ISSN NO. 2320-5407

Journal homepage: http://www.journalijar.com

INTERNATIONAL JOURNAL OF ADVANCED RESEARCH

RESEARCH ARTICLE

REVIEW PAPER ON DIFFERENT TECHNOLOGY ON ECHO CANCELLATION

Ajjaiah H.B.M [1], Prabhakar V. Hunagund[2]

1. PhD Student, Dept. of PG Studies and Research in Applied Electronics, Gulbarga University, Karnataka, India.

2. Dept. of PG Studies and Research in Applied Electronics Jnana Ganga, ulbarga University Gulbarga - 585106, India.

Manuscript Info

Manuscript History:

Received: 22 May 2015
Final Accepted: 19 June 2015
Published Online: July 2015

Key words:

Acoustic Echo Cancellation (AEC), Nonlinear AEC, LMS, RLS, APA, Volterra Filters.

Corresponding Author

Ajjaiah H.B.M

Abstract

This paper presents a review on linear and non linear Acoustic Echo Cancellation (AEC) based previous research. The results of previous work on Echo Cancellation (EC) and the present research work of variable step size Least Mean Square algorithm is compared with adaptive filters. The performance of digital transmission system and communication network is degraded due to Inter Symbol Interference (ISI) in the linear and non linear AEC. Due to complexity of the loud speaker, amplifier and low quality in non linear AEC are not taken into considerations. The implementation of linear AEC is based on the assumption of linearity. The performance of echo cancellation algorithms are degraded due to nonlinearities in the acoustic path. Therefore non-linear AEC systems are evaluated and reviewed along with linear AEC.

INTRODUCTION

The echo cancellation system is used to suppress the unwanted signals which are emitted by the head phone and loud speaker. These signals will interrupt and delay the fundamental signal in the digital transmission and communication system. However, when fundamental signal is delayed by few milliseconds and reaches its output is called an echo signal. In mobile communication the echo signal leads to noise and makes conversion not possible. For this reason acoustic echo cancellation systems are used to remove the unwanted signal and provides better quality in communication network. In such cases, Adaptive filters plays very important role to minimise the echo into zero error. Adaptive filters uses two methods to tackle the Gaussian noise or the echo signal. One method is put a filters at the beginning of the system to suppress these noise, but this leads to adverse effect on linear filtering, and second approach is voltera series filtering.

2. Review on Acoustic Echo Cancellation System

In the history of Acoustic echo cancellation, the very first research work was done in 1967 by MM Sondhi [1]. In their paper they synthesized the first replica of echo, which accommodates all the parameter along the transmission line. There are two methods to suppress the noise,one is LMS (Least Mean square) and other is RLS(Recursive Least Square). In the advancement of adaptive filters second novel work was found by V.Reddy [2]. In the work frequency and time domain are considered for speed of convergence. The improved LMS and RLS were presented in 1983 for echo cancellation [3]. In 1987 S.Marcos and O.Macchi studied the time varying signal through adapative filters.

In 1990, Reference [4] presented a paper on exponential step NLMS by Makino Y.Kaneda . Then they applied to asynchronous echo canceller for channel length with time varying signal. In 1994[5], Makino Y.Kaneda proposes exponential steo size on Recursive Least square with double convergence speed. In 1999 Stenger presented a paper on Volterra filters for echo cancellation in real time. A Variable step size affine LMS proposed by C Paleologu et. Al.[7] in 2008. In 2009, Chia-sheng proposed on nonlinear echo cancellation system[8].

Here a tap variant step-size based on the MMSE criterion for second-order volterra filter was implemented to speed up the convergence rate an optimum time. For sparse system identification in network echo cancellation, in 2011, Mohammad Shams *et. al.*[9] proposed several normalized subband adaptive filter based algorithms. The algorithm shows good performance compared to full band adaptive filtering algorithms. A windowing frequency domain adaptive filter and an upsampling block transform preprocessing were presented by Sheng Wu *et. al.*[10] in 2011 to solve the stereo acoustic echo cancellation problems.

In 2012, a nonlinear system identification method for echo cancellation was proposed by Cristian Contan *et. al.*[11]. Adaptation of the system was made using a Normalized Least Mean Square algorithm. They combined the adaptive linear, volterra and power filters. Then the system is applied where several sources of nonlinearities exist, having the overdriven amplifier, the small loudspeaker at high volume and the room with different absorbent walls. Result showed that the overall system performs better than the best single adaptive filter. In the same year, Cristian Contan, *et. al*. [12] implemented nonlinear models for adaptive Volterra kernels using a new updating technique for unknown feedback paths identification a nonlinear echo cancellation system. Here a combination of adaptive linear, Volterra and power filters was used and compared for Echo Return Loss Enhancement in second-order Volterra filter Normalized Least-Mean-Square algorithm for kernel adaptation. The implemented system gives a superior convergence speed.The concept of stereophonic acoustic echo suppression (SAES) method without preprocessing in a open-loop teleconferencing systems was proposed by Feiran Yang *et. al*.[13] in 2012. They used Wiener Filter based echo suppression in the Short-Time Fourier Transform Domain for estimation of echo spectra impulse responses.

