

A Review on Adaptive Algorithms

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Abstract - The need of adaptive filtering arises when the received signal in its course of propagation is corrupted by the noise which changes continuously. The existence of noise affects the processing and application of the signal. Adaptive algorithms form the core of these adaptive filters which adjusts its parameters according to the environment to get a optimal signal. This review presents a survey to know the work done on different adaptive algorithms LMS, NLMS, RLS which are applied in the fields of signal processing, communication, signal control applications.

Key Words: Adaptive algorithm, Adaptive filter, MATLAB, VHDL.

1. INTRODUCTION

Adaptive filters are used in the environment where the signal parameters or signal constraints change continuously because of their robustness and good tracking capabilities. Adaptive filters are time-variant and its coefficients are adjusted in order to optimize the specific objectives like mean square of the error signal. The algorithms which implement the functionality of the adaptive filters are complex, hence almost all the adaptive filters are digital filters. There are various adaptive algorithms developed for various applications, among them Least Mean Square algorithm, Normalized Mean Square algorithm and Recursive Least Mean Square algorithm are discussed in the paper. The research work done on these algorithms which are used for optimized MSE are discussed in this paper.

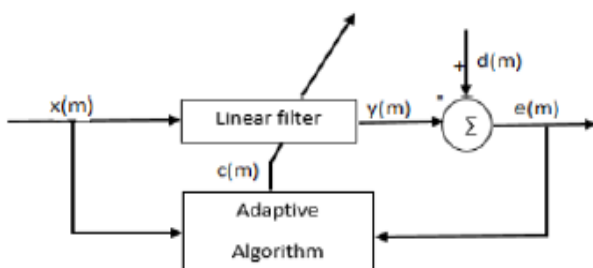


Fig. 1. General block diagram of adaptive filter.

1.1 Adaptive Algorithms

The difference between the conventional filters and adaptive filters is the capability of adaptive algorithms to adjust its filter coefficients without the knowledge of previous signal or noise characteristics.

The reference signal which is derived from the noise field is used. This reference signal is filtered and deducted from the initial input containing both the noise and the signal resulting in an error signal.

$x(n)$ - input signal with noise

$d(n)$ -desired signal

$e = x(n) - d(n)$ -error signal

Thus the adaptive filter parameters are adjusted in order to reduce the difference between $x(n)$ and $d(n)$ to make the error as minimum as possible. The three algorithms-LMS algorithm, NLMS algorithm and RLS algorithm have different convergence speeds.

2. LITERATURE SURVEY

Different researchers used different algorithms to carry out the process of noise cancellation. Some of the important research works are reviewed in this paper.

This paper discusses about the need of adaptive algorithms and different types of adaptive algorithms including LMS, NLMS, RLS algorithms. These algorithms are compared on the basis of MATLAB and verilog simulation results. The simulation results show that RLS algorithm has the highest convergence rate when compared to LMS and NLMS algorithm. LMS algorithm makes use of gradient descent method to search for the optimal condition, with mean square error as the cost function. The gradient requires the knowledge of actual values of the auto-correlation matrix which will be a difficult task to know. RLS algorithm is advantageous as it uses instantaneous values of auto-correlation matrix. The MSE for different SNR ratios show that the RLS is the quicker than both LMS and NLMS algorithms which the

NLMS is quicker than LMS algorithm in terms of its convergence rate. (1)

In this paper an adaptive noise cancellation technique using NLMS algorithm in GNU is discussed. According to this paper the adaptive filters has high noise removal efficiency when compared to direct filtering because of its higher noise rejection level. The advantages of NLMS algorithm are discussed in this paper. The filter belongs to the LMS family. The NLMS filter algorithm has the same principle of LMS but the weight control mechanism is different from LMS algorithm. The error estimation is done by subtracting the M-by-1 tap input vector from the desired response. The simulation is carried out in GNU radio companion. A corrupted cosine signal and an audio signal are given as an input and the simulation was carried out revealing the advantages of noise cancellation with NLMS algorithm. (2)

In this paper drawbacks and limitation of LMS algorithm are discussed. The statistical analysis of mean-square adaptive algorithm with uncorrelated gaussian data is presented. The analytical expression for steady state mean square error and performance degradation due to weight vector misadjustment are derived. It is found that the adaptive coefficient μ , which controls the rate of convergence of the algorithm must be restricted to an interval significantly smaller than the domain commonly stated. The outcome of this paper, places the fundamental limitations of mse performance and rate of convergence of LMS algorithm. As a result the LMS algorithm is very sensitive to the noise and in order to get a significant convergence rate other adaptive algorithms can be taken into consideration. (3)

