International Journal of Engineering and Advanced Technology (IJEAT)
ISSN: 2249-8958 (Online), Volume-5 Issue-5, June 2016

Decorrelation By Principal Component Analysis For Multi Channel Acoustic Echo Cancellation System

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Abstract—In multi-channel acoustic echo cancellation (MAEC) system, thenon-uniqueness problem and misalignment problem occurs due to the correlation between the reference signals. It could affect convergence performance of the adaptive filtering. So many methods are proposed to get minimum error rate. In this paper, fuzzy logic is used to get minimum error function. The decorrelation is applied through the PCA method. The adaptive fuzzy fusion algorithm improvises, update and check operators obtain optimal solution for defined objective function. To obtain better solution the control parameters are adjusted. It achieves a superior performance in the echo reduction gain and offers the possibility of frequency selective decorrelation to further preserve the sound quality of the system. Simulationresult for the proposed algorithm has shown a significant improvement in convergence rate compared with existing system.

Index Terms—Multi channel AEC, non-uniqueness problem, Misalignment problem, Principal component analysis.

I. INTRODUCTION

In recent years speaker phones and hand free cellular phones have been used widely for audio and video conferencing. A speaker phone and hand free cellular system allows full duplex communication. Full duplex communication means voice on both end of the line is transmitted continuously. The speech from far end caller is directed to speaker phones and then repeat itself by bouncing off the inside surface of the room. This repetition of sound is called an echo. These echo are picked up by the near end microphone creating a feedback loop where far end caller hears his own sound. To solve this problem it uses AEC to stop the feedback and allow full duplex communication. This shows the principle of the echo canceller for a single channel. In this $y[n]$ is the far end signal $r[n]$ is the undesired echo and $x[n]$ is the near end signal.The near end signal is superimposed with the undesired echo on port D.The received far end signal is available as reference signal for echo canceller and it generate a replica of the echo.i.e.,$r^{'}[n]$.This replica is subtracted from the near end plus echo to get the transmitted signal. If $r^{'}[n]$ is ideal for $r[n]$,$e[n] = r[n] – r^{'}[n]$ will be very small.

Manuscript published on 30 June 2016.
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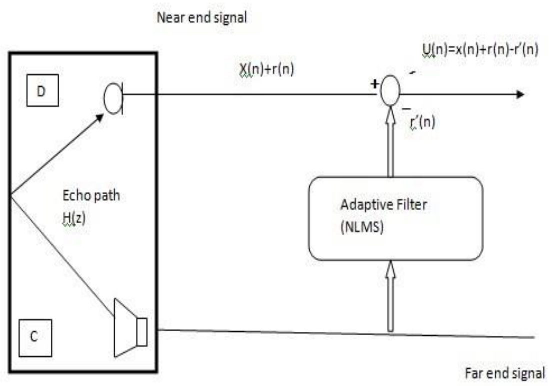

Figure 1. Block diagram of acoustic echo cancellation system

In multi-channel acoustic echo cancellation there is number of transmission channels are available so that it must model more echo path. The reason for this is because the impulse response from the far end speaker to each individual microphone also has to be modelled

A number of pre-processing methods have been proposed to decor relate stereo audio signals in order to enable an un-ambiguous identification of each echo path and to thus ensure robustness to changing sound source locations. Two-channel pre-processing approaches used in [1] can give high performance but Convergence speed is low. In [2] it explain the main problems encountered due to the strong cross correlation between the two input signals and we propose a new solution based on non-linear transformations to overcome these problems. A non-linear transformation technique is used here. It improves both the condition number of the covariance matrix and the misalignment. It is not efficient. The [5] presents a novel decorrelation procedure by frequency domain resampling in sub band. The new procedure expands on the idea of resampling in the frequency domain that efficiently and effectively alleviates the non-uniqueness problem for a multichannel acoustic echo cancellation system while introducing minimal distortion to the signal.

In, Stereophonic Acoustic Echo Cancellation [3] establish the relationship between inter channel coherence of the two input signals and condition of the corresponding covariance matrix for stereo acoustic echo cancellation application. Multichannel acoustic echo cancellation based on residual echo enhancement [4] The residual echo enhancement (REE) procedure proposed is able to improve the acoustic echo cancellation performance in a very noisy acoustic mixing environment by utilizing the natural learning ability of the least-mean square (LMS) algorithm without precise estimation of the signal statistics.

