Experimental Results with Increased Bandwidth Analysis Filters in Oversampled, Subband Acoustic Echo Cancellers

Phillip L. De León II, *Student Member*, *IEEE* and Delores M. Etter, *Fellow*, *IEEE*

***Abstract*—The motivation for adaptive filtering in subbands stems from two well-known problems in least-mean square fullband adaptive filtering. First, the convergence and tracking can be very slow if the input correlation matrix is ill-conditioned as in the case with speech input. Second, very high order adaptive filters are computationally expensive. One problem with adaptive filtering in subbands is the slow, asymptotic convergence associated with oversampled systems. Increasing the bandwidth of analysis filters relative to the synthesis filters is proposed to reduce the slow asymptotic convergence. This letter will motivate this approach and present experimental results illustrating the benefits of this modification.**

I. Introduction

The motivation for adaptive filtering in subbands stems from two well-known problems in fullband adaptive filtering. First, the convergence and tracking of a least-mean square (LMS) adaptive filter can be very slow if the input correlation matrix is ill-conditioned as in the case with speech input [1]. Second, very high order adaptive filters are computationally expensive. One application that has both speech input and typically needs a very high order adaptive filter is acoustic echo cancellation (AEC) [2].

The subband adaptive filter system, 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 system [2].

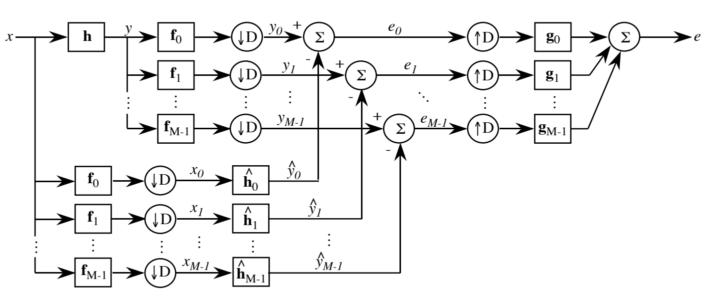

Figure 1: $\frac{M}{D}$ oversampled, M-subband AEC system.

Research in AEC using the subband adaptive filter system has shown modest success. In order to reduce the effects of aliasing in these systems, oversampled schemes ($D < M$) are often used [3] instead of critically-sampled ($D = M$) schemes which require additional adaptive cross filters to compensate for aliasing effects [2].

Theory [4] and experiments with the oversampled, subband AEC system have demonstrated a slow asymptotic convergence of the subband mean-squared error (MSE) and consequently a slow asymptotic convergence of the fullband MSE. Furthermore, large spectral peaks are observed in the subband error signal at the band edge. These problems can be traced back to the fact that the oversampled, subband input signal will likely generate an ill-conditioned correlation matrix. The ill-conditioning can be explained by Szegö's theorem on the asymptotic eigenvalue distribution of Toeplitz matrices [5]. In this case, the small eigenvalues are generated by the rolloff of the subband input power spectrum resulting in an asymptotic convergence of the subband MSE as well as large, band edge spectral components. In [4], postfiltering the band edge components is suggested as a remedy for this slow asymptotic convergence. In this letter, results from experiments based on this modification are presented.

II. Modified AEC System

One method to remove the band edge components might use bandpass filters but this has the undesirable effect of introducing spectral gaps in the reconstructed fullband signal. The alternate method proposed here is to increase the bandwidth of the analysis filters ($\mathbf{f}_0, \mathbf{f}_1, \cdots, \mathbf{f}_{M-1}$) relative to the synthesis filters ($\mathbf{g}_0, \mathbf{g}_1, \cdots, \mathbf{g}_{M-1}$). This has the effect of shifting the slowly converging spectral components of the subband error signals further out in the spectrum. Subsequent synthesis filtering removes these components from each subband and thus out of the fullband signal thereby increasing the convergence of the MSE.

The experiments with this proposed modification use a 4/3-oversampled, 4-subband AEC as in Fig. 1 ($M = 4$, $D = 3$). A simulated acoustic echo path ($\mathbf{h}$) was used with white noise input (x) and the results were averaged over 100 runs. Uniform-DFT filter banks were used for analysis and synthesis. The analysis and synthesis prototype filters were designed using the Parks-McClellan algorithm. The -3dB down frequency of the analysis prototype filter was set to $\frac{1}{2M} + \frac{\delta}{2}$

The authors are with the Department of Electrical and Computer Engineering, University of Colorado at Boulder.
This work was supported by the AT&T Cooperative Research Fellowship Program.