The objective of this paper is to study the tracking capabilities of the time-varying systems by an adaptive filter with LMS algorithm in the presence of white, zero-mean reference input. The tracking capabilities of time-variations by the LMS algorithm is one of the aspect while two cases of time variations of an unknown system i.e., zero mean time increment and deterministic bounded increments have been studied. (4)

The aim of this paper is to predict the performance of the LMS and NLMS algorithms in tracking the time-varying fields. In order to make the tracking possible a procedure is implemented and simulation results show the efficiency of the proposed algorithms. A simple smoothing on the incremented weights is applied to predict the weights for the next iteration. The trade-off between weight vector noise and lag weight vector by conventional LMS algorithm in time-varying environment is modeled by Markov process of order by Widrow. The algorithm has a very low order of arithmetic complexity. Moreover, this procedure could be combined with a wide class of adaptive filters (e.g., RLS, gradient lattice algorithm, etc.) to improve their behaviors. The proposed algorithm is

obtained by simplifying a Kalman filter. To this end, a Markov model of second order is considered for the weight vector. This model shows that the estimation of parameter increments inferred from the predicted parameters improves the tracking performance. (5)

In the paper prediction of stock market trends with the help of adaptive algorithms has been discussed. The advantage of using traditional techniques such as technical analysis and signal processing techniques such as moving averages and regression were limited due to the dynamic behavior of the markets. In this paper a very innovative approach of applying adaptive filters for the prediction of behavior of financial signals was implemented. The hybrid filters used are DCT-LMS, DCT-NLMS, DCT-RLS and Kalman filters. The proposed method is used to predict the values of five of the largest stock markets. The performance of hybrid adaptive filters is compared against the conventional filters like autoregressive (AR), Moving Average (MA) filters and adaptive filters like LMS, NLMS etc. The base technique considered is the Random Walk (RW) process which acts as the benchmark technique. The results show a high degree of prediction accuracy for the hybrid adaptive filters, which is very high when compared to conventional filters, thus indicating that hybrid adaptive filters can be successfully used for stock market prediction. (6)

This paper compares the adaptive algorithms like LMS algorithm, NLMS algorithm, RLS algorithm, TVLMS algorithm and FTRL algorithm on the basis of their computational complexity and SNR. The adaptive behavior of the algorithms is analysed. The MSE, algorithm and the required filter order are criteria on which the algorithms are analysed. Accordingly, there are many digital signal processing applications where second order statistics cannot be applied. Therefore adaptive algorithms can be used in those applications. The comparison of the algorithms at a constant sampling rate of 1.5KHz and at different variance values show that RLS algorithm has good SNR improvement and can be considered as fast converging algorithm. (7)

This paper has reviewed and studied the previous works done on adaptive algorithms in the field of acoustic echo cancellation. The echo cancellation is considered to be one of the important aspects of design of modern communication system. There are mainly two types of echo that are present that are hybrid echo and acoustic echo which are the main area of concern in this paper. Hybrid echo is caused due to the mismatch of impedance in transmission lines whereas Acoustic echo is a kind of noise signal in which the audio signal is resonated in real environment due to the reflection from surrounding objects, walls, floors or surfaces etc. Here along with the original required signal the attenuated, time-delayed images of this speech signal is produced which creates disturbance. Adaptive algorithms are used to remove the

echo and performance of each echo is evaluated. And the discussion is concluded by proving RLS algorithm has better echo cancellation when compared to other algorithms. (8)

This paper discusses the use of FPGA systems in place of Programmable Digital Signal Processor systems (PDSP) because of their greater flexibility and higher bandwidth due to its parallel architecture. In this discussion FPGA is used to implement the adaptive filters for noise cancellation. The VHDL design of adaptive filters is performed and analyzed on the basis of SNR and MSE. The FPGA platform is well suited for the complex real time audio processing. An adaptive noise cancellation process has successfully been implemented for filter order up to 256 using Spartan -3 FPGA XC3s400pq208-5 board. When tested with different signals, the system showed an improved performance compared to the original signal. (9)

In this paper the advantages of using NLMS algorithm over LMS algorithm are discussed. The Normalized Least Mean Square error (NLMS) algorithm is most popular due to its simplicity. The conflicts of fast convergence and low excess mean square error associated with a fixed step size NLMS are solved by using an optimal step size NLMS algorithm. The main objective of this paper is to derive a new nonparametric algorithm to control the step size and also the theoretical performance analysis of the steady state behavior is presented in the paper. The simulation experiments are performed in Matlab. The simulation results show that the proposed algorithm as superior performance in Fast convergence rate, low error rate, and has superior performance in noise cancellation. (10)

