Experimental Results of a Modified Architecture for Oversampled, Subband Acoustic Echo Cancellers

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Abstract—**The motivation for adaptive filtering in subbands stems from two well-known problems in LMS-based fullband adaptive filtering. First, the convergence and tracking can be very slow if the input correlation matrix is ill conditioned such as that found in speech. Second, very high order adaptive filters are computationally expensive. Adaptive filtering in subbands has been proposed to overcome these problems which are commonly encountered in acoustic echo cancellation applications. One problem with this technique is the slow, asymptotic convergence associated with oversampled architectures. A modification of the analysis bank is proposed to reduce the slow asymptotic convergence. This paper will present experimental results illustrating the benefits of this modified architecture.**

I. Introduction

In a hands-free communications system, acoustic coupling between the loudspeaker and microphone at the remote site produces an annoying echo for the local user. One method used to eliminate this echo is to use an adaptive filter to model the echo path with the output from this model subtracted from the output of the remote microphone. The desired end result is an echo-free signal.

Typical room reverberations can last on the order of 250ms depending on the size of the room among other things. At sampling rates of 16kHz this relates to the need for an adaptive filter on the order of 4000 taps. It is well known that very long adaptive filters are computationally expensive even when using such simple algorithms as the least-mean-square (LMS) algorithm [1]. Furthermore, the convergence and tracking of an LMS-based adaptive filter can be very slow if the input correlation matrix is ill conditioned such as that found in speech [1]. These problems make building an effective acoustic echo canceller (AEC) difficult [2].

One technique for overcoming these problems is to perform the adaptive filtering in subbands. The subband adaptive filter architecture, illustrated in Fig. 1, offers the possibility of performing the equivalent task of a fullband adaptive filter but with several key benefits that overcome some of the previously mentioned problems with the fullband architecture [2]. The first benefit is the possibility of fewer computations per input sample than when using a single adaptive filter since the subband signals are at a lower sampling rate and the adaptive filters are shorter in length. The second benefit is that convergence time may be decreased since the input signal is spectrally decomposed and each adaptive filter is working only on a portion of the input. In this case, each adaptive filter is in a way specialized to the subband input and may adapt at a faster or slower rate independent of the other adaptive filters.

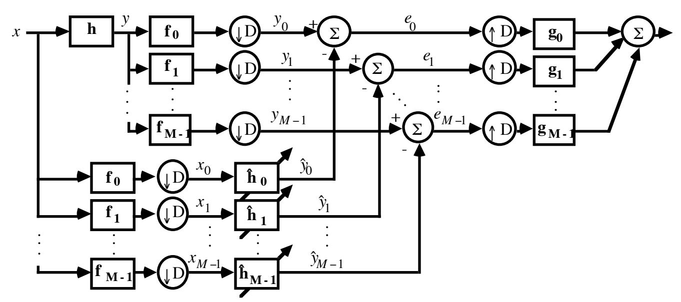

Figure 1: $\frac{M}{D}$ oversampled, M-subband AEC.

Research in AEC systems using the subband adaptive filter architecture has shown modest success. In order to reduce the effects of aliasing caused by downsampling, oversampled ($D < M$) schemes are often used instead of critically-sampled ($D = M$) schemes which require additional adaptive cross filters to compensate for the effects of aliasing [2]. Fig. 2 illustrates typical normalized computational complexities (subband / fullband) for both $2\times$ oversampled and critically sampled (without cross-filters) AECs as a function of the number of subbands.

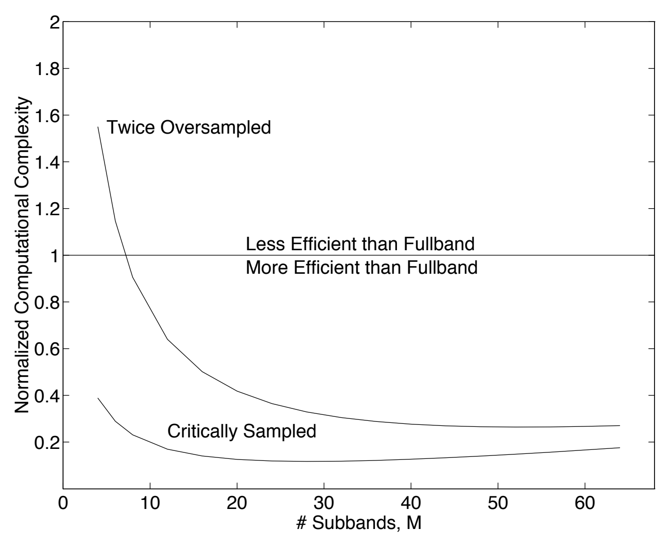

Figure 2: Normalized computational complexity for the subband AEC.

