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A Review on Acoustic Echo Cancellation Techniques

Pankush Singla[#1], Sandeep Kaur[#2], Sukhpreet Singh[*3]

[#] *Lecturer in C.S.E Department, MIMIT*
Malout (Punjab), India

[*] *Sukhpreet Singh*
Assistant Professor in C.S.E Department, Guru Kashi University
Talwandi Sabo (Punjab), India

Abstract-- **Acoustic echo is the noise which is created when sound waves are reflected by the wall or other things of the room. This acoustic echo get mixed with the original speech signal and degrades the quality of speech signal. The main objective is the cancellation of this acoustic echo for better communication and to provide an echo free environment. There are so many techniques which are used for removing of acoustic echo from the original speech signal. Different type of filters are used in each technique for filtering echo signal from speech signal. In this paper, we are making a review of some techniques which are related to removing echo signal in different type of environments.**

***Index Terms*— Acoustic echo, Signal, Filter, Communication.**

I. INTRODUCTION

In hands-free telephony and in teleconference systems, the main aim is to provide a good free voice quality when two or more people communicate from different places.

The problem often arises during the conversation is the creation of acoustic echo. This problem will cause the bad quality of voice signal and thus talkers could not hear clearly the content of the conversation, even thought lost the important information. 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 (figure I-1), 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.

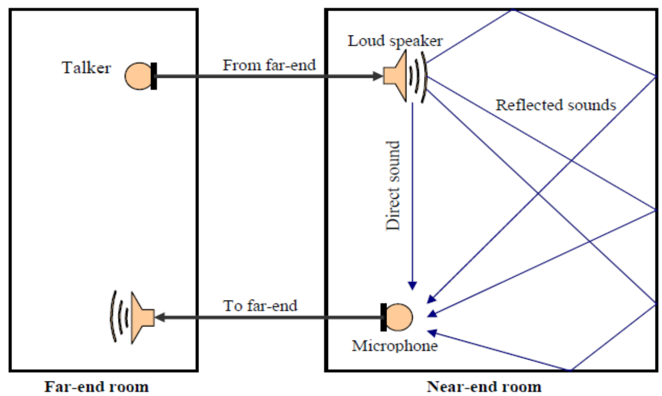

Figure 1.1 A teleconference system with echo paths of room.

Two main characteristics of echo are reverberation and latency. Reverberation is the persistence of sound after stopping the original sound. This sound will slowly decay because of the absorption by the materials constructing the environment. Latency or delay is the different time of the signal between the transmitter and receiver. In the case of teleconference system, the sound is generated from loud speaker and received by microphone, the delay can compute base on the distance between them (i.e., the length of the direct sound).

Delay = distance/speed of sound

Acoustic Echo Cancellation (AEC):
To handle with the acoustic echo problem above in teleconference systems, one can use voice switches and directional microphones but these methods have placed physical restriction on the speaker. The common and more perfective method is implementing the Acoustic Echo Cancellation (AEC) to remove the echo. AEC enhances greatly the quality of the audio signal of the hands-free communication system. Due to their assistance, the conferences will work more smoothly and naturally, keep the participants more comfortable.
Some echo cancellation algorithms are used for this purpose. All of them process the signals follow the basic steps below:

1. Estimate the characteristics of echo path of the room.
2. Create a replica of the echo signal.
3. Echo is then subtracted from microphone signal (includes near-end and echo signals) to obtain the desired signal. Adaptive filter is a good supplement to achieve a good replica because of the echo path is usually unknown and time-varying. The figure

below illustrates about three step of the AEC using adaptive filter.

II. LITERATURE REVIEW

Peter L. Chu "Weaver SSB Sub band Acoustic Echo Canceller".

