Polyphase Allpass IIR Filters for Subband Acoustic Echo Cancellation

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Abstract
Echo is the delayed and possibly distorted version of the transmitted signal/sound reflected back to the source. It degrades the received speech quality and is uncomfortable and annoying. To remove the echo, an adaptive filter which adjusts its transfer function according to an optimization algorithm based on error signal, is used. Here we propose Acoustic Echo Cancellation (AEC) using subband adaptive filter. Subband adaptive filtering scheme which is based on IIR filterbanks using allpass polyphase filters is used and ERLE and MSE are computed.

Keywords
Adaptive Filtering, echo, Acoustic Echo cancellation, ERLE, SAF

I. Introduction
The main aim in hands-free telephony and in teleconference systems is to provide a good free voice quality when two or more people communicate from different places. The problem is the creation of acoustic echo. Acoustic echo occurs when an audio signal is reverberated in a real environment, resulting in the original intended signal plus attenuated, time delayed images of the signal. It is also known as “multipath echo” and is produced by poor voice coupling between the ear piece and microphone in handsets, conference room and hands-free devices [1].

The acoustic echo deteriorates the speech quality, and then the acoustic echo canceller is used as one of the technologies that remove the echo [9]. Acoustic Echo Cancellation (AEC) is a critical problem in teleconferencing systems and hands-free mobile terminals [2] [6]. Acoustic Echo Canceller [5] is used in teleconferencing and its purpose is to provide high quality full-duplex communication. Adaptive filters are widely used for acoustic echo cancellation. The adaptive filters are made up of an echo estimator and a subtractor. The estimator monitors the received signal path and iteratively alters their characteristics in order to achieve an optimal desired output. The filter algorithmically alters its parameters in order to minimize a function of the difference between the desired output and its actual output. There are various adaptive algorithms for the AEC filter update, these are the least mean square (LMS), normalized mean square (NLMS), affine projection (AP), and recursive least square (RLS) algorithms. The Least Mean Squares (LMS) algorithm has been extensively used in various applications due to its robustness and simplicity.

II. Polyphase Allpass IIR Subband Adaptive Filtering
Multirate systems find application in communications, speech processing, spectrum analysis, radar systems and antenna systems. As the length of the adaptive filter increases, the computational complexity increases. This is a serious problem in acoustic applications such as echo and noise cancellation. In acoustic echo cancellation the highly correlated speech input signal and very large impulse response path of echo signal will slow down the convergence rate of adaptive filters if full-band adaptive filter is used. To solve this problems subband adaptive filters are used. The basic idea of Subband Adaptive Filter (SAF) is to use a set of parallel filters to divide the wideband signal into narrower band signals i.e. the digital signal is converted into multiple component signals, in which each component contains a small chunk of the spectral information of the original signal. Subband decomposition splits a full-band signal into multiple subband signals that allows for the processing of the information contained in each subband independently. In echo cancellation the excitation and microphone signals are split into several subbands signals, and then adaptive filtering can be applied to each signal. The subband decomposition is achieved by a set of bandpass filters called filterbank.

Fig. 2: Basic M-channel Subband Coding System

Subband analysis and synthesis is often performed using multirate filterbanks. Decimators and Interpolators are the basic building blocks in a multirate digital signal processing system [4-5].

A. Polyphase Decimation
The standard decimation method is computationally inefficient because it throws away the majority of the computed filter outputs [10]. To remove this inefficiency we must decompose the filter
H(z) into its polyphase components. The overall filter transfer function of a decimation filter \( H(z) \) is represented in the form

\[
H(z) = \sum_{k=0}^{M-1} z^{-k} H_k(z^M)
\]

Where \( H_k(z) \) is the kth polyphase IIR subfilter. The polyphase component structure for decimation is shown in Figure 3.

**B. Polyphase Interpolation**

The standard interpolation procedure is also computationally inefficient since the lowpass filter operates on a sequence that is mostly composed of zeros.

In Fig. 4 the output signal \( y[n] \) is produced by transforming the subsequences \( \{y_0[m], y_1[m], \ldots, y_{M-1}[m]\} \) into polyphase components of the output signal by upsampling by a factor \( L \) and adding a delay \( z^k (0 \leq k \leq M-1) \). For an interpolator, the overall transfer function is given by

\[
G(z) = L \sum_{k=0}^{L-1} z^{-k} G_k(z^L)
\]

Where \( G_k(z) \) is the kth polyphase IIR subfilter.

