Examining the Effects of Room Response in Oversampled, Subband Acoustic Echo Cancelers

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Abstract

In this paper we examine the effects of the room response on the convergence of oversampled, subband acoustic echo cancelers. Specifically, we have observed that spectral features of the room response such as peaks or nulls have definite and measurable effects on the convergence of the subband MSE which in turn has an effect on the reconstructed MSE. These effects can be traced back to slow converging spectral components located at the analysis filter band edge.

1: Introduction

The subband adaptive filter system (Figure 1) has been studied in the adaptive identification of systems with very long impulse responses. Subband systems offer the possibility of reduced computation and convergence time for spectrally dynamic input [1]. These properties have made the subband adaptive filter system popular for acoustic echo cancellation applications.

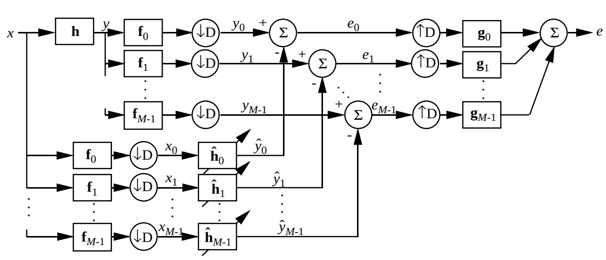

Figure 1: Subband adaptive filter system (*M*-subbands, *M* / *D* oversampling).

Asymptotic convergence of oversampled, subband adaptive filters has been shown to be caused by small eigenvalues located near the band edges of the analysis filters [2]. One technique has been proposed to overcome this slow convergence [3] but large variations in convergence time have still been observed. These variations are mainly due to the variations in band edge excitation energy resulting from the room response. In this paper, we demonstrate the above observations for various room responses.

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2: Background Theory

The mean-square error for subband m at sample k (on the downsampled time scale) is defined as:

$$\xi_{m,k} = \mathrm{E}\left[\left|e_{m,k}\right|^2\right]. \qquad (1)$$

The subband MSE for the LMS adaptive filter can be computed with [4]

$$\xi_{m,k} = \xi_{m,\min} + \sum_{n=1}^{N} \left|\phi_{m,n}^H \hat{\mathbf{h}}_{m,\mathrm{opt}}\right|^2 \lambda_{m,n} \left(1-\mu_m \lambda_{m,n}\right)^{2k} \qquad (2)$$

where $\xi_{m,\min}$ is the minimum mean-squared error (MMSE), $\left|\phi_{m,n}^H \hat{\mathbf{h}}_{m,\mathrm{opt}}\right|^2 \lambda_{m,n}$ is the nth modal power, and $\left(1-\mu_m \lambda_{m,n}\right)^{2k}$ is the nth decay rate, all for subband m. Each of the N "modes" in the summation of (2) is composed of a modal power and a decay rate. The calculation of these subband quantities is given in [5]. As shown in [2], $\lambda_{m,1}, \cdots, \lambda_{m,N}$ can be approximated by uniformly-spaced samples of the subband input power spectrum. This leads to a range of eigenvalues spanning the spectral dynamic range of the analysis filter. This fact suggests that the smaller (band edge) eigenvalues lead to slower decay rates. Clearly, the room responses which have the smallest band edge modal powers will have the fastest band edge convergence. Since band edge components dominate the sum in (2), we expect these same room responses to have the smallest subband MSE [2]. The modal power magnitudes, $\left|\phi_{m,n}^{H}\hat{\mathbf{h}}_{m,\text{opt}}\right|^2\lambda_{m,n}$ will be calculated in the next section for various room echo paths.

3: Room response effects on convergence

The subband adaptive filter system (Figure 1) used in this study has 4-subbands (M = 4) which are 4 / 3 oversampled (D = 3). Input to the system is zero mean, unit variance white Gaussian noise. The analysis filters, $\mathbf{f}_0,\cdots,\mathbf{f}_3$ are uniform-DFT filters [6] and the prototype low pass filter, $\mathbf{f}_0$ (129 coefficients) is designed with the Parks-McClellan algorithm. The synthesis filters, $\mathbf{g}_0,\cdots,\mathbf{g}_3$ are equal to the analysis filters, respectively. The echo path filter, $\mathbf{h}$ has 512 coefficients and each subband adaptive filter, $\hat{\mathbf{h}}_0,\cdots,\hat{\mathbf{h}}_3$ has N = 171 (512 / D) coefficients. The LMS algorithm [4] is used to update each subband adaptive filter with the step size for each adaptive filter, $\mu_m = \dfrac{1}{3N\mathbf{f}_m^T\mathbf{f}_m}$ where $\mathbf{f}_m^T\mathbf{f}_m$ is the input power to the mth subband [7].

