by using two different structures. The Active Vibration Control system implemented on Xilinx Virtex–4 FPGA. A comparison between fixed point and floating point data representations is took out on the basis of a FSM model. In VHDL the floating point LMS algorithm uses the Intellectual Property cores available from Xilinx Inc. Outcomes of two architectures shows that floating point implementation is better option in all aspects as area and performance [17]. S. Thilagam presented the structure of a real time adaptive NLMS filter for non-stationary noise cancellation in an automobile environment in 2013. In this paper, Xilinx System Generator 12.3 on Spartan 3E FPGA is used to realize proposed efficient Adaptive Noise Cancellor. Xilinx proposed System Generator tool for DSP design that enables the use of the Math Works model-based design environment, Simulink for FPGA design. Some steps like synthesis, placement and routing are used to develop an FPGA programming file and the design is evaluated in terms of hardware resources, speed and power dissipation [18]. Compensation of amplitude and phase dispersion is being done by Equalizers which results in the intervention of the transmitted signals with each other. In 2014, design of an adaptive equalizer has been presented by Ms. Manpreet kaur and Ms. Cherry using LMS algorithm. The fractional spaced adaptive equalizer is implemented on a Xilinx FPGA device and performs better in terms of fast convergence and bit error rate [19]. Rupali V. Mane and Dr. M. T. Kolte in 2014 surveyed the principles of Adaptive Noise Cancellation and its Applications. FPGA implementation of two adaptive Filtering Algorithms LMS and wiener are implemented in this paper. Here to accommodate the varying nature of speech signal wiener filter is implemented in adaptive manner in time domain rather than in frequency domain [20].

5. COMPARISON OF ADAPTIVE ALGORITHMS

In 2013 Subhash Chandra Tiwari, R.S. Prasad & Masood Anzar presented the Genetic Algorithm (GA’s) based tuning of LMS Adaptive Filter. A Comparative analysis for speech signals filtering is done and for speech signal highlighted the edge of each algorithm explicitly. The need to develop GA based LMS filter is emphasized and the technique is proposed. Moreover, it has been tried to develop a hybrid filter with better performance. Comparison of adaptive filters LMS filter gives a better filter when the signal value is above a particular threshold value. Use of NLMS gives a somewhat smaller initial state error thus NLMS algorithm is comparatively faster than LMS algorithm. LMS filter is the simplest filter with less numerical complexity. The performance of LMS filter depends on basically two parameters, i.e., order and step size of the filter. The design of GA based LMS filter gives optimized results [21]. Considering today’s technical scenario of communication system, possesses additive noise, signal interference and echo etc. Due to this reason the error is generated at the time of transmission of data. Hence adaptive filter is an appropriate option to reduce the noise or channel effects. In 2014 Shraddha K. Mendhe and Dr. S. D. Chede represented the adaptive echo cancellation using normalized least mean square (NLMS) algorithm. The NLMS algorithm is the normalized version of least mean square (LMS) algorithm. The NLMS algorithm demonstrates a more beneficial equalizer between performance & simplicity than least mean square algorithm. High speed of adder and subtractor to design LMS and NLMS algorithm is also realized as well [22].
6. CONCLUSION

This paper has reviewed the Adaptive filtering algorithm using FPGA regarding the present problem. There are number of adaptive algorithms available in literature and every algorithm has its own properties. A review of adaptive filters shows that the LMS algorithm is still a popular choice for its stable performance, high speed capability & high convergence rate, but aim of every algorithm is to achieve minimum mean square error at a higher rate of convergence with lesser complexity. So that the NLMS is used as the adaptive algorithm to achieve minimum mean square error at a higher rate of convergence with lesser complexity i.e. low computational cost and ease of implementation & robust and reliable. The FPGA platform is well suited for the complex real time audio processing. The high-speed capability and register rich architecture of the FPGA is ideal for implementing Adaptive filtering algorithm.

REFERENCES


AUTHORS

Devendra Goyal has received his B.E. Degree in 2009 Electronics & Communication Engineering from MIT, KOTA & now pursuing M. Tech in VLSI DESIGN from Poornima College of Engineering Jaipur. His current area of research includes Adaptive Filter, VHDL and currently working on the project related to it.

Mr. Manish Singhal passed B.E. in 2001 and M.Tech in 2011 from MNIT, Jaipur with specialization in VLSI Design. He has one year industry experience and after that he joined Poornima College of Engineering, Jaipur as a lecturer. He has written two books Microprocessor & Microcontroller and Microcontroller & Embedded System Design. He has published several papers in national and international journals and has guided 6 M.Tech students. His research areas include Microprocessor, Microcontroller, Embedded System Design, Digital VLSI Design and Image Processing.