A Weaver SSB sub band structure is used to implement an acoustic echo canceller. The structure has 29 bands of 250 Hz width covering the audio range from 0 to 7 kHz. The Weaver structure lowers each band pass region to baseband, allows for oversampling to eliminate aliasing components and is computationally efficient. The sub sampled components are purely real, as compared to the complex components found in some other sub band schemes. The adaptive filter update algorithm is a variant of the block NLMS. The double-talk divergence, echo suppression, and noise in algorithms all fully exploit the band pass structure to achieve performance difficult to attain in full-band or two-band acoustic echo cancellers. The acoustic echo canceller has been extensively field tested and has been shown to be robust[2].

Yang-Won Jung, Ji-Ha Lee, Young-Cheol Park*, Due-Hee Youn" A New Adaptive Algorithm For Stereophonic Acoustic Echo Canceller".

Stereophonic sound becomes more important in a growing number of hands-free applications where spatial realism is demanded. Such hands-free systems need stereophonic acoustic echo cancellers to reduce echoes that result from coupling between loudspeakers and microphones. In this paper, we propose a new adaptive algorithm for stereophonic acoustic echo canceller based on human auditory properties and affine projection (AP) algorithm. The proposed algorithm employs a pre-processor generating speech-like noise to de-correlate the input signals without degrading perceptual speech quality. The de-correlation is based on the masking patterns of the human auditory system: By showing that the AP algorithm can be represented by a vector update as a combination of the Gram-Schmidt (GS) orthogonalization followed by the normalized LMS (NLMS) algorithm, the proposed adaptive algorithm integrates the pre-processor into the adaptive algorithm. Subjective listening test and computer simulation show the effectiveness off the proposed algorithm[3].

Simon Doclo, Marc Moonen and Erik De Clippel "Combined Acoustic Echo and Noise reduction using GVSD-based optimal filtering".

This paper describes two schemes for combining acoustic echo and noise reduction using a GSVD-based optimal filtering technique. The GSVD-based filtering technique is a signal enhancement technique which has recently been proposed for noise reduction in multi-microphone speech signals. In many speech communication applications however also a far-end echo source is present. Therefore a combined echo and noise reduction scheme is needed. The first scheme combines a standard multi-channel adaptive echo canceller with the GSVD-based noise reduction technique. The second scheme incorporates the far-end echo reference directly into the GSVD-based signal enhancement technique without cancelling the echo in every microphone signal. The two different schemes are compared with regard to performance and computational complexity[4].

Siow Yong Low and Sven Nordholm "A blind approach to joint noise and Acoustic Echo Cancellation".

This paper introduces a new scheme which combines the popular blind signal separation (BSS) and a post-processor to jointly suppress noise and acoustic echo. The new L element structure uses the BSS as a front-end processor to spatially extract the target signal from the interference (noise and echo). Statistical measures are then employed to select the target signal dominant signal from the BSS outputs. The remaining $L - 1$ BSS outputs (noise and echo dominant) and the existing far-end line echo are then used as the reference signals in an adaptive noise canceller (ANC) to temporally enhance the target signal. The novel structure bypasses the need for any a priori information whilst compensating the separation quality of the BSS temporally. Real room evaluations demonstrate the efficacy of the scheme in both noisy double-talk and non double-talk situation[7].

L. Romoli, S. Cecchi, L. Palestini, P. Peretti and F. Piazza "A novel approach to channel decorrelation for stereo Acoustic Echo Cancellation based on missing fundamental theory".

De-correlation is a well known issue in the context of Stereophonic Acoustic Echo Cancellation: it is related to the problem of uniquely identifying each pair of room acoustic paths, due to high inter channel coherence. In this paper, a novel approach to de-correlate a stereo signal based on the missing fundamental phenomenon is proposed. An adaptive algorithm is employed to track the behavior of one of the two channels, ensuring a continuous de-correlation without affecting the stereo quality. Several results are presented comparing our approach with Masked Noise injection method in terms of Magnitude Square Coherence, Itakura Saito measure and Convergence Speed of adaptive filters in order to confirm the validity of the proposed approach[13].