We demonstrate in this paper that the technique can also be applied effectively in a multi-channel AEC setting to indirectly assist in the recovery of lost AEC performance due to the non-uniquenessproblem. It is computationally feasible but Measurement of coherence after decorrelation cannot be considered.

In [5] investigate several types of non-linearities used for the unique identification of receiving room impulse responses in stereo acoustic echo cancellation.The perceptual degradation is studied by psychoacoustic experiments in terms of subjective quality and localization accuracy in the medial plane.It provide good speech quality,implementation is difficult.Inter-channel decorrelation by sub-band resampling in frequency domain[6] presents a novel decorrelation procedure by frequency domain resampling in sub-bands. The new procedure expands on the idea of resampling in the frequency domain that efficiently and effectively alleviates the non-uniqueness problem for a multi-channel acoustic echo cancellation system while introducing minimal distortion to the signal.

In [7], it investigate in detail the decorrelation by resampling technique and a proper design strategy for SBR, which permits finely tuned, frequency specific control of the trade-off between decorrelation, measured in terms of the (magnitude squared) coherence, and audio quality, evaluated in terms of objective speech quality measures and a subjective listening test. It then derive new closed-form expressions to demonstrate how resampling affects the misalignment for different types of reference signals, i.e., a white Gaussian noise or a speech signal, with or without the far-end room impulse response. Following these analyses, it provide a novel, theoretically justifiable, and perceptually motivated SBR strategy for achieving fast AEC convergence with a minimal signal distortion.

The paper is organized as follows. Section II gives problem formulation. Section III introduces the methodology. Section IV gives conclusion.

II. CHARACTERISTIC ANALYSIS

For the simplicity of the MAEC the echo paths that are linked with one of the microphones are taken.The microphone signal is given by[7]

$$y[n] = Zn[n] + v[n] + \sum_{p=1}^{P} X_{dp}[n] \tag{1}$$

where$y[n]$ is the microphone signal,$Z_n[n]$ is the near end speech,$v[n]$ is the near end noise,P is the number of loudspeakers.$dp[n] = hTp, Kxp, K[n]$is the acoustic echo generated by thepthloudspeaker.

P adaptive filter are required to cancel the P echo signal,and the error signal is given by [7],

$$e[n] = y[n] - \sum_{p=1}^{P} w_{p,L}^{T}[n]x_{p,L}[n] \tag{2}$$

Where $wp, L[n] = [wp, 0[n], \dots, wp, L-1[n]]T$ is the adaptive filter coefficient vector for pthecho path.Here the actual length of near end room impulse response (K) is the different from that of adaptive filter coefficient vector (L).The minimum mean square error can be computed by solving the normal equation in [7],

$$R_{XPL}[n]W_{PL}[n] = r_{yXPL}[n] \tag{3}$$

Where $R_{XPL}[n] = E\{X_{PL}[n]X_{PLT}[n]\}$ } with E is the expectation operator. The Figure 2 shows the stereo acoustic echo cancellation system with P=2.

There are mainly two problems associated with multi-channel AEC system,non-uniqueness problem and misalignment problem. The non-uniqueness problem and misalignment problem arises during the multi-channel AEC due to the correlation between the reference signals that degrades the performance of adaptive filtering algorithm[9]

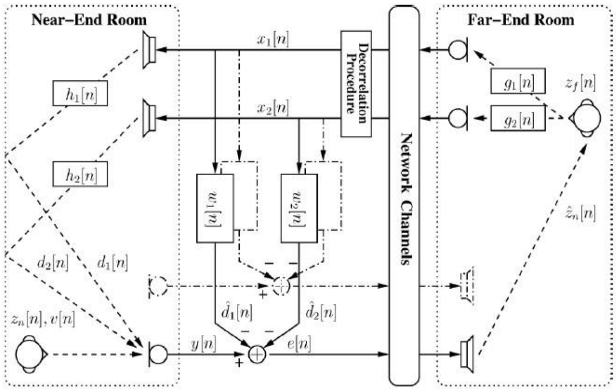

Figure 2. Block diagram of a stereo acoustic echo cancellation system [7]

A. Non-Uniqueness problem

Non uniqueness problem raises if the audio stream originate from the same source. This means that the normal equation to be solved by the adaptive filter is singular. So the echo canceller cannot provide a unique echo path solution. The filter coefficient does not converge to the true echo path impulse response. The fundamental problem that the two audio channels may carry linearly related signals which in turn may cause normal equation to be solved by the adaptive algorithm singular.i.e., there is no unique solution to the equation but an infinite number of solution and all these solution depends on the transmission room.

