Acoustic Echo Cancellation Employing Delayless Subband Adaptive Filters

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Abstract

The use of hands-free communication in cars, computer applications and video conferencing has created a demand for high-quality acoustic echo cancellation. In these applications the acoustic channel has typically a long impulse response in the order of 100ms. Typical lengths of adaptive FIR-filters can be 500-1500 taps. In order to reduce the complexity and also to improve the convergence rate, subband processing schemes have been suggested. This paper presents an implementation of a delayless subband adaptive filter. The study shows a possible suppression of about 30 dB and also a more rapid convergence compared to a fullband LMS-filter.

1 Introduction

In order to obtain a full-duplex hands-free communication, it is necessary to perform an acoustic echo cancellation of the far-end speaker[1, 2, 3]. In order to track variations in the acoustic channel the echo cancellation is often made adaptive. The filter length of the acoustic canceller is typically 500-1500 FIR taps. Long filters imply a large computational burden even with a simple adaptive filter algorithm such as LMS. Such filters also suffer from slow convergence rate, when the reference signal spectrum has a large dynamic range, i.e. a large eigenvalue spread in the corresponding signal correlation matrix. Subband techniques give a twofold advantage: the computational burden is essentially reduced by the number of subbands and it is also possible to get a faster convergence since the spectral dynamic range in each subband will be smaller [4, 5].

In this paper we present a slightly modified implementation of a delayless subband adaptive filter, a filtering scheme which was recently presented by Morgan and Thi [5]. This adaptive filter structure employs the benefits of adaptive subband filtering, but does not suffer from the inherent delay usually found in subband schemes. This is due to the fact that the FIR filtering is performed without delay directly on the full-band signal. The hands-free mobile phone needs a combination of acoustic echo cancellation and speech enhancement. Here, we concentrate on the acoustic echo cancellation. The filter has been implemented in real time on a TMS 320C31 processor.

The subband adaptive filter is presented in section 2. Section 3 presents the implementation and some results are presented in section 4. Finally section 5 discusses further improvements.

2 Subband Adaptive Filters

An acoustic echo canceller, see Fig. 1, identifies the total channel between the loudspeaker and the hands-free microphone. The identified impulse response is then used to electrically subtract the hands-free loudspeaker in the microphone signal and thereby inhibiting the echo received at the far-end. One of the fundamental characteristics of this channel is the bulk delay. A typical distance between loudspeaker and microphone is approximately 1 m. This corresponds to a 3 ms delay and with 8-12 kHz sample frequency this corresponds to about 20-30 samples. However, a 50 long FIR filter will only characterize the direct wave and give a suppression of about 5-10 dB. In order to achieve the suppression goal which is to suppress 30-40 dB, filter lengths of 500-1000 FIR taps are necessary.

The filter should also be able to follow variations in the acoustic environment. An appealing approach is to use a subband technique since this technique reduces the computational burden and also gives a

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faster convergence. The latter is due to the reduction of spectral dynamic range in each subband. A
major drawback is a delay that is introduced by the filter bank. This can however be circumvented by
using a modified structure for the subband adaptive filter [5].

2.1 The Delayless Subband Adaptive Filter

The delayless attribute of this technique comes from the fact that the new adaptive weights are computed
in subbands and then transformed to an equivalent full-band filter with means of an inverse FFT, see
Fig. 1. The filter works in real time on the loudspeaker signal. The coefficients are calculated separately
in each band. They can be calculated either by employing the error signal ε(k) or the microphone input
signal d(k). If the signal d(k) is used, a local error signal in each band is created and the calculations
does not need to be performed in real time. This will however give less suppression since the algorithm
is blindly working with respect to the real error signal. The full-band signal is divided into several
subbands signals by using a polyphase FFT technique [6].

A variation is when the outputs from the subband filters are downsampled only by a factor D=M/2.
This means that even subbands are centered at dc while odd subbands are centered at one half of the
decimated sampling frequency, see Fig 2. This fact must be considered in the polyphase filter bank.
Since we only consider full-band filters with real coefficients, it is enough to calculate M/2 complex
subband signals. The remaining signals can be found by utilising the complex conjugate symmetry.
This yields all bands except the mid subband M/2 (which is ignored in [5]). A complete treatment
is obtained by handling this band as follows: The input x[n] is first modulated with (−1)^n and then
filtered analogously to subband 0.

If the full-band filter has N taps, the filter length in each subband will be N/D. A N/D point FFT
will be calculated on the adaptive weights in each subband. These are subsequently stacked to form a
[0..N/2-1] element array. The array is then completed by setting element N/2 to zero and using the
complex conjugate of elements [1..(N/2-1)] in reverse order. Finally, the N element array is transformed
by a N point inverse FFT to obtain the full-band filter weights.

Figure 1: Delayless subband acoustic echo canceller
3 Implementation

The acoustic echo canceller has been implemented, in real-time on a DSP TMS320C31. There are two important observations to make before implementing the filter. First, the full-band filtering must be performed in real time, i.e. on a sample by sample basis. This implies that for each new sample a FIR filtering of the full-band filter must be done. Secondly, the adaptive filtering can be performed on buffered data if the microphone signal $d[k]$ is used for the adaptation. Preferably, the adaptive filtering should be performed at each sampling instant but this is not possible due to the computational load. Instead the loudspeaker signal $x[n]$ and the microphone signal $d[n]$ have been buffered. These buffered signals have been used to calculate the weights. In order to use the error signal $e[n]$ instead, the adaptation must run on a sample by sample basis, which requires a faster DSP unless shorter filters are accepted.

4 Results and Measurements

The results in this section have been produced by using the adaptive delayless subband adaptive filter. For both the Matlab simulations and the real time DSP implementation acoustic signals recorded on DAT tapes in a car environment have been used. In the Matlab simulations, typical cancellations of 23 dB were achieved with a 256 long full-band filter, i.e. 32 tap filters in each subband, see Fig. 3. The suppression was improved to approximately 30 dB for a 512 tap full-band filter. Figure 4, shows the suppression of a full-band LMS. The number of floating point operations were reduced with 20% with the subband approach. The adaptation and measurements have been evaluated by using bandlimited flat noise. The adaptation time shows an improvement as compared to the full-band filter, see Fig. 5. In the real-time implementation using a 256 long full-band filter, a suppression of approximately 16 dB have been achieved, see Fig. 6. However, there is a fundamental difference: The Matlab simulations were performed using a normalised LMS, which improved the convergence rate. Another limitation in the real-time implementation is the computational load which has limited the maximum FIR filter length to 256.

5 Conclusions and Further Improvements

The delayless subband adaptive filter has been an efficient tool to perform acoustic echo cancellation. Results show an improved convergence rate compared to a conventional LMS FIR filter. The real time implementation need to be further studied. Future plans are to apply the subband technique for active noise control applications.
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Figure 3: Suppression performance of a delayless adaptive acoustic echo-canceller(off-line simulation results).

Figure 4: Suppression curve of a full-band normalised LMS(off-line simulation results).
Figure 5: Learning curves for a full-band normalised LMS (dashed line) and a delayless adaptive acoustic echo-canceller (solid line). The filter weight error norm $||h - w||$ is used, where $h$ is the true impulse response of a known channel and $w$ is the adaptive FIR filter weights.

Figure 6: Suppression performance from the real time implementation.