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# REVIEW OF ACOUSTIC ECHO CANCELLATION TECHNIQUES FOR VOICE OVER IP

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**ABSTRACT**

Acoustic Echo Cancellation (AEC) has become a necessity in today's conferencing system in order to enhance the audio quality of hands-free communication systems. In recent years, many researchers and manufacturers have developed various AEC algorithms for telecommunication solutions in order to improve the quality of service. Many factors influence the design of an AEC system, such as computational complexity, memory consumption etc. The aim of this work is to review the most recent acoustic echo cancellation techniques and their applicability for current hands free applications. Therefore, this paper presents AEC systems challenges and comparison between these techniques is also presented.

**Keywords:** *Adaptive Filter, Acoustic Echo Cancellation, Noise Reduction, Voice Over Internet Protocol*

## 1. INTRODUCTION

In recent years, the telecommunication industry has evolved tremendously that new services are introduced daily. The usage of new services, such as Voice over Internet Protocol (VoIP), has become so widespread that it has transforms the ways human communicate. While the use of VoIP has been improving human's life, there has been a great demand for high quality VoIP solution that is able to provide clear and intelligible conversation without compromise. Users will be annoyed if the received speech is in a noisy condition, making conversation between the users difficult. As such, methods for suppressing, eliminating or compensating echo effects when the near-end speech signal is simultaneously transmitted are needed [1].

Acoustic Echo Cancellation (AEC) system is widely used to eliminate the undesired echo signal in many hands-free and teleconferencing solutions. Generally, when the far-end signal is delivered to the speaker, the Acoustic Echo Cancellation (AEC) system will filter and delay the far-end signal that is picked up by the microphone. The filtered far-end signal is then subtracted from the near-end signal so that only the near-end speech signal is sent to the far-end. Thus, the adaptive algorithm that is used in the filter should be able to predict the characteristics of echo path, replicate the echo signal and subtract the echo from the microphone signal in order to achieve an optimal desired output. Several adaptive algorithms has been proposed and used in the past namely, Least Mean Square (LMS), Normalised Least Mean Squares (NLMS), Recursive Least Square (RLS), Affine Projection Algorithm (APA), Subband Adaptive Filtering, and Frequency Domain Adaptive Filter (FDAF). There are several criteria that determine the choice of the adaptive algorithm to be used. One of the factor is the flexibility of the algorithm to operate in a unknown and changing environment [2].

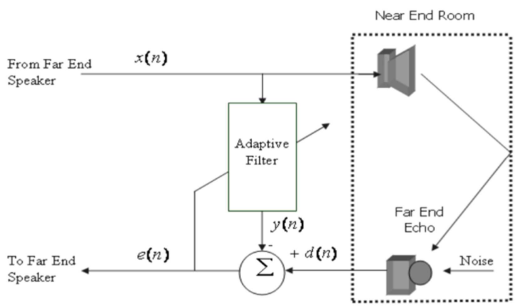


*Figure. 1. Acoustic Echo Canceller [3]*

It should also provide fast and stable convergence so that real speech signal will not be affected.

This paper is mainly focus in AEC system that involving adaptive filter based on LMS algorithms.

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In addition to AEC techniques that solve sampling rate mismatch between far-end and near-end signals. The rest of paper is divided into four sections. Section 2 discusses about the different adaptive algorithms proposed in AEC. In Section 3 and 4, strength and weaknesses of each AEC systems will be analyzed. Section 5 concludes the review and ideas for future work.

## 2. ADAPTIVE FILTERS

As shown in Figure. 1, adaptive filter will generate a replica of the echo, *y(n)* and the estimated echo is subtracted from the desired input signal *d(n)* yielding the estimated error signal,

$$e(n) = d(n) - y(n) \tag{1}$$

The estimated error signal will be piggybacked to the adaptive filter so that it can self-adjust the transfer function to achieve optimum performance [3].

Least Mean Square (LMS) algorithm, a stochastic gradient-based algorithm, is one of the most widely used algorithms in adaptive filtering. It is well known for its simplicity in computation and implementation [4]. However LMS algorithm is very sensitive to the spectral and power of input signal which makes it hard to adjust the step size and guarantee the stability of the algorithm[5, 6]. As such, normalized convergence parameter is developed to resolve this problem by normalizing the step size with power of input signal, resulting the convergence rate independent from signal power [7]. The advantage of new the algorithm, Normalized LMS (NLMS), is noticed when power of input signal is changing, making it suitable to predict echo. However it requires additional computational multiplication for normalization terms. Both LMS and NLMS have slow convergence rate when the input signal are highly correlated [8, 9].