Let gp, K be the pth far end room impulse response vector, $zf[n]$ is the far end source signal and $Zn[n] = v[n] = 0$.The pthreference signal is given by [7]

$$xp[n] = g_{p,KT} Z_{f,K}[n] \tag{4}$$

Consider L ≥ K and the room impulse response is linear and time invariant, then the reference signals Satisfy,

$$g_{2,K}^{T} X_{1,K}[n] = g_{1,K}^{T} X_{2,K}[n] \tag{5}$$

That is the solution for Multi-channel AEC depends not only on the near end room impulse response but also the far end room impulse response. The adaptive solution must vary when there is a change in far end room impulse response. This happens due to the multiple users in the room. Whenever the user turns to talk, the room impulse response is changing which causing degradation in the system performance.

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A. Misalignment problem

The mismatch between the adaptive filter and the actual room impulse response is termed as misalignment which is defined as in [7],

$$\Xi = \frac{\|h_{PL} - w_{PL}\|_2^2}{\|h_{PL}\|_2^2} \quad (6)$$

The length of the adaptive filter can be much lesser than the length of the near end room impulse response.

The coherence gives the amount of correlation between the two reference signals in the frequency domain. The power spectral density(PSD) and the cross spectral density(CSD) is given by [7],

$$S_{x_i x_j}[k] = \sum_{n=-\infty}^{\infty} r_{x_i x_j}[n] \exp^{-j\pi kn/L} \quad (7)$$

Where $rxixj$be the auto correlation and cross correlation for $i = j$ and i 6= j.(7) represent PSD when $i = j$ otherwise represent CSD.The coherence is defined by [7],

$$C_{x_1 x_2}[K] = \frac{|S_{x_i x_j}[k]|^2}{S_{x_1 x_1}[k] S_{x_2 x_2}[k]} \quad (8)$$

III. METHODOLOGY

A decorrelation process is needed before the near end playback to improve the signal quality by reducing the non-uniqueness problem. Principal component (PC) method is used to decorrelate the reference signals. PCmethod, that uses an orthogonal transformation to convert a set of observations on possibly correlated variables into set of values of linearly uncorrelated variables called principal components. The number of principal components are less than or equal to the original variables. This transformation defined in such a way that the first principal component has the largest possible variance in the data as ,and each succeeding component in turns has the highest variance possible under the constraints that it is orthogonal to the preceding components. The resulting vectors are an uncorrelated orthogonal basis set. The principal components are orthogonal because they are eigenvectors of the covariance matrix, which is symmetric PC method is sensitive to the relative scaling of the original variables

Frequency domain resampling is to interpolate across frequency rather than across time[7],with appropriate expansion or compression, to reduce the computation through Fast Fourier Transform(FFT)

The step involved these are first, partitioned the input stereo audio signal to a limited range. Echo is created using a chebyshev filter. Right audio signal is combined with this echo. Then the total signal is left audio signal, right echoed signal and the near end noise. An adaptive filter with NLMS algorithm is used to remove this echo. Finally the step size of the adaptive filter made to adaptive. Figure 3 shows the block diagram of the proposed method.

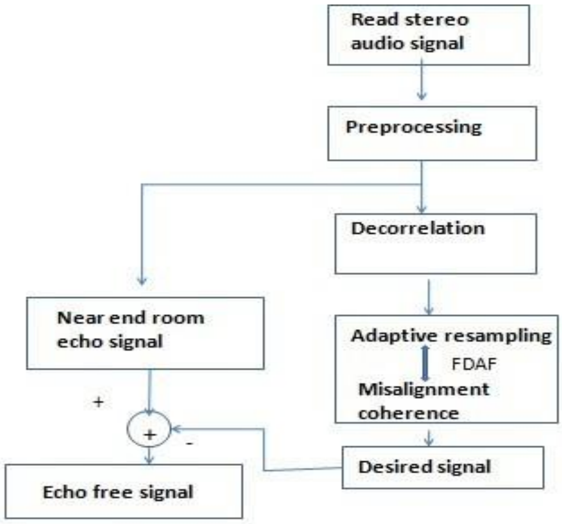

Figure 3. Echo Return Loss Enhancement Graph

Because of the adaptive nature of the adaptive filter step size the better result with maximum Echo Returns Loss Enhancement (ERLE) is obtained.

