International Journal of Engineering and Advanced Technology (IJEAT)
ISSN: 2249-8958 (Online), Volume-4 Issue-6, August 2015

A Fuzzy Logic Based Acoustic Echo Cancellation System

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Abstract*—Although a handful of inter channel decorrelation procedures have been proposed in the past to mitigate the non-uniqueness and lower the misalignment of adaptive filter, introduced audible distortion limits the performance of adaptive filtering algorithm. In this paper fuzzy based adaptive resampling algorithm has been proposed. The power of fuzzy adaptive resampling is that the amount of de-correlation can be finely controlled by applying fuzzy logic in adaptive filtering algorithm. The proposed procedure expands on the idea of fuzzy based adaptive resampling in the frequency domain that efficiently mitigates the non-uniqueness problem for a multichannel acoustic echo cancellation (AEC) system while introducing minimal distortion to the signal quality. The performance of the system can be evaluated by the true echo return loss enhancement and signal to noise ratio (SNR) per sub-band, to better demonstrate the superiority of proposed procedure over other methods.

***Index Terms*— fuzzy adaptive resampling, multi-channel acoustic echo cancellation (AEC), non-uniqueness problem, resampling**

I. INTRODUCTION

In a multi-channel acoustic echo cancellation system, the non-uniqueness problem occurs due to the correlation between reference signals, the filter coefficient does not converge to the true echo path impulse responses. It could affect convergence performance of the adaptive filtering algorithm. The multi-channel AEC solution is dependent on the far-end room impulse response, must re-converge when the far-end speech activities change. A number of inter-channel decorrelation procedures [1]-[5] have been proposed in the past to alleviate the non-uniqueness problem. However, these decorrelation techniques may not achieve an optimal steady-state performance of an adaptive filter and are usually performed in a full-band manner, leaving no possibility for frequency-selective decorrelation. By frequency-selective decorrelation, varying degrees of decorrelation can be applied for different frequency channels. Although the phase modulation procedure [6] allows a frequency-selective choice of phase modulation parameters by employing sub-band decomposition, the relationship between phase modulation and decorrelation remains unclear.

The frequency domain resampling introduces time-varying delay across channels [7]. For discrete time signals, decorrelation by time expansion/compression can be achieved by resampling a signal to a higher or lower sampling rate. In delay smoothing for different combinations of the resampling ratio, (expansion or compression) and for the resampling direction (forward or backward) give rise to four possibilities: forward expansion, forward compression, backward expansion, and backward compression [5]. The variable delay between the two channels can be achieved by properly resampling the reference signals, and the delay can be varied smoothly across time to eliminate the distortion associated with sudden changes in the delay.

A decorrelation algorithm based on a resampling procedure [1], which was motivated by the analysis of the sampling rate mismatch problem inherent to audio processing [2]. The resampling procedure was extended to the frequency domain [3], and the sub-band resampling (SBR) [4]. When applied to the noise robust frequency domain multi-channel acoustic echo cancellation system [5], decorrelation by resampling achieves a faster echo path tracking than other decorrelation procedures [1], [3]. In sub band resampling the amount of decorrelation can be finely controlled ,that is less resampling or signal modification is required at lower frequencies and vice versa at higher frequencies , the amount of resampling can be arbitrarily controlled in each frequency bin such that the perceptually less significant sub-bands can be more aggressively decorrelated [4], [5].

The objective of this work is to demonstrate the effectiveness of fuzzy adaptive resampling on multi-channel acoustic echo cancellation while still preserving a high speech quality. In the experimental evaluation it shows that the proposed scheme out performs the other decorrelation method in terms of echo return loss enhancement and signal to noise ratio. This paper is organized as follows: section 2 briefly describes the typical adaptive filter and the normalized least mean square algorithm. The system aspect of decorrelation is described in section 3. Section 4 describes about adaptive resampling based on the proposed fuzzy logic. Finally section 5 summarizes main contribution of this paper.

II. FREQUENCY DOMAIN ADAPTIVE FILTERING

A. Typical Adaptive Filter

An adaptive filter is a computational device that iteratively models the relationship between the input and output signals of a filter. According to the adaptive filtering algorithm, an adaptive filter self-adjusts the filter coefficient. Fig.1, shows a typical structure of an adaptive filter, where x(n) is the input
signal to the linear filter, y (n) is the output signal, d(n) is the Desired signal, and e(n) is the error signal.