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 (NLMS) algorithm was used to update the adaptive filters.

An upper bound on δ is computed as follows. Assume that the prototype analysis filter has significant (at least -70dB) stopband attenuation at all frequencies beyond $f_{c'} + \frac{\delta}{2}$ 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:

$$\delta \le 2\left(\frac{1}{2D} - f_{c'}\right). \tag{1}$$

Choosing δ within this bound should not significantly increase the aliasing in the AEC since only the signal beyond $f_{c'} + \frac{\delta}{2}$ (which is small by design) is aliased. The previously described analysis filter prototype used in the experiments had:

$$f_{c'} = 0.143 \tag{2}$$

which can be seen with the solid line in Fig. 2. Substitution of (2) into (1) with $D = 3$ yields an upper bound for δ of:

$$\delta \le 0.0473. \tag{3}$$

Four values of δ were used in the experiments: 0%, 33%, 67%, and 100% of the bound so that $f_{c'} + \frac{\delta}{2}$ was evenly spaced from the original stopband frequency of the analysis prototype filter to the maximum given by $f_{c'} + \frac{0.0473}{2}$. Fig. 2 contains a plot of the magnitude responses of the analysis filters for the various values of δ.

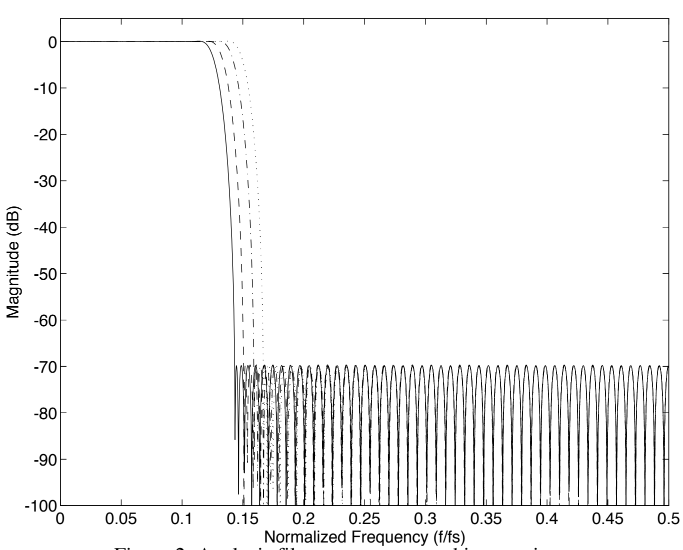

Figure 2: Analysis filter prototypes used in experiments.

III. Experimental results

As the bandwidth of the analysis filter is increased, the band edge spectral components of the subband error signals are shifted beyond the cutoff of the synthesis filters as illustrated with subband 0 in Fig. 3 at sample index $k = 5000$. Note that for δ equal to the upper bound in (3), the band edge spectral components are excited to a higher degree than the others. This is because of a spectral peak in the echo path located along this band edge. In general this is unpredictable, since the echo path is unknown.

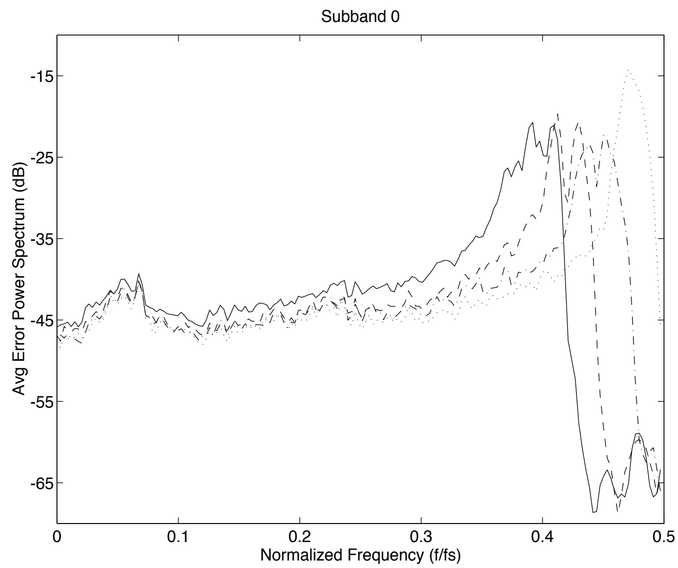

Figure 3: Average power spectrum of subband 0 error signal at $k = 5000$.