Experiments with the oversampled, subband AEC architecture have demonstrated a slow asymptotic convergence of the subband mean-squared error (MSE) (solid line, Fig. 5) and consequently a slow asymptotic convergence of the fullband MSE (solid line, Fig. 6). Furthermore, large spectral peaks are observed in the subband error signal at the band edge (solid line, Fig. 4). A slight modification of the filter bank architecture is proposed to reduce the band edge spectral components of the subband error signal and reduce the slow asymptotic convergence. In this paper, results from experiments based on this modified architecture are presented.

II. Modified AEC Architecture

The large, band edge spectral components in the subband error signals could be filtered out with simple bandpass filters before synthesis. However, doing this will introduce spectral gaps in the reconstructed error signal and lead to amplitude distortion. An alternative method is to increase the bandwidth of the analysis filters and then use the standard bandwidth synthesis filters to not only eliminate the images from upsampling but to also shave off the band edge components.

The experiments with this proposed modification use a 2×-oversampled, 4-subband AEC as in Fig. 1 ($M = 4, D = 2$) and are performed using computer simulations (100 simulations are performed and the results are averaged). A measured acoustic echo path of 512 taps was used to simulate room reverberation and white noise was used as input. Efficient polyphase, uniform-DFT filter banks were used for analysis and synthesis and the prototype filters were designed using the Parks-McClellan algorithm [3]. The -3dB down frequency of the analysis prototype filter was set to $\frac{1}{2M} + \frac{\delta}{2}$ where δ is the amount of bandwidth increase, assuming a normalized sampling frequency, $f_s = 1$. The -3dB down frequency of the synthesis prototype filter was set to $\frac{1}{2M}$. Finally, the normalized LMS algorithm was used to update the adaptive filters [1].

An upper bound on δ is computed as follows. Assume that the prototype analysis filter for the M subbands analysis filter bank has significant (at least -70dB) stopband attenuation at all frequencies beyond f_c and a downsampling factor of D is used in the system. Therefore the analysis filter prototype should be designed so that the band edge of the subband signal is located no further than the Nyquist frequency of the downsampled signal. This requires:

$$\delta \le 2\left(\tfrac{1}{2D} - f_c\right). \tag{1}$$

Choosing δ within this bound should not significantly increase the aliasing in the AEC since only the signal beyond f_c (which is small by design) is aliased. The previously described analysis filter prototype used in the experiments had

$$f_c = 0.144. \tag{2}$$

Substitution of (2) into (1) with $D = 2$ yields a maximum value for δ of

$$\delta \le 0.2120. \tag{3}$$

In the experiments, four evenly-spaced values of δ were used from 0 to half the bound in (3). These values were:

$$\begin{aligned} \delta_0 &= 0 \text{ (no bandwidth increase)} \\ \delta_1 &= 0.0353 \\ \delta_2 &= 0.0707 \\ \delta_3 &= 0.1060 \end{aligned} \tag{4}$$

Fig. 3 contains a plot of the magnitude responses of the analysis filter prototypes for the various values of δ as given in (4).

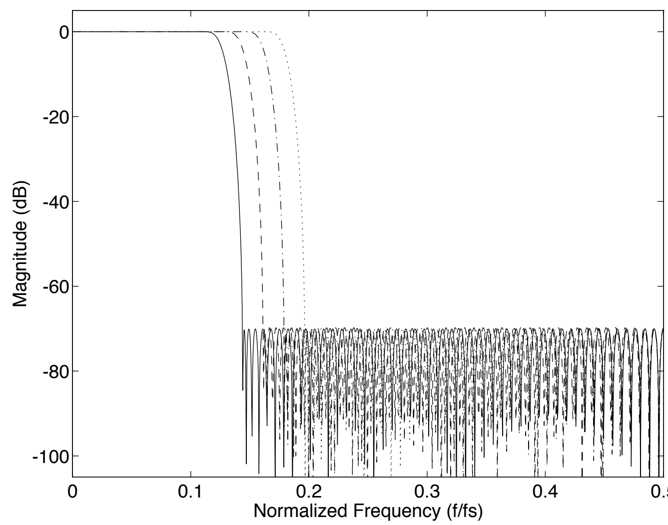

Figure 3: Analysis filter prototypes used in experiments.
δ_0 — ; δ_1 -- ; δ_2 -· ; δ_3 ···

III. Experimental Results

As the bandwidth of the analysis filter is increased, the band edge spectral components of the subband error signal are shifted beyond the cutoff of the synthesis filters as illustrated in Fig. 4.

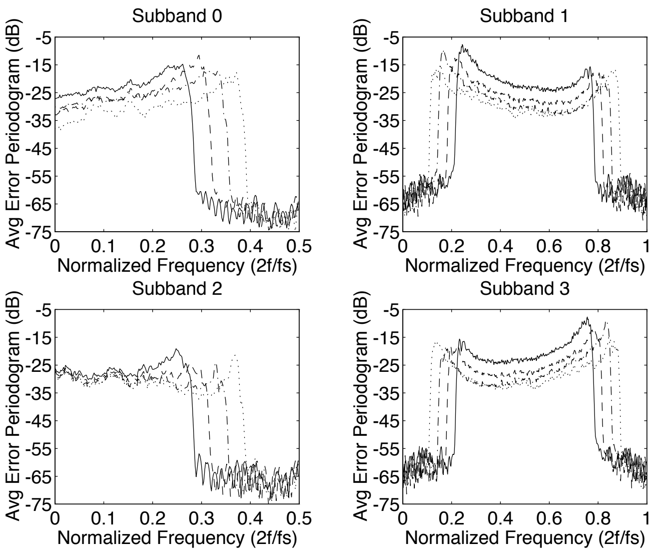

Figure 4: Average spectrum of subband error signal at $k = 4000$.

