Multi-Channel Acoustic Echo Cancellation Using Harmony Search Algorithm

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Abstract — Non uniqueness and misalignment problems that associated with Multi channel acoustic echo cancellation are due to the correlation between the sound signals. Our aim is to reduce these problems by improving the sound quality. So many methods are proposed for the maximum result. But we cannot achieve the best result. Here we use Harmony search algorithm to get the optimum result. Through harmony search algorithm we get optimum result by reducing the error rate. The amount of decorrelation in each sub-band, measured in terms of the coherence, can be controlled arbitrarily by varying the parameters. The HS algorithm improves, 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. Simulation results for the proposed algorithm have shown a significant improvement in convergence rate compared with existing techniques.

Index Terms — Multi channel acoustic echo cancellation, Coherence, Principal component method, Harmony search algorithm, ERLE, Frequency domain adaptive filter.

1 INTRODUCTION

Consider a telephonic conversation; the far end caller speech is broadcasted to the near end loudspeaker. This speech is repeated itself by bouncing off wall etc. These repetitions of signal are called an echo. These echoes are picked up by the near end caller so that the far end caller can hear his own sound. Echo cancellation is used in telephone to improve the sound quality by reducing the echo, which is created at the time of transmission. This method of echo removal is called acoustic echo cancellation (AEC). AEC not only remove the acoustic echo but also the line echo. Figure 1 shows acoustic echo cancellation system.

So the problem associated with MAEC is also different from conventional ACE. Non uniqueness problem and misalignment problem are two problems that mainly affect MAEC system. These two problems are mainly due to the correlation between the two signals. A decorrelation process before the near end playback can be reducing these problems by improving the signal quality.

Many approaches are used to solve the problem associated with MAEC system. First introduce some distortion to the incoming far end signal to decorrelate the channels. But one cannot distort too much as it will cause a degradation in sound quality. Introduce some nonlinear transformation such as half wave rectifier is an effective method of decorrelating the signals without loss of sound quality [3]. After the decorrelator, we need to find the correct echo path. Therefore the two channel RLS algorithm has chosen for adaptive filter. The two main disadvantages with RLS algorithm are calculation complexity and stability with non-stationary signals such as speech. FRLS algorithm has been show fast convergence for conditional stereo phonic echo cancellation problem. The high calculation complexity of FRLS adaptive filter is reduced by applying the filter in a sub band structure. Here the signals are decomposed with a filer bank into several sub band signals each one approximately corresponding to a given frequency region. The one adaptive filter is applied to each sub band.

The first echo canceller was developed by M. Kallinger J. Bitzer and K. D. Kammeyer. NLMS algorithm has been most commonly used adaptive algorithm [1]. Performance of NLMS is sufficient in most monophonic echo cancellers. It has too slow convergence rate. The fast recursive least square algorithm has very fast convergence rate but disadvantage of instability. The two channel frequency domain algorithm is a block based algorithm i.e., blocks of the input signals are

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Multi-channel acoustic echo cancellation (MAEC) is entirely different from that of monophonic acoustic echo cancellation.
transformed by a discrete Fourier transform from the time domain to frequency domain. The adaptive filtering is then performed in the frequency domain. We can have individual adaptive step size parameter in each frequency bin so that the convergence rate for frequency regions with low signal energy in comparison with standard NLMS algorithm. The convergence rate can also increase by overlapping data.

The paper is organized as follows. Section II gives problem formulation. Section III introduces the methodology. Section IV gives experimental and the last section give conclusion.

2 Problem Identification

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 [5]

$$y[n] = Z_n[n] + v[n] + \sum_{p=1}^{P} d_p[n]$$

(1)

where y[n] is the microphone signal, Zn[n] is the near end speech, v[n] is the near end noise. P is the number of loudspeakers. dp[n] = hT p; Kxp; K[n] is the acoustic echo generated by the pth loudspeaker.

P adaptive filter are required to cancel the P echo signal, and the error signal is given by [5],

$$e[n] = y[n] - \sum_{p=1}^{P} w^T_p L[n]$$

(2)

Where w; L[n] = [wp; 0[n]; :: : wp; L[n]]; is the adaptive filter coefficient vector for pth echo path. Here the actual length of near end room impulse response (K) is the different from that of adaptive filter coefficient vector (L).Figure 2 shows the multi-channel acoustic echo cancellation system. The minimum mean square error can be computed by solving the normal equation in [7],

$$R_{XPL}[n]w_{PL}[n] = r_{yXPL}[n]$$

(3)

3 Methodology

The Multi-channel AEC is process as following:

- A far-end audio (FE) signal is reached to the system.
- The FE audio signal is reproduced.
- The FE audio signal is filtered and delayed to the near-end signal.
3.1 Harmony Search Algorithm

Mainly Harmony Search consists of three steps. These are:

- Initialize the problem and algorithm parameters
- Initialize the harmony memory
- Improvise a new harmony from the HM set
- Updating HM
- Checking stopping criterion

A decorrelation process is needed to decrease the coherence between the two signals. Resampling approach is used to address the non-uniqueness problem. The idea decorrelation by by resampling is used to exploit the effect of sampling rate mismatch found out in [7]. A slight sampling rate mismatch between the audio channels is enough to decrease the correlation for a sufficient MAEC system, i.e., Decorrelation can be decreased by introducing a sampling rate mismatch between the correlated reference signals.

4 Results

Stereo phonic audio is the input, so left and right channel signals. The right signal is added with the echo created by using a chebyshev second order filter. The total signal coming microphone are left channel signal, right channel signal and the noise present in the room. Our aim is to reduce these echoes [4] by Frequency Domain Adaptive Filter (FDAF). The problems associated with MAEC are non-uniqueness and misalignment problem. These are due to the correlation between the two signals [6]. First we want to apply a decorrelation process to decrease the correlation. Principal Component (PC) method is used to decorrelate the signals. Then apply adaptive filter. Then Echo Return Loss Enhancement (ERLE) is plotted. Later apply Harmony search algorithm to get optimum result. Harmony search dimension taken is 10. The parameters in HS algorithm are, Harmony Memory Considering Rate is 0.95, Pitch Adjustment Rate is 0.6 and the number of iterations is 20. Through the HS algorithm we get the best parameter that can be given to the adaptive filter so it will get optimum acoustic echo cancellation.

Fig. 3 Block diagram of proposed method.

Fig. 4 MAEC output before HS algorithm
5 CONCLUSION

In this paper we introduce a new approach to the multi-channel acoustic echo cancellation system which is the Harmony Search algorithm. Non uniqueness and misalignment problems that associated with Multi channel acoustic echo cancellation are due to the correlation between the sound signals. Our aim is to reduce these problems by improving the sound quality. The HS 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.

REFERENCES