As illustrated in Fig. 4, there is minor difference between the subband 0 MSEs for the various values of δ (with the exception of δ equal to the upper bound in (3)). This exception is a result of the effect noted above.

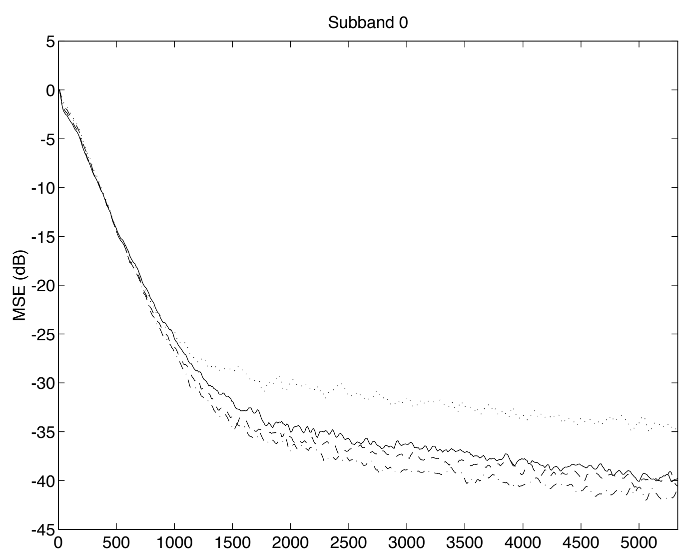

Figure 4: Subband 0 MSE.

The final result of shifting the large spectral components of the subband error signal, which are removed by the synthesis filters, is the increase in the convergence of the fullband error signal as indicated in Fig. 5.

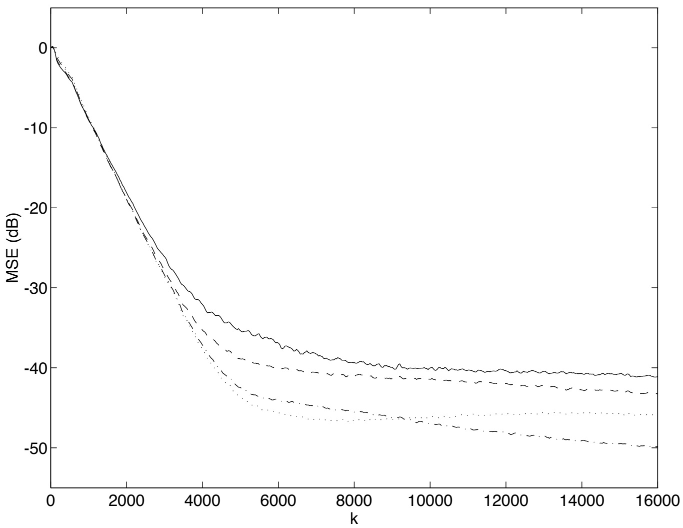

Figure 5: MSE of AEC.

The subband AEC with maximally increased analysis filter bandwidth now attains -40dB of echo cancellation in about 4500 samples compared to about 9500 samples for the conventional subband AEC (no analysis filter bandwidth increase)—a factor of 2.1 better. An alternate evaluation is to note that after 5000 samples, the MSE for the conventional subband AEC is down -35dB as compared to the AEC with maximally increased analysis filter bandwidth which is down -45dB.

IV. Conclusion

Experimental results with the oversampled, subband AEC system with increased bandwidth analysis filters have demonstrated better convergence than conventional designs. The increased bandwidth analysis filters are used in conjunction with the synthesis filters to remove the slowly converging, band edge spectral components. For fixed D, this modification adds no additional computational overhead to the system and is thus an inexpensive improvement.

Acknowledgment

We wish to thank Dennis R. Morgan for his contributions to this work.

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] M. M. Sondhi and W. Kellerman, "Adaptive echo cancellation for speech signals," in *Advances in Speech Signal Processing*, S. Furui and M. M. Sondhi, Eds., New York: Marcel Dekker, 1992, ch. 11.

[4] D. R. Morgan, "Slow asymptotic convergence of LMS acoustic echo cancellers," submitted for publication in *IEEE Trans. Speech and Audio Processing*.

[5] R. M. Gray, "On the asymptotic eigenvalue distribution of Toeplitz matrices," *IEEE Trans. Information Theory*, vol. IT-18, pp. 725-730, Nov. 1972.