Manuscript published on 30 August 2015.
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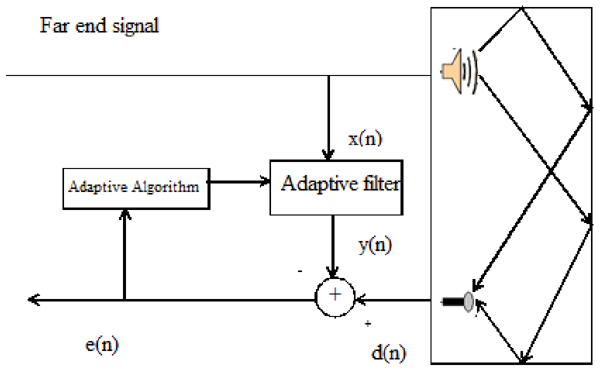

Fig.1. Adaptive Filter

One of the important applications of adaptive filter includes acoustic echo cancellation. The adaptive filter compute the difference between desired signal, d(n) and the adaptive filter output, y(n). The error signal, e(n) thus obtained is fed back to the adaptive filter. The coefficients of the filter are self-adjusted algorithmically in order to minimize e(n). In the case of acoustic echo cancellation, adaptive filter estimates the echo contribution in the microphone signal.

B. Normalized Least Mean Square Algorithm

Normalized Least Mean Square (NLMS) algorithm is an adaptive algorithm, widely used for echo cancellation. NLMS algorithm can be implemented using the following steps [8]:
The output of the adaptive filter can be calculated as

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = W^T(n)X(n) \qquad (1)$$

where $X(n)=[x(n), x(n-1), \ldots x(n-N+1)]^T$ is the input vector and $W(n)=[w_0(n), w_1(n), \ldots w_{N-1}(n)]$ is the weight vector.
Error signal is calculated as

$$e(n)=d(n)-y(n) \qquad (2)$$

The filter tap weights are updated for next iteration

$$w(n+1)=w(n)+\mu(n)\frac{e(n)x(n)}{x^T(n)x(n)} \qquad (3)$$

where $\mu(n)$ is the step size value for the input vector.

III. SYSTEM ASPECT OF DECORRELATION

The NLMS Algorithm can iteratively solves the normal equation

$$R_{xx}W=r_{XY} \qquad (4)$$

where $R_{xx}=E\{XX^T\}$ is the auto correlation matrix of X ,{.} is the expectation operator, X is the input vector, Y is the output vector, W is the weight vector and r_{XY} is the cross correlation vector between X and Y. A small mismatch in the sampling rate between X and Y, merely 0.01% is enough to break down the correlation structure of r_{XY} for significant convergence in the NLMS algorithm. Conversely, the conditioning of r_{XY} can be improved by resampling X frame-wise [9].

A. Frequency Domain Resampling

For continuous-time signals, the de-correlation can be achieved by the time-scaling property of the continuous-time Fourier transform (CTFT), and is given by

$$CTFT\left\{x\left(\frac{t}{R}\right)\right\} = RX(RW) \qquad (5)$$

Frequency Domain Resampling (FDR) is implemented by resampling a signal to a higher/lower sampling rate f_s^-.The goal of FDR is to interpolate across frequency rather than across time, with appropriate expansion or compression of the spectrum, to reduce the computation via the fast Fourier transform (FFT). The expansion/ compression ratio is related to the resampling ratio as $R = f_s / f_s^-$. The resampling ratio R may be adjusted separately over each sub-band in the frequency domain [5].

IV. FUZZY ADAPTIVE RESAMPLING

Fuzzy logic is an effective approach to handle non-linear signals such as stereo 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.

A. Fuzzy Logic Control Systems

The typical Fuzzy Logic Control (FLC) system is shown in Fig.2. It consists of four components fuzzifier, fuzzy rule, base inference engine and defuzzzifier [9].

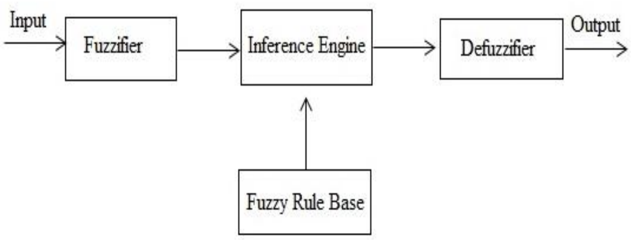

Fig.2. Fuzzy Logic Control System

The fuzzifier transforms crisp measured data into suitable linguistic values. The fuzzy rule base stores knowledge of operation of the process of human experts. The inference engine has capability of stimulating human decision making. The defuzzifier is used to obtain non fuzzy decision.

