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Literature Review on Echo and Noise Cancellation Techniques.

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Abstract— ***Echo and Noise are the two main barriers of Communication these days. The signals get distorted during transmission and hence proper communication of the information is not possible. Thus a very efficient method is required to remove these two. An adaptive algorithm is an algorithm that changes its behavior based on information available at the time it is run. In this paper all the methods for Echo and Noise Cancellation using adaptive algorithms have been reviewed based on past researches.***

Index Terms— **Adaptive Algorithm , Echo Cancellation , Noise Cancellation.**

I. Introduction

In many speech communication applications, e.g., audio-conference and hands-free IP telephony, the received multi-microphone speech signals are corrupted by acoustic background noise as well as by echo signals. The noise and echo components significantly degrade the intelligibility of the desired signal, and restrict the performance of subsequent speech processing systems, e.g., speech coding and speech recognition systems. Therefore, efficient methods for joint noise reduction and echo cancellation are generally desirable. An adaptive filter is one that self-adjusts the coefficients of transfer function according to an algorithm driven by an error signal. The adaptive filter uses feedback in the form of an error signal to define its transfer function to match changing parameters. The adaptive filtering techniques can be used for a wide range of applications, including echo cancellation, adaptive channel equalization, adaptive line enhancer, and adaptive beam forming. In last few years, a lot of algorithms have been developed for eradicating the distortion from the signals. This paper presents analysis of algorithms and gives comparative study on various governing factors such as stability, computational complexity, filter order, robustness and rate of convergence.

Echo is the phenomenon in which delayed and distorted version of an original sound or electrical signal is reflected back to the source".There are two types of echo: **1**. Electrical echo: caused by the impedance mismatch at the hybrids transformer which the subscriber two-wire lines are connected to telephone exchange four wire lines in the telecommunication systems. **2**. Acoustic echo: caused by the reflection of sound waves and acoustics coupling between the loudspeaker and the microphone. In teleconference system, the speech signal from far-end generated from loud speaker after directing and reflecting from the wall, floor and other objects inside the room is receipt by microphone of near-end, as the result, this makes the echo that is sent back to the far-end. The acoustic echo problem will disturb the conversation of the people and reduce the quality of system. This is a common problem of the communication networks.In all practical situations, the received speech waveform contains some form of **Noise** component. The noise may be a result of the finite precision involved in coding the transmitted waveform (quantization noise), or due to the addition of acoustically coupled background noise. Depending on the amount and type of noise, the quality of the received waveform can range from being slightly degraded to being annoying to listen to, and finally to being totally unintelligible. The problem of removing the unwanted noise component from a received signal has been the subject of numerous investigations.

Adaptive algorithm starts its computation from prescribed initial condition and use information contained in the input data in order to estimate the weights of the filter. As the parameters of adaptive filters are updated from one iteration to next, it means that the parameters of the filter become information reliant and this provides that the adaptive filter in reality is a non-linear system. A system is said to be non-linear if it do not obeys the principle of superposition otherwise system is linear. Adaptive filters have extensive applications. They are used for adaptive noise and echo cancellation system identification, channel equalization, adaptive inverse system configuration and adaptive linear prediction.

1.1 General block diagram of the adaptive filters:

Here w represents the coefficients of the FIR filter tap weight vector, x(n) is the input vector sample, is a delay of one sample, y(n) is the adaptive filter output, d(n) is the desired echoed signal and e(n) is the estimation of the error signal at time n. The aim of an adaptive filter is to calculate the difference between the desired signal and the adaptive filter output, e(n). The error signal is fed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize a function of this difference, which is known as the cost function.

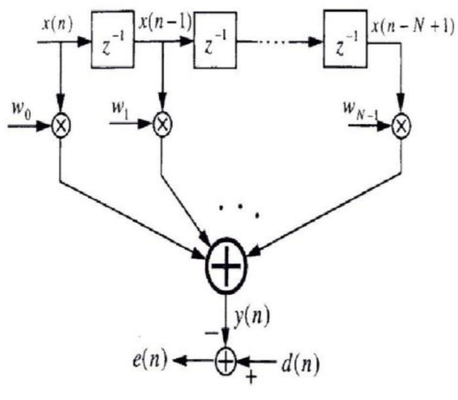

Fig-1: Block diagram of Adaptive filter.

