ISSN: 2278 – 909X
International Journal of Advanced Research in Electronics and Communication Engineering (IJARECE)
Volume 4, Issue 6, June 2015

Review on Adaptive Filter Algorithm and Process of Echo Cancellation

R. M. Potdar[1], Dr. Mekhram Meshram[2,] Dr. Ramesh Kumar[3], Bhupeshwari Chandrakar[4]

***Abstract*— This review paper is carried out in two concerning. Firstly, a survey is completed to know the effort on adaptive filters and secondly to know how and where the adaptive algorithms which controlled the evolution of desired signals are being used in number of applications. The aim of proposed survey is to know the process of echo cancellation.**

***Index Terms*—Adaptive filters, Adaptive algorithms, echo cancellation.**

I. INTRODUCTION

A literature review is very significant in any research work as it clearly establishes the need of the work. It generates queries about improvements in these studies and allows unsolved problems to emerge and thus obviously express all boundaries regarding the progress of the research work. Various recent papers have studied through this research, after these review we have proposed a new methodology.

II. USAGE AND WORK DONE ON ADAPTIVE FILTERS

The Signal processing field has been made large contributions over the past thirty years. Due to the advances in digital circuit design, digital signal processing systems have become attractive. For filtering application digital signal processing techniques includes digital systems. A signal is handled by digital systems to control the information of the input signal. The Adaptive filters are suitable in any new environment. It is a powerful device for signal processing and control applications in time variation environment of input statistics. To reduce the signal corruption inspired by predictable and unpredictable noise by use of adaptive filters. Its applications such as identification, inverse modelling, prediction and interference cancellation are essential to explicate the problem of acoustic echo and noise cancellation. Investigators have established various algorithms for active interference cancellation to obtain adaptive filter mainly LMS, NLMS and RLS algorithm. RLS has a better learning rate than LMS based adaptive filter. Rate of convergence, maladjustment, numerical robustness, computational requirements and stability are the performance measures of adaptive algorithm.

III. ECHO AND PROCESS OF ECHO CANCELLATION

The adaptive algorithm provides several application of interference elimination, and then with the help of these algorithms we can changes the signal characteristics could be faster. LMS and NLMS adaptive filters are widely used in signal processing application, because it is so simple in implementation and computation. The RLS algorithm is the "ultimate" adaptive filtering algorithm since it is exhibiting the best convergence behavior [1]. A new algorithm is developed by Chansarkar, M.M.Desai and U.B. in 1997; due to persistent and bounded data disruptions to be bounded this algorithm ensures the hardened bias in the bulk vector. An estimated recursive implementation is called as the Robust Recursive Least Squares algorithm. The RLS algorithm is shocking 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 elimination. When compared with FSS sub band LMS algorithm simulation result show that the proposed algorithm yields a lower steady-state maladjustment as well as a lower residual MSE. In comparison with the NLMS sub band algorithm, the suggested algorithm results in a slower convergence rate but also a lower steady-state failure alteration. It has 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, the adaptive algorithm is used for channel estimation, interference cancellation, channel equalization etc. The LMS algorithm is one of the most important adaptive algorithms. It offers residual fault level & the convergence speed are decided by the step size. The Variable Step-Size LMS algorithm is accommodated for obtaining both the residual fault level & highest speed of merging. Several VS-LMS algorithms have reviewed and a modified VS LMS algorithm proposed in 2007 by Li Yan [4]. Raj kumar Thenua & S.K. Agarwal implemented hardware of NLMS Algorithm for Adaptive Noise Cancellation in 2010. 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 Patki evaluated the noise cancellation simulation outcomes. According to these outcomes only after 20 iterative operations, this algorithm can be become stable and provides stronger ability to boost SNR of weak signal as compared to LMS, NLMS, Variable size, sign LMS filter. Entirely 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 elimination of unsolicited 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 on adaptive filters. The proposed 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]. *RLS (Recursive Least Square) Algorithm* [1] algorithm attempts to minimize the cost function in Equation (1). In Equation (1), *k* 1is the time at which the RLS algorithm commences and is a small positive constant very close to, but smaller than 1. With values of 1 more recent input samples, this results in a scheme that places more emphasis on recent samples of observed data and tends to forget the past

.