B. System Model

From equation (3) it is clear that step size, $\mu(n)$ of the adaptive filter controls how well and how fast the adaptive filtering algorithm converges to optimum filter tap weights. Therefore the step size value should be chosen carefully. If the value of μ (n) 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. One the other hand, if step size, $\mu(n)$ is chosen to be too small, time required to converge to the optimal coefficient values will be too large. Therefore it is necessary to have an upper bound for $\mu(n)$, and which is given by

$$0 \langle \mu \langle \frac{1}{\lambda_{\max}} \qquad (6)$$

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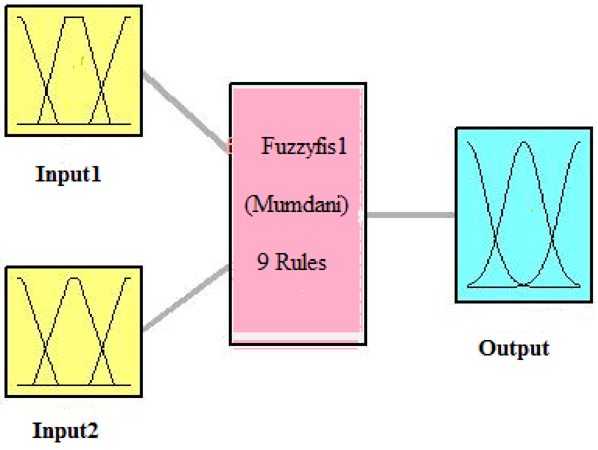
where λ is the convergence factor for weight updates and λ_{max} is the largest eigenvalue of the auto correlation matrix, $R_{XX}=E\{XX^T\}$. If this condition is not satisfied, the algorithm becomes unstable and w(n) ,the filter co efficient weight diverges.



System Fuzzyfis1: 2 inputs,1 output, 9rules

Fig.3.Fuzzy Architecture

Fig.3 shows the overall architecture of fuzzy adaptive resampling. Step size (μ) of the frequency domain adaptive filter can be obtained by performing fuzzy based resampling in each sub-band. The proposed fuzzy system two input variables namely, FFT input signal power (δ) and convergence factor for weight updates (λ). In fuzzy based systems the input- output relationship is described by a collection of rules, IF-THEN statements, which involves linguistic variables. In the proposed system consist of nine rules which can give appropriate values for resampling the input signal.

Table.1.Fuzzy Rules

IF "input signal power is low" and "convergence factor for weight updates is low" THEN "resample value is high"-Rule no. 1.
IF "input signal power is low" and "convergence factor for weight updates is medium" THEN "resample value is medium"-Rule no. 2
…………..
…………..
…………..
IF "input signal power is high" and "convergence factor for weight updates is high" THEN "resample value is low"-Rule no. 9.

The membership functions for input1and input2 are shown in Fig.4. and Fig.5.respectively. Similarly, the membership function for output is shown in Fig.6.