Gradient-based LMS-algorithm (Widrow-Hoff) or a recursive least squares (RLS) are complex especially for full band implementation. The dynamic characteristic of speech including intervals of complete silence is proven to be a problem in adaptive filtering [10]. In addition the far from white spectral character slows down the adaptation speed causing long convergence time and making the system sensitive to changes of the acoustic room response. Finally the near-end speech and background noise if present also put demands on the system design.

In the other hand, Frequency Domain Adaptive Filter (FDAF), which was proposed in 1992 by Shynk [11], is designed to achieve fast convergence rate and low computational cost, where desired signal and input signal are transformed into discrete frequency domain using discrete Fourier transform (DFT). Instead of linear convolution and correlation that are performed in LMS and NLMS adaptive filter, circular operation is performed in frequency domain on a block-by-block rule instead of sample by sample in LMS and NLMS [6, 9]. According to [9] FDAF has attractive computational and convergence rate when the block size has the same amount of filter length. The major drawbacks of FDAF is has long delay due to some restore operation that is required to perform the circular operation [12]. By splitting the impulse response into equal parts to produce time and frequency convolution mixed together, leads to new version of FDAF called Partitioning Block FDAF filter (PBFDAF). In PBFDAF, the length of block can be adjusted to achieve cheap acoustic echo canceller with acceptable level of delay [9, 12].

Subband adaptive filter (SAF) [8, 13] is designed to exploit the subband properties to perform more efficient signal processing. The input signals in subband are decomposed into multiple parallel channels and synthesis to construct the fullband signal at the output. Thus, the input signal and output signal are decomposed into N spectral bands using analysis filters, and each filter has an independent adaptation feedback and it computes its error internally. The fullband error signal is constructed using synthesis filter bank. LMS and NLMS can be adopted to subband adaptive filter to minimize the Least Mean Error (LME), several types of SAF are proposed and explained in [8]. Delayless structure of closed loop was found more suitable for real time application such AEC system. However, the decomposed the input signal and synthesis the fullband error signal introduces delay which undesired in real time AEC system. A comparison study had done by [12] between SAF and FDAF concludes that in real time acoustic echo cancellation, SAF introduces unwanted delay and suffer from residual errors while FDAF does not suffer from such problems equivalent to SAF.

The sub-and realization will be able to reduce the complexity by dividing the signal into and applying adaptive filters to a decimated signal in each sub-band. In addition the spectral variability within a sub and is reduced as compared to the full band signal. To maintain transparency of the near

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end speech signal one will require the cascade of the analysis and synthesis filter banks to provide perfect reconstruction.

## 3. SAMPLING RATE MISMATCH

Besides choosing the right adaptive filter, there are other factors that may impact the performance of the AEC system. For example different sampling frequencies of D/A and A/D converters can degrade the voice quality. The deterioration of performance is due to the nonlinear time-varying disturbances of the effective echo path caused by the offset, as well as buffer overflow or underflow [14, 15]. Two kinds of sampling rate should be taken into account to improve the AEC system, first the sampling rate of play back audio in PC, second, sampling rate offset of A/D and D/A converter are not exactly the same which degrade the performance of echo cancelation system.

Stokes and Malvar [16] addressed the effects of different sampling rate between microphone and playback audio signal of CD-quality or any other played sound such as 44.1kHz in the PC which is usually higher than the captured sampling rate signal from the microphone. In order to cancel played back audio signal from captured signal by microphone, sampling rate should be converted before it is fed to AEC system. Frequency Domain SRC was proposed in [16] to correct the sampling rate of PC play back signal. Nevertheless, several SRC for audio applications was designed [17, 18], using either FIR filter or Farrow filter and their modification. In many cases such audio videoconferencing, FIR filter can perform efficiently with less penalties in terms of time and memory.

In fact, the worst case occurs when personal computer with commercial audio hardware is used for teleconferencing which could result small offset of sampling rate between far-end (microphone) signal and far end (speaker) signal. According to Robledo-Arnuncio et al. [19] sampling rates could vary among the components due to:

- Clock signal generators have a certain tolerance in their nominal frequency.
- A temperature change can affect the operating frequency of devices.
- In different devices, the clock signals used for the audio hardware are often obtained by applying different division factors to a higher-frequency clock. Therefore, the user may not find the same expected nominal frequencies for different devices.