The Normalized Least Mean Square (NLMS) Algorithm is one of the adaptive algorithms usually used in echo cancellation system. By using NLMS algorithm output of adaptive filter can be computed as

$$y[n] = WT(n)X(n) \quad (9)$$

Where $X[n] = [x(n), x(n-1) \ldots . x(n-N+1)]$ is the input vector and $W(n) = w0(n) \ldots wN-1(n)$ is the weight vector.The error signal is

$$e(n) = d(n) - y(n) \quad (10)$$

Next filter coefficient is given by,

$$W(n+1) = W(n) + \mu(n)e(n)x(n)/xT(n)x(n)$$

(11) Where$\mu(n)$ is the step size factor.

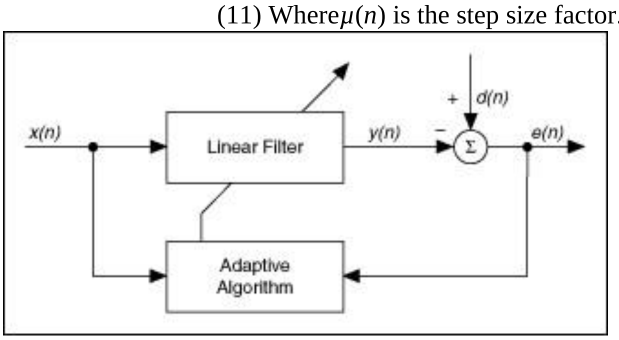

Figure 4. Adaptive filter

A. Fuzzy Logic System

Adaptive fuzzy fusion algorithm is an effective approach to handle non-linear signals such as stereo audio signals. The proposed procedure expands on the idea of fuzzy based adaptive resampling in the frequency domain, to directly assist the frequency domain adaptive filtering algorithm[8].Figure shows the Fuzzy Logic Control(FLC) system block diagram. The value of step size is chosen such that

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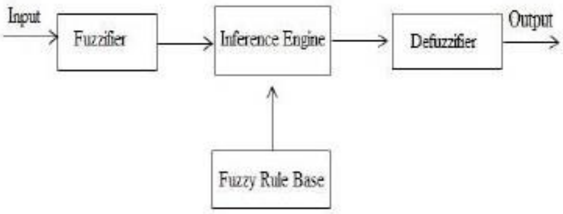

Figure5. Block diagram of FLC [8]

adaptive filtering algorithm converges to optimum filter tap weights. If the value of step size is chosen to be large, the filter coefficient may change by a large value and at the second instant the coefficient may keep oscillating with a large variance about the optimal coefficient weights. if step size is chosen to be too small, time required to converge to the optimal coefficient values will be too large. The upper bound of step size is given by[8]

$$0 < \mu < 1/\lambda max \qquad (12)$$

Where λ is convergence factor and λmax largest eigenvalue of the auto correlation matrix $R = E\{XXT\}$.

IV. SIMULATION RESULT

The multi-channel acoustic echo cancellation performance is evaluated by using adaptive fuzzy fusion algorithm which gives the optimum result compared with the existing system. Usually echo is measured in terms of ERLE.A higher value of ERLE means minimum amount of echo and smaller value of ERLE means maximum amount of echo. Figure 6 shows the output of the multi-channel acoustic echo cancellation system. Figure 7 shows the ERLE graph of the proposed method which gives the optimum value compared with the existing system.