The aim of this paper is to implement the adaptive digital Least Mean Square (LMS) and delayed LMS (DLMS) Finite Impulse Response (FIR) filters on Field Programmable Gate Array (FPGA) chips for typical noise cancellation applications and compare the behavior of LMS and DLMS adaptive algorithms in terms of chip area utilization and the filter critical path time or filter frequency. The direct FIR architecture is considered for filter designing and the VHDL hardware description language is used for algorithm modeling. The obtained results by the synthesise tool QUARTUS II on a single STRATIX II chip, EP2S15F484C3, from ALTERA Inc. demonstrate that the DLMS algorithm which has a pipeline architecture is faster than LMS algorithm while it uses more chip area due to using extra registers. (11)

This paper proposes a VHDL implementation of a variable step size Least Mean Square (NLMS) adaptive algorithm. The envisaged application is the identification of an unknown system. The good convergence of NLMS algorithm has made us to choose it. It also has good stability. Adaptive filtering constitutes one of the core technologies in digital signal processing and finds

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, wire less communication and echo cancellation. A HDL implementation is developed for a 4th order NLMS adaptive filter. As compared conventional LMS it has been proven that NLMS Algorithm has good behavior. ModelSim simulations results altogether with plots obtained in Matlab prove the same. (12)

COMPARISION OF ADAPTIVE ALGORITHMS			
ALGORITHM	SPEED OF CONVERGENCE	STABILITY	ROBUSTNESS
LMS	SLOW	STABLE	LESS
NLMS	FAST	STABLE	LESS
RLS	FAST	UNSTABLE	MORE

3. CONCLUSION

This paper has reviewed the algorithm using MATLAB and VHDL simulation. There are various algorithms modified algorithms developed to meet the requirements of the application. The RLS algorithm and NLMS algorithm shows good results in terms of convergence and tracking capability. Whereas the LMS algorithm is the simplest and stable algorithm.

REFERENCES

1. *Comparative Study of Adaptive Algorithms Using Matlab and Verilog*. Parvathy, A and Narayanan, G. 2018, p. 6.
2. *Adaptive noise cancellation using NLMS algorithm in GNU radio*. J. Adarsh, P. Vishak and R. Gandhiraj, Coimbatore, : s.n., 2017.
3. **A. Feuer, E. Weinstein**,. *Convergence analysis of LMS filters with uncorrelated Gaussian data*. s.l. : IEEE Transactions on Acoustics, Speech, and Signal Processing, 1985.
4. *Tracking capability of the least mean square algorithm: Application to an asynchronous echo canceller*. **S. Marcos, O. Macchi**, s.l. : IEEE, 1987, Vol. 35.
5. *Prediction in LMS-type adaptive algorithms for smoothly time varying environments*. **Gazor**, s.l. : in IEEE Transactions on Signal Processing,, 1999, Vols. 47,.
6. *Application of hybrid adaptive filters for stock market prediction*. **Nair, Binoy.Mohandas, V.Sakthivel, N.Nagendran, S.Nareash, A.Nishanth, R. Ramkumar, S.Kumar, D.** s.l. : International Conference on Communication and Computational, 2010.

7. *A technical review on adaptive algorithms for acoustic echo cancellation.* **A. Deb, A. Kar, M. Chandra.** s.l. : IEEE, 2014.

8. *Comparison between Adaptive filter.* **Jyothi, Dhiman.** s.l. : International Journal of Science Engineering and Technology Research (IJSETR), 2013, Vol. 2.

9. *Design and Implementation of Adaptive filtering algorithm forIEEE.*, **Mrs. A. B. Diggikar, Mrs. S. S. Ardhapurkar.** s.l. : IEEE,, 2012.

10. *FPGA implementation of optimal step size NLMS algorithm and its performance.* **L. Bharani, P. Radhika,** s.l. : International Journal of Research in Engineering and Technology (IJRET), 2013, Vol. 02.

11. *Performance Evaluation of LMS and DLMS Digital Adaptive FIR Filters by Realization on FPGA.* **Hesam Ariyadoost, Yousef S. Kavian, Karim Ansari.** s.l. : International Journal of Science & Emerging Technologies, 2010, Vol. 2.

12. *FPGA Implementation of NLMS Algorithm for Identification of unknown system.* **K. R. Rekha, Dr B. S. Nagabushan, Dr K.R. Nataraj,** s.l. : International Journal of Engineering Science and Technology,, 2010, Vol. 2.