WIDEBAND ECHO CONTROL CHALLENGES

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1. INTRODUCTION

The introduction of wideband speech coding in telecommunication results in a significant improvement in the overall user experience. The frequency components from 50 to 200 Hz add to the naturalness and the components from 3400 to 7000 Hz add to the intelligibility of the speech. However, this increase in bandwidth introduces many challenges in speech processing and in particular echo canceller design.

Many of the traditional echo canceller algorithms work well on narrow-band signals but may not be suitable for wideband applications. Furthermore, most of the echo canceller performance standards are based on level calculation and do not discriminate between frequency bands. Some of the factors which contribute to the challenge include the obvious increase in the adaptive filter length and thus reduction in convergence speed as well as a significant increase in the required memory and CPU usage. Other less tangible factors include user expectations and possible increase in the perception of residual echoes. All of these factors need to be considered when choosing an algorithm for wideband echo cancellation.

This paper presents an overview of some of the possible candidates for echo cancellation algorithms in wideband telecommunication.

2. TYPES OF ECHO

There are two types of echo sources; electrical and acoustic. Electrical echo is caused by the hybrid (2 to 4 wire converter) found in the public switched telephone network (PSTN). The electrical echo impulse response is typically very short (< 8 ms) and the echo return loss (ERL) provided by the PSTN hybrid is relatively high (> 10 dB). In general, the electrical echo impulse response remains stable during a call and can easily be handled by an electrical (or line) echo canceller (LEC).

Acoustic echo is present when the signal from the loudspeaker is picked up by the microphone. This echo is more noticeable when a device is used in speaker mode. In this case, an acoustic echo canceller (AEC) must be used to remove this acoustic echo. Contrary to hybrid echo, the acoustic echo impulse response can vary significantly during a call. The ERL can also change during the call so the AEC must be able to adapt to the changing acoustic environment.

3. ACOUSTIC ECHO CANCELLATION METHODS

Acoustic echo cancellation algorithms use adaptive filters [1]-[4] to remove echo. Figure 1 shows a block diagram of a typical acoustic echo canceller algorithm. Echo is picked up by the microphone from a direct path and also from reflections on walls and other objects. For this reason, acoustic echo impulse response can be very high (> 100 ms) and represents a major challenge to AEC designers.

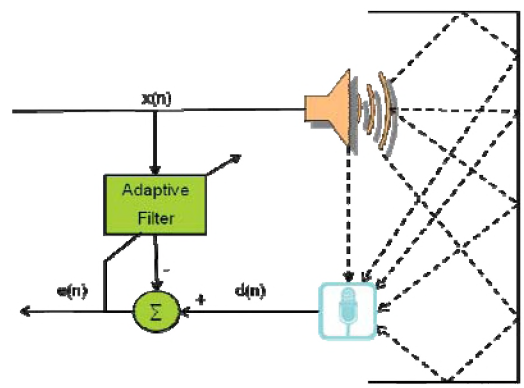

Figure 1. Acoustic Echo Canceller block diagram

The AEC input signal d(n) contains the near-end speech as well as the acoustic echo. A double-talk detector (DTD) is needed to detect the presence of near-end speech and freeze the adaptation process in order to prevent the adaptive filter from diverging. The acoustic echo ERL can be very low (~ 0dB) which produces a challenge for the DTD. A traditional level-based DTD is often not very effective for AEC.

There are many different types of algorithms used to implement the adaptive filter used by the AEC. In general, the AEC algorithms can be categorized as time-domain or frequency-domain.

4. TIME-DOMAIN AEC ALGORITHMS

In a time-domain adaptive filter AEC, an input signal x(n) is filtered by an adaptive filter. The output from this filtering operation is subtracted from a desired signal d(n) to produce the error signal e(n). This error is used to update the filter coefficients. The filter update is typically done using the Least Mean Squares (LMS) algorithm.

