Echo Cancellation System in VOIP using MATLAB

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ABSTRACT

Acoustic echo cancellation (AEC) method reduces undesired acoustic echo arriving at microphone and also emphasizes the target talker's voice. It cancels the non linear acoustic echo which appears due to distortion and ultimate pure delay caused due to both room echo and audio input and output buffers. This AEC method reduces more than 40 dB of the undesired echo.

Keywords *– Acoustic Echo Cancellation, Conventional AEC, Adaptive Filter, Voice Over Internet Protocol.*

I. INTRODUCTION

Now a day's voice over internet protocol (VOIP) phone applications which run over smart phones and tablets have very popular AEC helps to prevent detrimental acoustic echo and howling which is generated due to coupling between loudspeakers and microphones. There are several AEC techniques of which consist of an adaptive filter (ADF). It identifies the acoustic echo path and cancels out the acoustic echo but still some residual echo remains back due to which echo reduction is used which reduces the residual echo in the error signal.

II. PROPOSED SYSTEM

2.1 ADF and ER:

The ADF and ER when combined together improve the performance. But there are three limitations of AEC loudspeaker distortion, microphone sensitivity variation and audio input output delay variation. Hence a new AEC method is proposed which uses three new techniques that is loudspeaker distortion cancels out both linear as well as non linear echoes, microphone sensitivity variation tracks the residual echo level instantaneously while audio input output delay variation can calculate pure delay between room echo and buffer.

2.2 Conventional AEC:

The conventional AEC method is shown in figure 1.

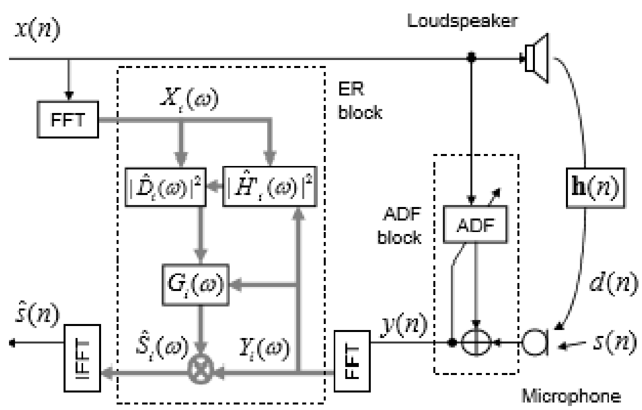

Fig.1 Conventional AEC Method.

The echo reduction gain can be calculated using the following Wilmer filtering method,

$$G_i(\omega) = \frac{|Y_i(\omega)|^2 - |\hat{D}_i(\omega)|^2}{|Y_i(\omega)|^2}$$

Where,

$|\hat{D}_i(\omega)|^2$ is the estimate of the residual echo level and is given by,

$$|\hat{D}_i(\omega)|^2 = |\hat{H}_i'(\omega)|^2 |X_i(\omega)|^2$$

Where,

$$|\hat{H}_i'(\omega)|^2 = \left(\frac{E[|X_i(\omega)||Y_i(\omega)|]}{E[|X_i(\omega)|^2]}\right)^2$$

Where, E[.] is the reasmble average.

Proposed AEC:

The proposed AEC Method, reduces the issues of distortion variations in level and delay with ADF, ER and DE techniques. ADF eliminates linear and non linear echoes, ER takes the residual echo levels and DE calculates pure delay caused from echo and buffer produces audio input output.

Non linear ADF compensates non linear echo caused by the saturation effects because of small loudspeaker and poor amplifier. The output signal X(n) is given by,

$$u(n)=\begin{cases} a(n) & (x(n)>a(n)) \\ x(n) & (|x(n)|\le a(n)), \\ -a(n) & (x(n)<-a(n)) \end{cases}$$

Where, a(n) denotes non negative hard clipping threshold.