The spectral modification technique was used to remove the echo from the microphone signal. In 2012, a paper on State-Space Frequency-Domain Adaptive Filtering for Nonlinear Acoustic Echo Cancellation was presented by Sarmad Malik and Gerald Enzner [14].In this paper they have implemented adaptive acoustic echo cancellation in the presence of an unknown memoryless nonlinearity preceding the echo path. Result shows that the algorithm provides effective nonlinear echo cancellation in the presence of continuous double-talk, varying degree of nonlinear distortion, and changes in the echo path. In 2013, a general framework for designing echo cancellers for full-duplex communication systems with discrete multi-tone (DMT) modulation system in an arbitrary mixed domain were introduced by Neda Ehtiati and Beno Champagne [15]. They derived a new mixed-domain echo cancellation structure, achieved by a generic decomposition of the toeplitz data matrix at the transmitter in terms of arbitrary unitary matrices. This framework proposes a new canceller based on discrete trigonometric transformations. They have shown that this canceller has a faster convergence rate than the existing ones with similar complexity and is more robust. The eigen value spread of the input autocorrelation matrix is reduced using transform domain LMS algorithm to increase the convergence speed of the LMS algorithm. Performance in terms of convergence speed, steady-state MSE and robustness to power-varying inputs is further improved by using a new regularized transform domain normalized LMS (R-TDNLMS) algorithm proposed by S. C. Chan,*et. al.*[16] in 2013. In 2013, a new affine projection algorithm for acoustic echo cancellation and acoustic feedback cancellation applications with intermittent update of the filter coefficients was proposed by Albu, F. *et. al*.[17] where the update interval was determined according to the adaptation state. Result shows improved performance and reduced average computational complexity compared with other similar algorithms.

A delay less partitioned block frequency-domain adaptive filter (PBFDAF) algorithm was introduced in 2013 by Feiran Yang *et. al.*[18] to increase the convergence rate as well as computational efficiency. Here a delay compensation method is used to compensate the error path delay to speed up the convergence rate. In 2013, Stanciu, C. and Ciochina, S. [19] introduces a algorithm where RLS is combined with the dichotomous coordinate descent (DCD) algorithm to reduce the matrix inversion arithmetic complexity. The algorithm gave an improve performance in double talk like high disturbance situations. In 2013, Danilo Comminiello [20] introduced a new class of nonlinear adaptive filters based on hammerstein model. Such filters derived from the functional link adaptive filter (FLAF) model, defined by a nonlinear input expansion, which enhanced the representation of the input signal through a projection in a higher dimensional space, and a subsequent adaptive filtering. In order to give robustness against different degrees of nonlinearity, a collaborative FLAF is proposed based on the adaptive combination of filters. Such architecture allows achieving the best performance regardless of the nonlinearity degree in the echo path. In 2013, a Multiple Sub-filter (MSF) parallel structure based on multipath acoustic echo model was proposed by Ravinder Nath [21] using the basis that each sub-filter will compensate the echo contributed by each path of multipath acoustic channel.In 2014, an algorithm is given for more accurate estimates of the impulse response between each loudspeaker and microphone by Yellepeddi, A. and Florencio, D. [22]. The algorithm has array structure and it takes the sparsity of the reflections arriving at the array to give improved performance in echo cancellation applications, using both synthetic and real data.

Different Algorithm Review

Algorithm	Finding	Suitability	Citation
LMS	• Simplicity and convergence guaranty in stationary environments • If step size is small then it may take long time to converge.	For Linear AEC	[23], [24]
TDNLMS	• Exploits the decorrelation properties of transformations such as DFT, DCT and wavelet transform. • Improves the convergence speed of the conventional LMS.	For Linear AEC	[25]
RLS	• Faster convergence rates, but in general require computational resources, often too large for a real-time implementation. • Suitable for Non Stationary environment.	for Non Linear AEC	[3]
FRLS	• Fast version, i.e., the fast recursive least squares algorithms • It suffers from numerical problems when implemented with finite-word-length arithmetic's.	for Non Linear AEC	[26]
MSAF	• A multiband-structured subband adaptive filter algorithm • Speed up the convergence time of the NLMS algorithm.	for Non Linear AEC	[27]
AP	• Include LMS like complexity and memory requirements (low), and RLS like convergence (fast).	for Non Linear AEC	[7]
FAP	• Retain the good performance of AP algorithms but with a computational load close to that of LMS. • Offer, in the presence of correlated signals, convergence rates higher than LMS and tracking capabilities better than RLS.	for Non Linear AEC	[17], [30],
PBFDAF	• Reduce the computational load remarkably • The low-complexity methods that exploit the fast block convolution techniques in the DFT domain	for Linear AEC	[18]