**C. Analysis and Synthesis Banks**

The two basic types of filter banks are analysis and synthesis banks. An analysis bank is a set of analysis filters \( H_k(z) \) which splits a signal into \( M \) subband signals [10]. A synthesis bank consists of \( M \) synthesis filters \( G_k(z) \), which combines \( M \) signals into a reconstructed signal. The analysis filter \( H_0(z) \) is of the form

\[
H_0(z) = \frac{A_0(z) + A_1(z)}{2}
\]

Where \( A_0(z) \) and \( A_1(z) \) are all-pass filters of the form

\[
A_0(z) = a_0(z), A_1(z) = z^2 a_1(z)
\]

Using polyphase IIR filter structures to perform the subband filtering [8] makes it feasible to decrease the number of calculations, convergence improvement and improving the accuracy of adaptation and allows efficient implementation. They are known to be very attractive for very high performance filters that can be designed using a very few coefficients.

**IV. Performance Parameter of AEC**

Performance parameters are the parameters by which we can judge the quality of the echo canceller. The performance parameters tell us about the performance of the acoustic echo canceller.

**A. Echo Return Loss Enhancement (ERLE)**

Echo Return Loss Enhancement (ERLE) [9] is tool to measure the potential of echo cancellation. It is defined as the ratio of power of desired signal to the power of the residual signal. ERLE is the most important measure of how much in dB the echo is suppressed by the acoustic echo canceller i.e., the ratio of the echo signal to the residual echo signal [4]. ERLE is determined in an echo cancelling arrangement by estimating an echo signal from a received signal and is subtracted from an incoming signal to produce an output signal. In order to evaluate the performance of an echo cancelling system, the ratio of the expected value of the microphone output signal...
squared $E[d(n)]$ divided by the expected value of the error signal squared $E[e(n)]$ is monitored. This quantity, in dB, is called the Echo Return Loss Enhancement, or ERLE. Mathematically, it can be expressed as

$$\text{ERLE} = -10 \log \left( \frac{E[d^2(n)]}{E[e^2(n)]} \right) \text{dB}$$

$$= 10 \log \left( \frac{P_d(n)}{P_e(n)} \right) \text{dB}$$

where $d(n)$ is desired signal and $e(n)$ is the residual signal. ERLE depends on the size of filter design and algorithm used. More the value of ERLE, the better is the echo canceller. For LMS algorithm, ERLE value for allpass filters in subband acoustic echo cancellation lies in the range $[-20 \text{dB}, 70 \text{dB}]$. The allpass filters in subband acoustic echo cancellation offers better performance when compared to fullband acoustic echo cancellation.

**B. Mean Square Error (MSE)**

Mean square error (MSE) is a dominant quantitative performance metric in the field of signal processing. MSE is a metric indicating how well a system can adapt to a given solution. A small minimum MSE is an indication that the adaptive system has accurately modeled, predicted and adapted and converges to a solution for the system. A very large MSE usually indicates that the adaptive filter cannot accurately model the given system or the initial state of the adaptive filter is an inadequate starting point to cause the adaptive filter to converge. There are a number of factors which will help to determine the minimum MSE including, but not limited to; order of adaptive system, quantization noise, measurement noise and error of the gradient due to finite step size. The mean square error is used as a performance index in sub-band acoustic echo cancellation systems. The MSE energy of residue echo is used to measure the performance. The MSE is defined as

$$MSE = \frac{P_e(n)}{P_d(n)}$$

**V. Simulation Results**

In this section we have performed a number of simulations to verify the performance of our proposed algorithm. Different lengths of adaptive filter with different value of step size have been studied. Fig. 7 shows the speech signal used. Figure 8 shows the comparison of magnitude response of FIR polyphase and IIR polyphase filter. Figure shows the ERLE and MSE plots of the echo canceller designed.
Fig. 10: MSE with filter length 8 and $\mu=0.01$

Fig. 11: ERLE with Filter Length 8 and $\mu=0.02$

Fig. 12: MSE with filter length 8 and $\mu=0.02$

Fig. 13: ERLE with filter length 8 and $\mu=0.08$

Fig. 14: MSE with filter length 8 and $\mu=0.08$

Fig. 15: ERLE with filter length 32 and $\mu=0.01$

Fig. 16: MSE with filter length 32 and $\mu=0.01$
VI. Conclusion

This paper investigated the use of polyphase allpass IIR filters in subbands for acoustic echo cancellation. Simulation results are presented comparing the different length and step size of adaptive filter. On comparing the simulation results of ERLE and MSE we can conclude that adaptive filter with long filter length i.e. L=32 offers better performance.

References