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II. LITERATURE REVIEW

Upal Mahbub et al. proposed method, in which the echo cancellation is performed prior to the noise reduction operation. The major problem of getting a separate reference signal is tactfully overcome by using the delayed version of the echo and noise suppressed signal. Using such reference signal, the gradient based adaptive filter algorithm is derived to obtain an optimum Wiener-Hopf solution and thereby a modified LMS update equation is proposed. In the spectral subtraction based noise cancellation scheme noise floor is also updated in a regular fashion. The performance of the proposed method is measured in terms of ERLE (dB) and SDRI (dB). It is found that the proposed AENC can provide high values of these performance measures indicating an excellent echo and noise cancellation performance.[1]

Ms. Mugdha. M. Dewasthale et al., proposed an improved NLMS algorithm based on constant step size, constant filter order and a ratio of energy spectral density (ESD) of speech and reference noise signal for updating weights of adaptive filter is proposed. Proposed algorithm shows improved performance. From rigorous experimental analysis and testing conclude that proposed algorithm outperforms LMS and NLMS algorithms in terms of SNR, MSE and convergence time.[2]

Rachana Nagal et al. LMS and Normalized LMS have been demonstrated in frequency domain. Their corresponding results have also been shown after their implementation in both the time as well as frequency domain. Later their results have been compared on the basis of the SNR and the output graphs obtained. In the LMS algorithm, problem arises in deciding the value of the step size so, Normalized LMS algorithm have been implemented. NLMS gives a standard equation for the step size using two constants Ị and c. The results obtained in time domain have less SNR value and more noise in the output signal as compared to the one in the frequency domain. So, implementation of both LMS and NLMS in frequency domain raise the SNR around 8-9 times. This will provide good quality of reconstructed signal as compared to time domain in both the cases.[3].

Vaibhav Narula et al. paper presents a novel approach for Active Noise Cancellation through the technique of Adaptive filtering. In majority applications, algorithm used for practically implementing this concept is LMS (Least Mean Square) Algorithm. The work done here includes testing of LMS algorithm and its variants for Low and Mid Range frequency audio environments. The experimental results demonstrated in this paper are quite beneficial in identification of specific LMS variant corresponding to particular category of signals in terms of application.[4]

John H˚akon Husøy et al . In this paper developement of a normalized circulantly pre-conditioned LMS adaptive filter. Thus, just as the NLMS algorithm improves on the LMS algorithm incorporating a signal dependent adaption parameter, the NCPLMS does the same for the PLMS algorithm. This considerably simplifies the sometimes tricky trade-offs involved in selecting an appropriate adaption parameter, μ, that is encountered in non-normalized algorithms. In particular a solution to the problem of selecting an optimum circulant preconditioner for the NCPLMS algorithms. Through experiments we have demonstrated that, through the use of NCPLMS, we can successfully shift the superior convergence behavior observed for NLMS with white input signals to input signals having user-determined auto-correlation properties given by some set of autocorrelation matrices. This may be useful in applications where some a priori knowledge of input autocorrelation properties is available. We supported this through some simulations in which the NCPLMS algorithms was successfully applied in a line echo cancellation setting.[5].

V´ıtor H. Nascimento et al. Adaptive combination of filters is a simple, but effective method to circumvent the compromises that AFs present. However, if a combination of two filters is used to alleviate the trade-off regarding step-size selection, its computational cost is approximately twice that of each component. In this paper, we have shown that, under certain conditions, it is possible to reduce the computational cost of the combination simply that our proposal would obtain a suitable performance since the undermodeling carried out by the fast component would have no negative consequences in the performance of the combination, but would speed up the convergence and reduce the computational cost.[6].