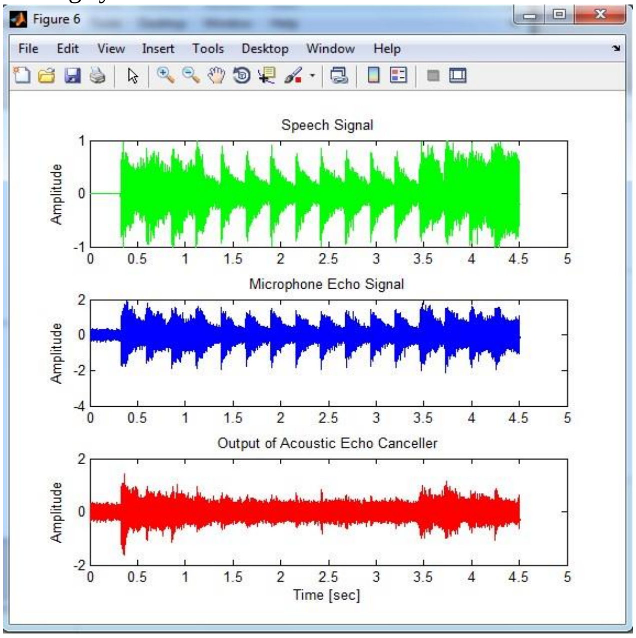

Figure6. Output of multi-channel acoustic echo cancellation system

V. CONCLUSION

A new theory to Multi channel acoustic echo cancellation problems are developed using the adaptive fuzzy fusion algorithm .Adaptive fuzzy fusion algorithm uses the a FLC system, is a variable step size adaptation algorithm. From the experimental results the echo removal using the proposed method was found that more quality than other existing algorithms. This is due to the maximum ERLE with minimum error rate.

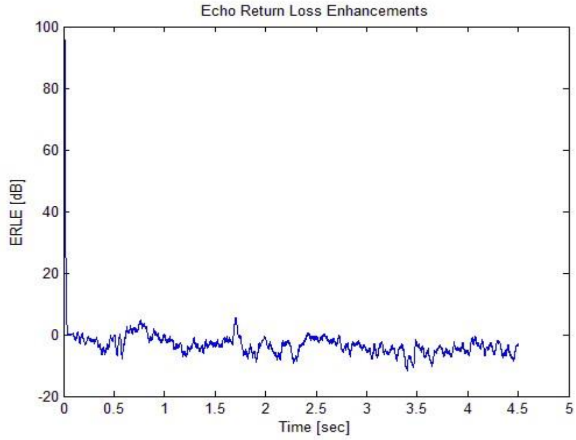

Figure 7. Echo return loss enhancement graph

REFERENCES

1. J. Herre, H. Buchner, W. Kellermann,"Acoustic echo cancellation for surround sound using perceptually motivated convergence enhancement,"IEEE ICASSP.,2007,pp.17-20. .
2. J. Benesty, D. R. Morgan, and M. M. Sondhi, 'A Better Understanding and an Improved Solution to the Specific Problems of Stereophonic Acoustic Echo Cancellation ," inIEEETranc.speech audio process,vol.6,no.2,pp 156-165,Mar.1998.
3. W. H. Khong, J.Benesty,andP.A. Naylor,"Stereophonic Acoustic Echo Cancellation: Analysis of the Misalignment in the Frequency Domain," IEEE Signal process.Lett., vol. 13,no. 1, pp 33-36. Jan.2006.
4. T. S.Wada and B.-H. Juang, "Multi-channel acoustic echo cancellation based on residual echo enhancement with effective channel decorrelation via resampling ,"inProc.IWAENC,2010.
5. D. R.Morgan, J. L. Hall, and J. Benesty, "Investigation of several types of nonlinearities for use in stereo acoustic echo cancellation, IEEE Trans. Speech Audio Process ,"Proc.IEEETranc.speech audio process., vol. 9, no. 6, pp. 686696, Sep. 2001.
6. J.Wung,T.S.Wada,and B.H.Juang,"Inter-Channel Decorrelation By SubBand Resampling In Frequency Domain," inProc.IEEE ICASSP , 2012, pp.29-32.
7. J.Wung,T.S.Wada, and B.H.Juang,"Inter channel decorrelation by sub band resampling for multi channel acoustic echo cancellation," IEEE Tranc. on signal processing ,vol. 62,no. 8,April 15,2014.
8. A.M.Geethu,K.S.Smitha, and D.Chengzhi,"A Fuzzy Logic Based Acoustic Echo Cancellation System,"International Journal of Engineering and Advanced Technology ,vol. 4, no. 6,August 2015, ISSN.2249-8958.
9. M.M.Sondhi, D.R. Morgan,andJ.L.Hall,"Stereo phonic acoustic echo cancellation -An overview of the fundamental problem, "IEEE Signal processing. Lett,vol. 2,no. 8,pp.148-151,Aug.1995.

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