Arbitrary Sampling Rate Conversion (ASRC) is found more suitable to correct the small change in sampling rate rather than SRC which is used for rational conversion. Variable Fractional Delay ASRC (VFD ASRC) is used to correct the sampling rate (increase or decrease) of digital signals [20]. An estimation method of sampling rate offset is propose d in [21] by extend the LMS algorithm, and then correct it through two mechanisms: frame-step control and phase rotation. While, Pawig [14] suggested using a least mean squares (LMS) adaptive algorithm to estimate the frequency offset (FOE) and match the signals using arbitrary sampling rate conversion (ASRC). Robledo et al. [22] states that sampling rate correction can be achieved efficiently by employing a simple interpolation procedure in time domain instead of the conventional approach of up-sampling followed by down-sampling.

Blind sampling rate offset estimation [23] is designed for compensation in beam forming applications. The proposed method utilizes speech-absent time segments, where the interference statistics is assumed slowly time-varying, and the sampling rate offsets are assumed fixed. An estimation procedure for the sampling rate offsets is proposed based on the coherence between the received signals. Miyabe [24] proposed Short-time Fourier Transform (STFT) domain to compensate the sampling rate offset, by applying the linear phase shift. The effects of mismatch in acoustic echo are discussed in next section.

## 4. COMPARISON AND CHALLENGES IN ACOUSTIC ECHO CANCELLATION SOLUTION

Basically, Acoustic echo cancellation (AEC) is defined as a scheme to remove the echoed signal that is applied on hands free communication systems full-duplex. In most AEC systems, adaptive filter is used. Adaptive filter algorithms are widely applied in acoustic echo canceller (AEC) such as namely Recursive Least Square (RLS) [25] filter and Least Mean Square (LMS) filter. The state-space Kalman Filter is a recursive least square error method for estimation of a signal distorted in transmission through a channel and observed in noise. Unlike Kalman Filter that is used to model the dynamics of the signal process, RLS is a recursive implementation of Wiener filter that is used for stationary processes [26]. A relatively simpler algorithm, LMS, uses the gradient search

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method to search for the least square error filter coefficients.
Figure. 1 illustrates the operation of AEC system, where far-end signal *x* is played out of the speaker. The acoustic echo (*d*) is captured by the microphone, along with the near-end signal (*s*) and the noise signal (*n*), the microphone signal is indicated in Figure. 3 by *y* [27].

Adaptive filter is used the far-end signal *x* to estimate the acoustic echo signal *(d)* that should be removed from the near-end signal (*s*). The estimated echo signal by filter is subtracted from the microphone signal and the result (e) which no longer contains the speaker signal (acoustic echo). However, two main challenges have been addressed in designing an AEC system for PCs; first challenge is discussed in pervious section sampling rate mismatch of signals at adaptive filter inputs [14, 21, 28]. Second challenge is caused by re-sampling processing delay between far-end and near-end signals which may degrade the performance of AEC system [22, 29].

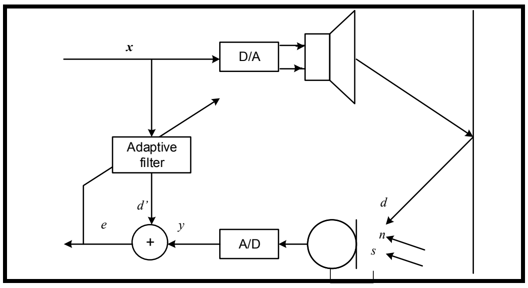


*Figure. 2. Basic Operation of Acoustic Echo Cancellation*

In this section, the related works is discussed according in terms of sampling rate mismatch and misalignment of signals at filter inputs. Pawig [14] studied the effects of different sampling rates of the D/A converter and the A/D converter of low-quality PC audio hardware may cause increasing or decreasing delay, which results in lost or repeated samples, which in turn affects the adaptive filter algorithm for the AEC system and deteriorates echo estimation.

Figure.2. illustrates the effect of sampling rate offset on echo path. In the experiment, Pawig used two kinds of offsets (Δf = 0 Hz, Δf = 6 Hz) and compared between each case by fixing the other coefficients and setting the echo path length (M = 300) and using the NLMS algorithm with step size of α (k) = 0.5 and filter length of N = 300. The 6 Hz offset changed and shifted the peak (Figure.2 d, e, f) to the right, thus affecting estimation of the echo path. On the other hand, when the offset was 0 Hz, the peak was stable in (Figure.2 a, b, c) and the adaptive filter will be able to remove the acoustic echo signal.