Adaptive Voletera Filter	• Combat nonlinearities in the acoustic echo path of hands-free telephones, which are caused by low-cost audio components • Echo reduction improvement of 7 dB over conventional linear adaptive filters is achieved	for Non Linear AEC	[6]
Simplified Adaptive Voletera Filter	• Equivalent to the cascade structure for nonlinear AEC. • Has much fewer coefficients than the conventional direct form 2nd order VF does, while enjoys a better convergence performance compared to both the cascade canceller and the conventional 2nd order VF.	for Non Linear AEC	[12], [28],
Dynamic Impluse Response Method	• Focused on the nonlinearity affecting the IR, and propose a dynamic IR model for nonlinear system identification. • Decreases the computational costs of modeling and widely applicable in a real environment.	for Non Linear AEC	[29],
Orthogonalized Power Filters	• Standard approach in non linear environment • Combine time-domain orthogonalization methods with a DFT-domain implementation of power filters. • Model consists of the cascade of a linear filter, a memoryless nonlinearity, and a second linear filter.	for Non Linear AEC	[30]
Adaptive DFTdomain Volterra Filters	• Method for estimating the optimum number of second-order kernel diagonals of an adaptive Volterra filter. • An efficient version, requiring only minor additional computations as compared to a single Volterra filter.	for Non Linear AEC	[30]
Nonlinear AEC Based on Filter Combinatio n	• Based on the adaptive combination of linear and nonlinear echo cancellers • In this approach two or more adaptive filters adaptively combine their outputs obtaining a combined scheme. • Combination of a linear filter and a Volterra filter	for Non Linear AEC	[31]
Cascaded Nonlinear	• Greater robustness to changes in the acoustic channel than an existing power filter approach. • Model divides the LEM system into two main approaches- • Based on nonlinear post filtering to suppress the residual non-linear echo and Volterra series and non-linear adaptive filtering.	for Non Linear AEC	[32]

Nonlinear AEC Based on Optimum Step Size	• An optimum time- and tap- variant step-size control method for the second-order Volterra filter • Based on an optimum MMSE criterion between coefficients errors of real echo path kernel and adaptive coefficients. • Overcome the slow convergence rate.	for Non Linear AEC	[34],
PWL Approximation Method	• Based on Amplitude threshold decomposition for memoryless nonlinearity. AEC scheme for the cascade echo path model of memoryless nonlinearity and linear filter.	for Non Linear AEC	[8]
Volterra Filters with EQK	• EQK(Evolutionary Quadratic Kernels) • Exhibits a computational complexity suitable for practical implementations.	for Non Linear AEC	[35]
NRES Based on Partial Channel Modeling	• Nonlinear residual echo suppression (NRES) module is introduced to suppress nonlinear residual echo. • Efficient scheme with low computational complexity compared to adaptive Volterra filter.	for Non Linear AEC	[33]
R-TDNLMS	• Regularized transform domain normalized LMS algorithm • Improve convergence performance, steady-state MSE and robustness to power-varying inputs	for Non Linear AEC	[36]
Spectral Feature Based NRES	• A method for nonlinear residual echo suppression that consists of extracting spectral features from the far-end signal	for Non Linear AEC	[16]
Sliding Window Leaky Kernel AP algorithm	• Outperforms the linear APA, resulting in up to 12 dB of improvement in ERLE at a computational cost that is only 4.6 times higher	for Non Linear AEC	[37]
MSF Based on Multipath AEC Model	• Based on multipath acoustic echo model • MSF can track variations in the channel parameters of the multipath model faster than conventional echo canceller.	for Non Linear AEC	[39]
FLAF Model	• Structure is based on Hammerstein model. • Two FLAF-based architectures- the split FLAF and collaborative FLAF	for Non Linear AEC	[21]
FLAF Model	• Structure is based on Hammerstein model. • Two FLAF-based architectures- the split FLAF and collaborative FLAF	for Non Linear AEC	[20]
EMD Based Subband Adaptive Filtering	• Track the change of the room much faster than the normalized adaptive filtering structure. • Best ERLE • Enhance speech under various noise environments.	for Non Linear AEC	[39]
EVSSLMS	• The system does not diverge in a real value region and finally can converge, • The system can search different regions efficiently.	for Non Linear AEC	[40]

2. **Conclusion**

The Papers on both linear and non-linear echo-cancellation are studied based on algorithms used. A summary of algorithms used and their outcomes are given in tabulation form. The performance of the algorithms are evaluated in terms of various parameters such as ERLE, Misalignment, Misadjustment, Convergence rate, Robustness (disturbances and numerical), Computational requirements (operations and memory) and stability. It is shown that a further improvement is required in these parameters to improve the effective implementation of echo cancellation system

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