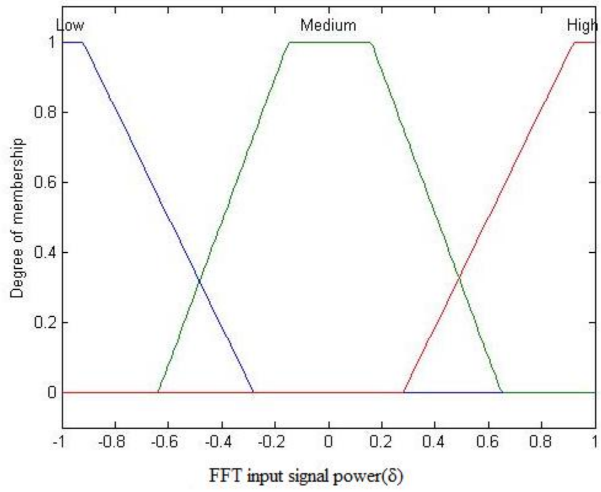


Fig.4. Fuzzy membership functions for input 1

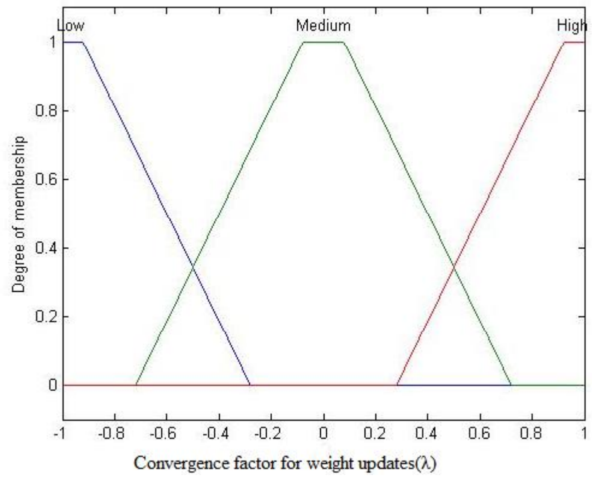


Fig.5. Fuzzy membership functions for input 2

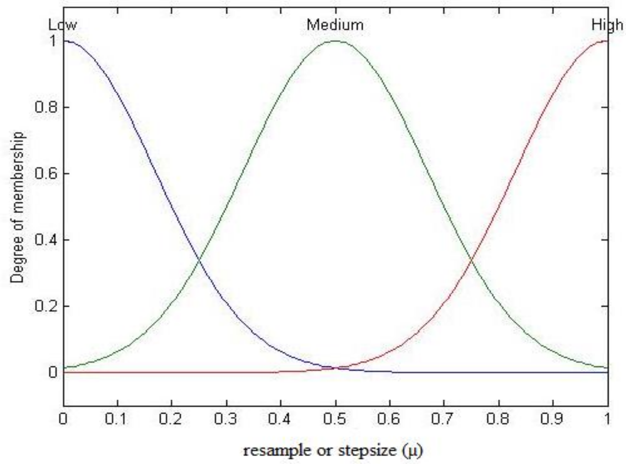


Fig.6. Fuzzy membership functions for output

V. SIMULATION AND RESULTS

In order to evaluate the performance fuzzy adaptive resampling based multi-channel acoustic echo cancellation system, it is necessary to find the echo return loss enhancement of the system. Echo is typically measured in terms of Echo Return Loss (ERL).

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ERL is the ratio between
the original signal and the echo level expressed in decibels (dB). Echo Return Loss Enhancement is a measure of the loss of the signal that comes back as echo. A higher ratio corresponds to a smaller amount of echo. Comparison of echo return loss enhancement of proposed method with conventional method is shown in Fig.8. From figure it can be seen that echo return loss enhancement of proposed method is high when compared with conventional method. Comparison of the signal to noise ratio (SNR) of the proposed method with conventional method is shown in Fig.9.

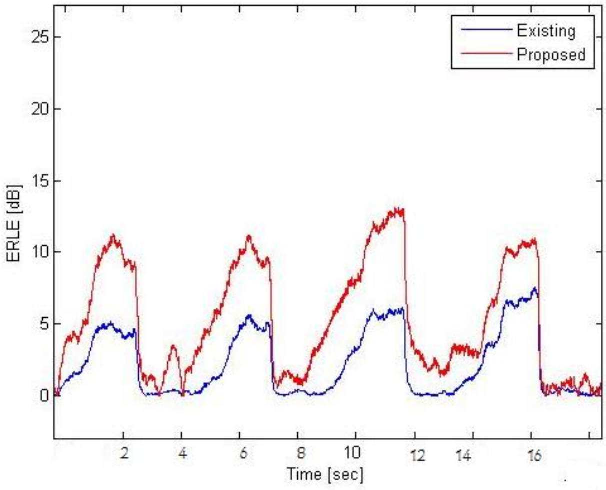

Fig.8.comparison of echo return loss enhancement

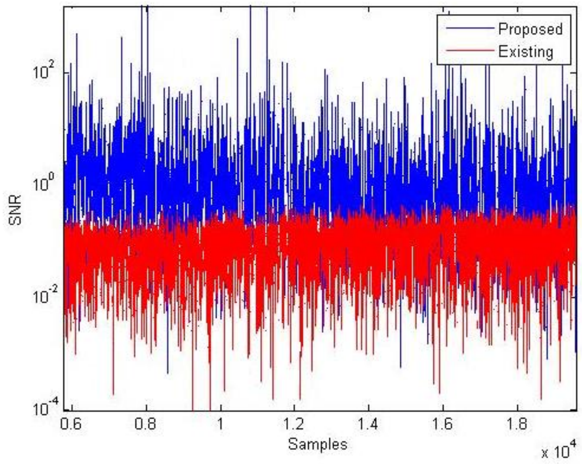

Fig.9.comparison of signal to noise ratio (SNR)

VI. CONCLUSION

Fuzzy logic is an effective approach to handle non-linear signals such as stereo signals. Tap wieghts obtained by applying fuzzy adaptive resampling is more unique and have faster convergence. The proposed procedure is based on the idea of fuzzy based adaptive resampling in the frequency domain, to directly assist the frequency domain adaptive filtering algorithm. In the experimental evaluation it has been shown that the proposed method out performs other conventional methods in terms of echo return loss enhancement, signal to noise ratio.

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