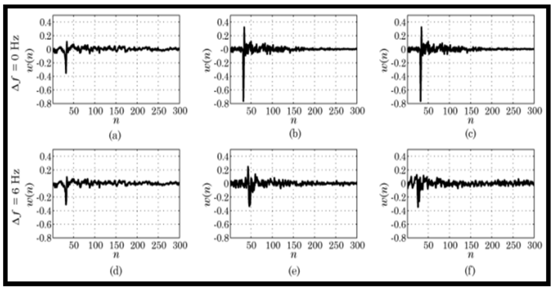


*Figure. 3. Effect of sampling rate offset in echo path [14]*

Pawig proposed a framework to tackle of this problem as illustrated in Figure.3. The offset re-sampling is estimated by comparing the near-end signal, yc(kTy), and the far-end filtered version, dc^(tk). The author proposed ASRC to correct the frequency offset. However, the sampling rate correction is applied for far-end signal which add more complexity to the system. Thus, every VOIP connection, Pawig framework has new sampling rate offset Δf to correct. Moreover, sampling rate estimation achieve high delay to update with the ASRC with far-end sampling rate.
Frequency Domain Acoustic Echo Canceller (FDAEC) that addresses the problem of sampling rate offset is proposed by Abe [21], utilizing the concepts proposed in [24] of using STFT to estimate the sampling rate offset. Thus, sampling rate correction is achieved through two schemes, frame-step control and phase rotation. The estimation and correction are carried out in a single feedback loop without an external re-sampling filter as it is shown in Figure.4. The designed framework in [21] increases the complexity of system which make it hard to adopt in real time application which require low complexity and efficient delay less.

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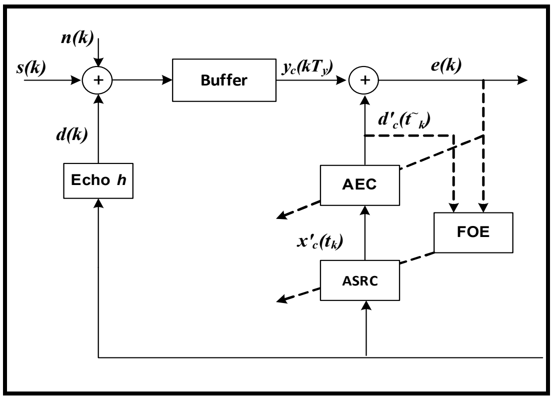


Figure. 4. *Solving The Sampling Rate Problem With AEC*

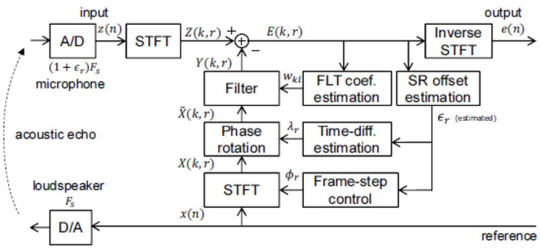


Figure. 5. *Block Daigram of Frequency Domain Acoustic Echo Canceller*

Delayless adaptive filter called partitioned block frequency-domain adaptive filter (PBFDAF) proposed in [30] to tackle the issues of delay and complexity of AEC. Delayless PBFDAF eliminates the input-output delay and have a uniform distribution of the computations. The evaluation of proposed model had been carried only to present the complexity, delay and tracking ability of proposed methods. More experiment should be taken to enhance the ability of proposed methods to cancel the acoustic echo.

Ding [31] proposed a drift-compensated adaptive filtering (DCAF) scheme (Figure. 5). They divided the proposed scheme into three parts. The first part consists of timing drift estimation and compensation. The timing drift is dynamically estimated by evaluating time averages and compensated for by re-sampling the signal d(n) at the same sampling rate as the signal x(n). The re-sampling is conducted by up-sampling the signal d(n) to factor I and then decimating it by a time-varying factor D(n) ≈ I to get the wanted sampling rate with a sampling frequency approximately equal to the x(n) sampling rate. The second part is the Ratchet FAP (Ratchet Fast Affine Projection). Ding chose the Ratchet FAP as the adaptive filter algorithm for the AEC system because it is better than other FAP algorithms in terms of performance and stability. The third part in the proposed scheme is the peak position adjustment, which works to monitor the position of the main part of the coefficients and adjusts the signal if needed. Changes requires in some part of Ding proposed structure:

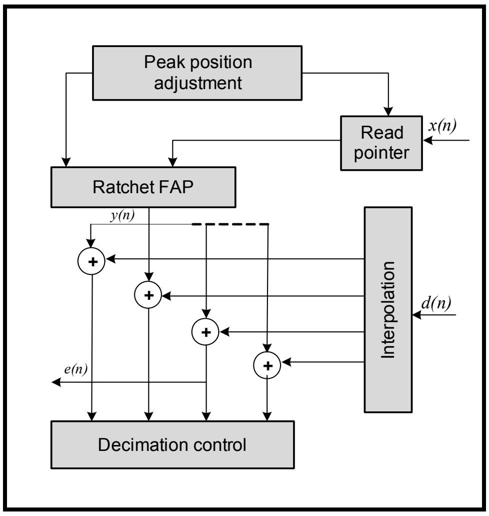


Figure. 6. *Drift-compensated adaptive filtering*

The read pointer for x(n); Coefficients of the adaptive filter, which are shifted one sample to the left or to the right (depending on the need) with a zero appended to the opposite end; The autocorrelation matrix estimate of the Ratchet FAP adaptive filter. Sums are also shifted and appended accordingly.

However, the decimation and interpolation for re-sampling the near-end signal make using the DCAF scheme problematic. It does not work with arbitrary sampling rate signals.

Table1 illustrates the differences among the methods that have been proposed to solve the sampling rate mismatch problem in AEC. Each method uses different techniques to match the input signals to the AEC system, and each method has advantages and disadvantages.

## 5. OPEN RESEARCH ISSUES

Nowadays, the AEC systems gaining more attention due to increasing of telecommunication applications that enable hands free speaker. In the other hand, AEC still suffer from different issues that downgrade its performance in such

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applications. These issues are recommended to be considered from researcher in order to well remove the acoustic echoes during the conversations.

First, FDAF filter is proved to be suitable due to fast use the block and Fast Fourier Transform (FFT) techniques which enhances the fast response of filter. However, FDAF still suffer from long conversation in real-time applications which lead the adaptive filter stop working. FDAF filter should have some modifications to adapt with such scenarios. future research in this field remains necessary. The following topics are the suggested directions for further research:

Second, mismatch of sampling rates between input signals at adaptive filter affect the performance of echo canceller. So sampling rate correction applying but the main problem is the proposed estimators of offset frequency it takes long time to predict the frequency offset.

Finally, to enhance the quality of voice communication, the effectiveness of double-talk-detection (DTD) system should enhance and integrated with AEC system.

## 6. CONCLUSION

Acoustic echo cancelation system is affected by two type of sampling rate mismatch, first from play back audio such CD quality sampling rate which can be solved efficiently using sampling rate conversion (SRC). Second sampling rate mismatch is presented due to lack of source clock in the audio devices. This problem leads to unavoidable sampling rate mismatch at the input of adaptive filter. This paper review several researches to correct the sampling rate, most works estimate the sampling rate mismatch then correct the sampling rate. However, some research shown delay and computational complexity that may affect the quality of VOIP conversion. Such applications require less delay and high efficiency to ensure the level of audio quality. Besides, such process to correct the sampling rate could lead to misalignment at adaptive filter inputs.

*Table 1: Summary Of Recent Research In Acoustic Echo Cancelation*

| Year | Researcher | Problems | Suggested solution | Issues |
|---|---|---|---|---|
| **2004** | Stokes [16] | Playing audio during voice chat affects AEC | Interpolated frequency domain SRC | Correct the sampling rate of play back only and does not consider the sampling rate offset from microphone. |
| **2007** | Enrique | Sampling rate mismatches affect AEC | Using interpolation to simulate small sampling rate mismatches and to analyze their effects on AEC | Quantify the requirement of a rate mismatch correction for AEC |
| **2008** | Qin Li [28] | Clock drifting, time-varying delay, and glitch recovery | RSO (Relative Sample Offset ) | Introduces the problem of time drifting at the input of adaptive filter. |
| **2010** | Pawig [14] | Sampling frequency offset between loudspeaker and microphone in commercial audio hardware | Using LMS adaptive algorithm to estimate the frequency offset and resynchronize signals using ASRC | Delay of frequency offset should be minimized and using FDAF can enhance the performance of AEC |
| **2011** | Ding [31] | Timing drifts between two inputs into the system | DCAF (drift-compensated adaptive filtering ) | To improve speech in the presence of asynchronous interference with corrupted target speech |
| **2013** | Y. Feiran [30] | Complexity and Delay in AEC | Modified version of PBFDAF filter | Does not consider the the problem of sampling rate mismatch if filter inputs. |
| **2014** | Abe [21] | Sampling frequency offset between loudspeaker and microphone | Applying STFT to estimate the sampling rate offset | Increasing delay and complexity |

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