III. CONCLUSION

In today's technology, the main area of interest of engineers is removal acoustic echo from the original speech signal. For better communication between various speakers, we need to provide a good environment which is free from disturbance so that the quality of communication is improved. For achieving this goal, we need these methods which filter this acoustic echo from speech signals. This paper review some methods relate to this technology for removing of this unwanted echo signal form original voice signal. This paper will be very helpful for the readers to get familiar with acoustic echo and methods for its removal so that new inventions can be done to improve these methods.

REFERENCES

[1] S.Makino and Y.Kaneda "Acoustic Echo Canceller Algorithm Based On The Variation Characteristics Of A Room Impulse Response" 1990.
[2] Peter L. Chu "weaver SSB Sub band Acoustic Echo Canceller" 1993 IEEE.
[3] Y.Jung, J.Lee, Y.Park, Due-Hee Youn" A New adaptive algorithm for stereophonic Acoustic Echo Canceller" 2000 IEEE.
[4] D.Simon, M.Moonen and C.Erik "Combined Acoustic Echo and Noise reduction using GVSD-based optimal filtering" 2000 IEEE.
[5] S.Gustafsson et al., "A Psychoacoustic Approach to Combined Acoustic Echo Cancellation and Noise Reduction", IEEE Transactions on speech and audio processing, VOL. 10, NO. 5, JULY 2002.
[6] H.Michael "Acoustic Echo Cancellation Using Digital Signal Processing", The School Of Information Technology And Electrical Engineering, The University Of Queensland, November 2003.
[7] Y.Siow and N.Sven "A blind approach to joint noise and acoustic echo cancellation" ICASSP 2005 IEEE.
[8] P.Åhgren, "Acoustic Echo Cancellation and Doubletalk Detection Using Estimated Loudspeaker Impulse Responses", IEEE Transactions on speech and audio processing, VOL. 13, NO. 6, NOVEMBER 2005.
[9] S.Christian et al,"Low-Complexity Adaptive Filtering Implementation for Acoustic Echo Cancellation", IEEE Region 10 Conference TENCON 14-17 Nov. 2006
[10] L.Fredric et al., "Efficient Multichannel NLMS Implementation for Acoustic Echo Cancellation", Hindawi Publishing Corporation EURASIP Journal on Audio, Speech, and Music Processing, Volume 2007, Article ID 78439.
[11] G.Cristina et al., 'Echo Cancellation Using The Lms Algorithm', U.P.B. Sci. Bull., Series C, Vol. 71, Iss. 4, 2009, ISSN 1454-234x.
[12] H.Nguyen et al,"Implementation of the LMS and NLMS algorithms for Acoustic Echo Cancellation in teleconference system using MATLAB", Växjö University ISSN 1650-2647,December 2009.
[13] L. Romoli, S. Cecchi, L. Palestini, P. Peretti and F. Piazza "A Novel approach to channel decorrelation for stereo acoustic echo cancellation based on missing fundamental theory" ICASSP 2010 IEEE
[14] Z.Chen, "Design and Implementation on a Sub-band based Acoustic Echo Cancellation Approach", (IJACSA) International Journal of Advanced Computer Science and Applications, Vol. 2, No. 6, 2011.
[15] S.Dhull et al., "Performance Evaluation of Adaptive Filters Structures for Acoustic Echo Cancellation", International Journal of Engineering (IJE), Volume (5) : Issue (2) : 2011

Pankush Singla, Lecturer,
C.S.E Department, MIMIT Malout
Qualification: B-Tech(C.S.E) from GTB College Chhapianwali.,
M-Tech(Comp. Engg.) from Punjabi University Patiala.

Sandeep Kaur, Lecturer,
C.S.E Department, MIMIT Malout
Qualification: B-Tech(C.S.E) from Guru Gobind Singh College of Engg. & Tech., Talwandi Sabo.,
M-Tech(Comp. Engg.) from Punjabi University Patiala.

Sukhpreet Singh, Assistant Professor,
C.S.E Department, Guru Kashi University,Talwandi Sabo
Qualification: B-Tech(C.S.E) from Chitkara Instt. of Engg. & Tech., Rajpura,
M-Tech(Comp. Engg.) from Punjabi University Patiala.