Botond Sandor Kirei et al. presents a scheme for the acoustic echo canceller. Because of its simplicity, the LMS algorithm is the most popular adaptive algorithm. However, the LMS algorithm suffers from slow and data-dependent convergence behavior. The NLMS algorithm, an equally simple, but more robust variant of the LMS algorithm, exhibits a better balance between simplicity and performance than the LMS algorithm. Due to its good properties the NLMS has been largely used in real-time applications. The VSLMS algorithm has a better performance than NLMS at a slightly larger cost (N multiplications more). The variable step size allows stability when dealing with non-stationary signals. On the contrary the VSNLMS is rather disappointing: weak performance at a higher complexity. The RLS algorithm has the greatest attenuation of all studied algorithms and converges much faster than the LMS algorithm. Due to the large number of multiplications it is rather costly to be implemented. Taking into consideration both the average attenuation (convergence speed) and the number of multiplications (hardware complexity), the best choices for real time acoustic echo cancellation are the NLMS and the VSLMS algorithms.[7].

E.Hari Krishna et al. HOT based LMS adaptive filtering for acoustic echo cancellation from audio signals has been presented. The Convergence & Computational complexity analysis of different adaptive algorithms shows that HOT –LMS is efficient. The computed ERLE measure and SFM indicated that HOT - LMS is superior in cancelling echo in audio signal. As demonstrations in this paper showed follows, that HOT-LMS significantly reduced the computational burden, the authors are presently working on VLSI implementation issues of the presented algorithm by exploiting pipe lining architecture.[8].

Lin Bai et al. adaptive algorithm for ANC to filter a noisy speech signal is proposed in this paper. The new algorithm further relaxes the constraint in the CS-LMS algorithm. Instead of forcing the a posteriori error to be as smooth as possible, the new algorithm attempts to minimize the estimation error of the a posteriori error and the estimation is obtained using the concept of Taylor's expansion. CS-LMS can be seen as a special case of the new algorithm with $k = 0$. Analysis and simulation results show that the new algorithm can obtain better EMSE performance than NLMS and the new algorithm can get performance improvement by properly increasing the order k.[9]

Abhishek Tandon et al. proposed an algorithm for echo cancellation in multi-channel systems by employing a partial updating of a predetermined set of the filter coefficients and introducing a low-pass filter in the feedback path. The new algorithm has been shown to provide a good performance with a low computational complexity.[10]

Sabri M. Hanshi et al. new AEC system framework has been proposed that can handle the mismatch in the sampling rate of the input signals and generate a balanced sampling rate output. [11].

Subhash C. Tanan et al. a novel method for acoustic echo and noise cancellation in a generalized sidelobe canceler framework is described. The primary contribution of this work is the development of multichannel adaptive Kalman filter (MCAKF) in a modified generalized sidelobe canceler (MGSC) framework. Additionally, in this work both the near end speech signal and noise is assumed to be unknown. In the proposed method speech acquired by a microphone array is subject to adaptive beamforming using MVDR method. On the other hand a blocking matrix filter is used to attenuate the near end speech signal while passing both the noise and residual echo. A MCAKF is developed in this context to also estimate the noise andresidual echo. Hence, a difference of MCAKF output and the adaptive beamformer (ABF) output gives an estimate of the near end speech signal. The performance of proposed method is evaluated using subjective and objective measures on the ARCTIC database. Distant speech recognition experiments are also conducted on the ARCTIC database. The proposed method gives reasonable improvements both in terms of perceptual evaluation and distant speech recognition.[12]

III. CONCLUSION

The main problem in echo and noise cancellation occurs in choosing the right type of filter algorithm that gives best result in all the parameters like convergence time, stability, complexity and must also be cost effective. Their performance measure must give best result in terms of ERLE and SNR. In traditional filter used earlier LMS was the most popular due to its low complexity but its stability is less and convergence time is more. So many new algorithms have come up which can be employed in filtering process having greater stability and still not increasing the complexity of overall system. It must also be able to give clear speech by removing echo and noise present in it.

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