5. FREQUENCY-DOMAIN AEC ALGORITHMS

Traditional time-domain adaptive filter algorithms adapt filter coefficients on every input sample. This adaptation process requires significant amount of DSP resources. In order to reduce complexity, some frequency-domain approaches process the input signal in “blocks” of samples.

5.1 BFDAF Algorithm

The Block Frequency Domain Adaptive Filter (BFDAF) algorithm diagram is shown in figure 2. In this algorithm, the input samples are accumulated in blocks of N samples. The block of samples is then converted to the frequency domain using the Fast Fourier Transform (FFT). Filtering is accomplished by a multiplication in the frequency-domain which is equivalent to performing a convolution in the time-domain. The filter coefficient adaptation can be done directly in the frequency-domain using a LMS based process.

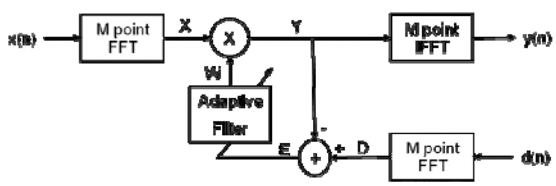

Figure 2. BFDAF AEC Block Diagram

There are two major drawbacks in the BFDAF algorithms. The first drawback is the delay introduced by the block processing. This delay can be significant as it is dictated by the length of the filter. The other significant drawback in the BFDAF algorithm is the speed of convergence since the adaptive filter coefficients are only updated once every N samples. To improve on the convergence speed and to reduce the processing delay, the multi-delay block frequency-domain adaptive filter algorithm [2] (MDF) was developed.

5.2 MDF Algorithm

The MDF adaptive filter algorithm diagram is shown in figure 3. In this algorithm, the input signal is accumulated into m*N blocks of samples. The size of the adaptive filter is controlled by the number of blocks (m) and the block length (N). This algorithm improves the speed of convergence significantly over the traditional BFDAF algorithm.

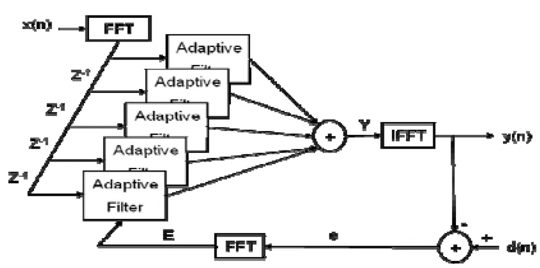

Figure 3. MDF AEC Block Diagram

Furthermore, the processing delay introduced by the MDF algorithm is smaller since the block length (N) is smaller than the BFDAF algorithm. The performance of the MDF algorithm is described in [2].

6. CHALLENGES

Acoustic echo cancellation in cellular telephony presents many challenges to DSP engineers. One of the challenges is the nature of the acoustic echo itself. The length of the adaptive filter (number of taps) required for the AEC can be very high, especially in wideband mode where the sampling frequency is 16 kHz.

Another important challenge that faces AEC designers is the environment in which the algorithm must operate. Mobile devices are often used in very noisy environment and the AEC algorithm must be robust to high levels of background noise. This also puts additional constraints on the design of the double-talk detector (DTD). This challenge is even more present in wideband telephony as more noise will be present along with the near-end speech.

The introduction of wideband devices on the market represents another important challenge for DSP algorithms developers. The first impact of wideband is the obvious increase in DSP resources (MIPS/memory) required by going from 8 kHz to 16 kHz sampling rate. The other impact of wideband is the user expectation. Residual echo artifacts will be more noticeable in a wideband cellular call. Wideband cellular users may also be less tolerant on echo artifacts as they are paying more for devices and will expect better voice quality.

7. CONCLUSIONS

This paper describes some basic acoustic echo canceller algorithms and challenges. It was shown that the MDF algorithm could be a good candidate to perform AEC in the context of wideband telephony. Furthermore, the MDF algorithm can be implemented to work for both narrowband and wideband mode by controlling the block length (N) at run-time. This paper aims to give algorithm developers an overview of challenges involved in the design of acoustic echo cancellation algorithms for cellular telephony.

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