This u(n) is transformed into frequency domain signal u(ω) and frequency domain estimate of echo signal can be calculated as:

$$V(\omega)=W(\omega)U(\omega)$$

Instantaneous ER eliminates the residual echo level by separating level and spectral structure from the residual echo. The residual echo level changes when the microphone sensitivity varies even if the echo path changes. As a result, the power spectrum of residual echo can be calculated by,

$$|\hat{D}_i'(\omega)|^2=\hat{g}_i|\hat{H}_i'(\omega)|^2|X_i(\omega)|^2$$

$$\hat{g}_i=\max\left[\frac{\sum_{m=0}^{Q-1}\sum_{\omega=0}^{N-1}|Y_{i-m}(\omega)|^2|\hat{H}_{i-m}'(\omega)|^2|X_{i-m}(\omega)|^2}{\sum_{m=0}^{Q-1}\sum_{\omega=0}^{N-1}\left(|\hat{H}_{i-m}'(\omega)|^2|X_{i-m}(\omega)|^2\right)^2},1\right]$$

Where,

$|\hat{H}_i'(\omega)|^2$ is the estimated power frequency response of residual echo path. Delay estimation technique calculates the echo path transfer segment as:

$$\begin{bmatrix} H_0^{"}(\omega) \\ \vdots \\ H_{L-1}^{"}(\omega) \end{bmatrix} \approx \frac{\begin{bmatrix} X_0(\omega) & 0 & 0 \\ \vdots & \ddots & 0 \\ X_{M-1}(\omega) & \ddots & X_0(\omega) \\ 0 & \ddots & \vdots \\ 0 & 0 & X_{M-1}(\omega) \end{bmatrix}^H \begin{bmatrix} Z_0(\omega) \\ \vdots \\ Z_{M+L-2}(\omega) \end{bmatrix}}{\sum_{i=0}^{M-1} X_i^*(\omega)X_i(\omega)}$$

Where,

$Z_i(\omega)$ is frequency domain microphone signal, H is complex conjugate transposition, L and M are the number of multi delay filter.

III. RESULTS AND DISCUSSIONS

The proposed AEC method consist of sampling frequency switch (SFS), Sampling frequency converter (SFC), Analysis filter (AF), Loss control (LC), Synthesis filter (SF), Sound device control, Delay estimator, Buffer and acoustic echo controller. The signal enters decode followed by AEC software and the sampling frequency is selected by SFS. Then it is decoded into two or three sub bands whose ranges are 0-4, 4-8, 8-16 KHz. The delay between the acoustic echo and sound device is estimated by delay estimator and then the signal undergoes gain control by the LC. Synthesis filter (SF) resynthesizes the sub band signals and selecting the sampling frequency the output is obtained at the encoder.

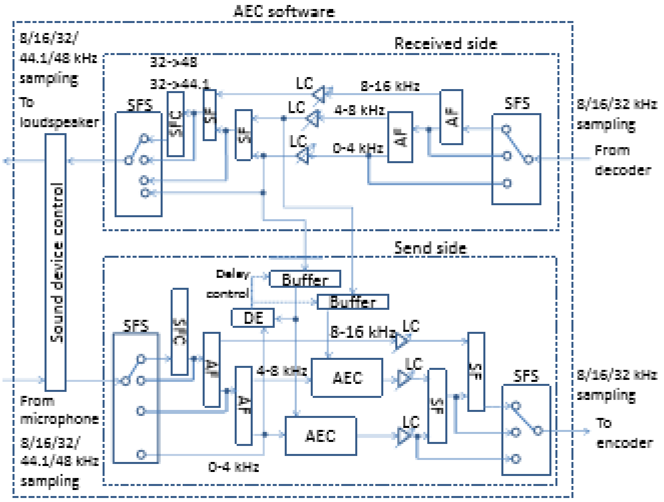

Fig.2 Block diagram of AEC software implemented in VOIP application.

IV. CONCLUSION

The proposed AEC method helps to reduce the undesired acoustic echo in smart phones and tablet model which is implemented in voice over internet protocol (VOIP) hand free phone application which is better than conventional AEC method. The proposed AEC method estimates instantaneous residual echo variation non linear acoustic echo and pure delay which results in better performance than conventional AEC method.

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