δ_0 — ; δ_1 – – ; δ_2 –· ; δ_3 ····

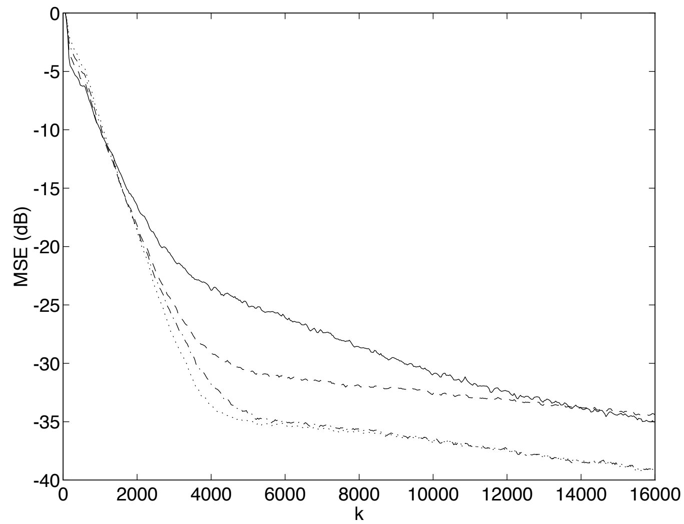

Figure 6: Normalized MSE of AEC with white noise input.

δ_0 — ; δ_1 – – ; δ_2 –· ; δ_3 ····

Shifting the slowly-converging spectral components of the subband error signal further out has the effect of slightly increasing the normalized subband MSE convergence as seen in Fig. 5.

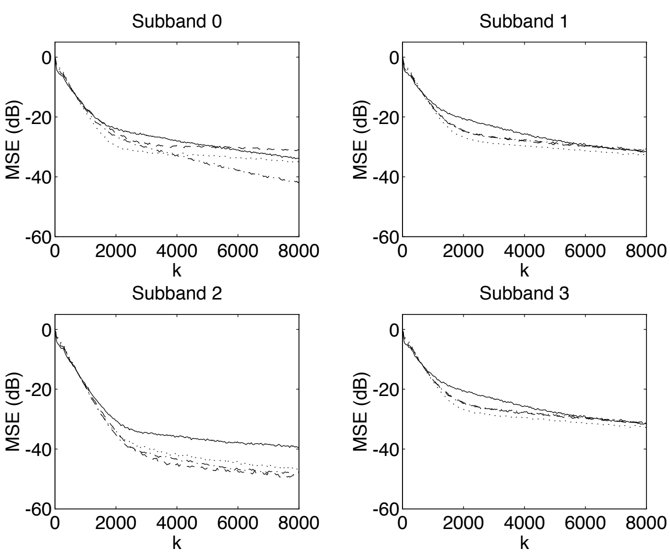

Figure 5: Normalized subband MSE.

δ_0 — ; δ_1 – – ; δ_2 –· ; δ_3 ····

The final result of shifting the large spectral components of the subband error signal, where they are shaved off by the synthesis filters, is the increase in the convergence of the fullband error signal as illustrated in Fig. 6.

The subband AEC with analysis filter bandwidth increased by δ_3 now attains -35dB of echo cancellation in about 16000 samples compared to about 5000 samples for the conventional subband AEC—a factor of 3.2 better. An alternate evaluation is to note that after 4000 samples, the MSE for the conventional AEC is down about -34dB as compared to the AEC with analysis filter bandwidth increased by δ_2 which is down about -24dB.

IV. Conclusion

Experimental results with oversampled, subband AEC architectures with increased bandwidth analysis filters have demonstrated better convergence than those architectures without the increase. The increased bandwidth analysis filters are used in conjunction with the synthesis filters to remove the slowly converging, band edge spectral components. This modification adds no additional computational overhead to the system and is thus an inexpensive improvement.

References

[1] S. Haykin, *Adaptive Filter Theory*. Englewood Cliffs, N.J.: Prentice-Hall, 1991.

[2] A. Gilliore and M. Vetterli, "Adaptive Filtering in Subbands with Critical Sampling: Analysis, Experiments, and Application to Acoustic Echo Cancellation," *IEEE Trans. Signal Processing*, vol. 40, pp. 1862-1875, Aug. 1992.

[3] R. E. Crochiere and L. R. Rabiner, *Multirate Digital Signal Processing*. Englewood Cliffs, N.J.: Prentice-Hall 1983.