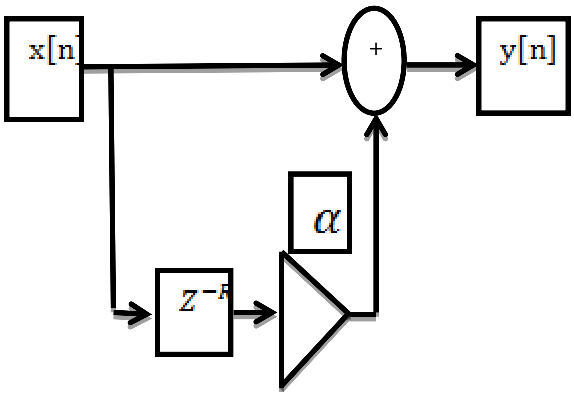
$$\zeta(n) = \sum_{k=1}^{n} \lambda^{n-k}\, e_n{}^{k}(k)$$

When compared to LMS algorithm, RLS algorithm offers a faster convergence and lower error at steady state. But, this RLS algorithm is more computationally complex and if proper design procedures are not followed, RLS algorithm may diverge away resulting in instability.

III. (a) ECHO [7]AND PROCESS OF ECHO CANCELLATION

Echoes are simply generated by delay components. The direct sound and a single echo appearing after R sampling periods later can be generated by the FIR filter as shown in Fig. 1.

$$H(Z) = 1 + \alpha Z^{-R}, |\alpha| < 1 \qquad (1)$$



The transfer function of the echo filter is given by Equation (1). In the above transfer function, the delay parameter *R* denotes the time the sound wave takes to travel from the sound source to the listener after bouncing back from the reflecting wall, whereas the parameter , with 1, represents the signal loss caused by propagation and reflection. There are two types of Echo – *Acoustic Echo* and *Hybrid Echo*. Hybrid Echo is generated in PSTN Network.

III. (b) ECHO CANCELLATION

Echo cancellation is the process of removing echo signals from a voice communication system in order to achieve quality audio awareness. The change of echo reduction began in the late 1950s, and continues today as new included landline and wireless cellular networks put additional requirement on the performance of echo cancellers. Echo cancellation consists of in first knowing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it is removed by 'subtracting' it from the transmitted or received signal. This technique is usually implemented on DSP's using adaptive filters.

IV. CONCLUSION

This paper has reviewed the Adaptive filtering algorithm 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 proposed algorithm is to achieve minimum mean square error at a higher rate of convergence with minor difficulty. 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.

V. REFERENCE

[1] Bernard Widrow,et. al. "Adaptive Noise canceling: Principles and Applications", Proceedings of the IEE,1975,Vol.63,No.12,Page(s):1692-1716.

[2] Saymon Haykins, and Thomas Kailath, Adaptive Filter Theory, Fourth Edition, Pearson Education.

[3] B.widrow and S.D.Stearns, adaptive Signal Processing. Englewood Cliffs, NJ:Prentice-Hall. 1985,p.474.

[4] Yuu-Seng Lau, Zahir M. Hossain, and Richard Harris, "Performance of Adaptive Filtering Algorithms", Proceedings of the Australian Telecommunications, Networks and Applications Conference(ATNAC), Melbourne, 2003.

[5] Hyun-Chool Shin, Ali H. Sayed, and Woo-Jin Song, "Mean Square Performance of Adaptive Filters using Averaging Theory", IEEE Signal Processing Letters, May 1999, Vol. 6, pp. 106-108.

[6] R. C. North, J. R. Zeidler, T. R. Albert, and W. H. Ku, "Comparison of Adaptive Lattice Filters to LMS Transversal filters for Sinusoidal Cancellation", Acoustics, Speech, and Signal Processing, 1992.ICASSP-92, March 1992, Vol. 4, pp. 33-36.

[7] Syed .A.Hadei and M.loftizad, "A Family of Adaptive Filter Algorithms in Noise Cancellation for Speech Enhancement", International Journal of Computer and Electrical Engineering, April 2010, Vol. 2, No. 2, pp. 307-315.

[8] V.R Vijaykumar,P.T. Vanathi,P.Kanagasabapathy, "Modified Adaptive Filtering Algorithm for Noise Cancellation in Speech Signals",Electronices and Electrical Engineering,Kanus:Technologija,2007.No.2(74). P.17-20.

[9] Ying He,*et. al*. "The Applications and Simulation of Adaptive Filter in Noise Canceling", 2008 International Conference on Computer Science and Software Engineering,2008,Vol.4,Page(s):1-4.

[10] Sanaullah khan,MArif and T.Majeed, "Comparison of LMS,RLS and Notch Based Adaptive Algorithms for Noise Cancellation of a Typical Industrial Workroom",8th International Mutitopic Conference,2004,Page(s):169-173.

[11] Yuu-Seng Lau, Zahir M.Hussian and Richard Harris, "Performance of Adaptive Filtering Algorithms:A Comparative Study", Austrailan Telecomminications,networks and Applications Conference(ATNAC),Melbourne,2003.

[12] Vinay K. Ingle, John G. Proakis, Digital Signal Processing Using Matlab, Third Edition, Cengage Learning, 2012

[13]https://ccrma.stanford.edu/~jos/pasp/Acoustic_Echo_Simulator. html

[14] Sanjit K Mitra, Digital Signal Processing, Third Edition, Tata McGraw-Hill Companies, 2006.

[15] Irina Dornean, Marina Topa, Botond Sandor Kirei, Marius Neag, "Sub-Band Adaptive Filtering for Acoustic Echo Cancellation", IEEE European Conference on Circuit Theory and Design, pp 810 - 813, 2009.

[16] Raj Kumar Thenua, S. K. Agrawal, "Hardware Implementation of Adaptive Algorithms for Noise Cancellation", International Journal of Information and Electronics Engineering, Vol. 2, No. 2, March 2012.

[17] Amit Munjal, Vibha Aggarwal, Gurpal Singh, "RLS Algorithm for Acoustic Echo Cancellation", Proceedings of 2nd National Conference on Challenges & Opportunities in Information Technology (COIT-2008), RIMT-IET, Mandi Gobindgarh, pp 299-303, March 29, 2008.

[18] Pushpalatha G.S, Shivaputra, Mohan Kumar N, "Spectrogram Study of Echo and Reverberation", Signal Processing and Information Technology, LNICST, Vol. 62, 2012, pp 210-213.

[19] Spectrogram version 5.0, Available: http://smedor.com/gram50.zip.

Mr. R. M. Potdar, (Phd Scholar) received B.E. Electronics, M.Tech.(Hons) Instrumentation & Control. He is having a total teaching experiences of 15 years and industrial experiences of 5 years. His interests are in Image Processing, Neural Network & Fuzzy Logic System Design. His specialization subjects are Adaptive Control System, Optimal Control System, Control System Design, Satellite communication and Optical Communication. He has published the papers in 8 international journals and 4 national journals and also attended 2 international conference and 14 national conferences. Also he is having Life Membership of Indian Society of Technical Education, India (ISTE).

Associate Professor Dr.Mekhram Meshram, received B.E. Electronics , ME Microwave & Radar. He is having a total teaching experiences of 24 year . His interests are in Development and Characterization of Ferrite Based Microwave Absorber, Investigation on Channel capacity of cellular communication network,. His field of specialization in Electronics and Telecommunication. He has published 22 research papers.

Professor Dr. Ramesh Kumar, received BE Computer Science, M.Tech Computer Science. He is having total teaching experiences of 20 year. . His interests are in development of a simulator for micro-optic-electro-mechanical system. His field of specialization in computer science and engineering. He has published the papers in 3 international journals and 5 national journals and also attended 5 international conference and 05 national conferences. He has published 42 research papers. Also he is having Life Membership of Indian Society of Technical Education, India (ISTE).

ISSN: 2278 – 909X
International Journal of Advanced Research in Electronics and Communication Engineering (IJARECE)
Volume 4, Issue 6, June 2015

Bhupeshwari Chandra , received BE Electrical and Electronics Engineering and M.Tech (Pursuing) from Bhilai Institute Of Engineering at durg. Her interest in communication sytem, power system and digital signal processing.