To illustrate the variation in convergence characteristics, simulations with the subband adaptive filter system are performed with four different echo path filters: ideal, notch, room A, and room B which are explained as follows. The ideal echo path filter, which yields a flat frequency response and uniformly excites all modes, provides a "convergence reference" and is described by

$$h_n = \begin{cases} 1, & n = 10 \\ 0, & \text{otherwise} \end{cases}\cdot \qquad (3)$$

The notch echo path is designed so that spectral nulls are located at the crossover frequencies between each of the subbands as in Figure 2.

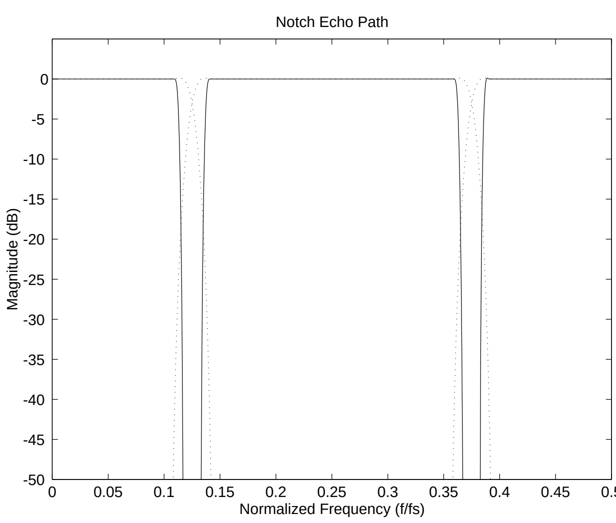

Figure 2: Frequency responses of notch echo path (solid line) and analysis filters (dotted line).

The reason for using this filter in the demonstration is to minimize band edge excitation energy or the eigenvector projection magnitudes corresponding to the band edge modes. By reducing the modal power at the band edge, the subband MSE should be correspondingly reduced. Room A and B are actual room echo paths (measured at AT&T Bell Laboratories [8]) whose frequency responses are illustrated in Figures 3 and 4.

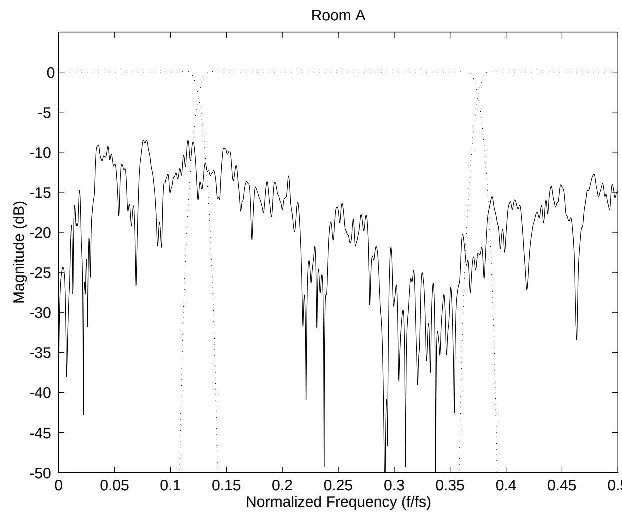

Figure 3: Frequency responses of Room A echo path (solid line) and analysis filters (dotted line).

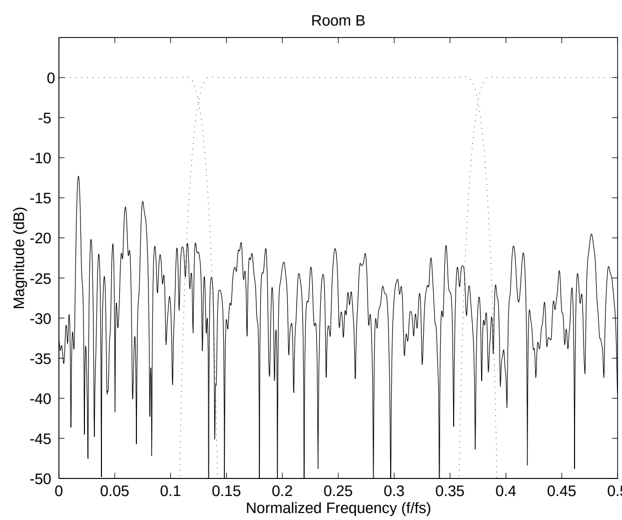

Figure 4: Frequency responses of Room B echo path (solid line) and analysis filters (dotted line).

The eigenvalues for the input correlation matrix are given in Figure 5. As mention earlier, these eigenvalues are roughly equal to uniformly spaced samples of the input spectrum and the smaller eigenvalues lead to slower decay rates. The subband eigenvector projection magnitudes, $\left|\phi_{m,n}^{H}\hat{\mathbf{h}}_{m,\mathrm{opt}}\right|^{2}\lambda_{m,n}$ are computed for the various room echo paths and the results for subband 2 are given in Figure 6.

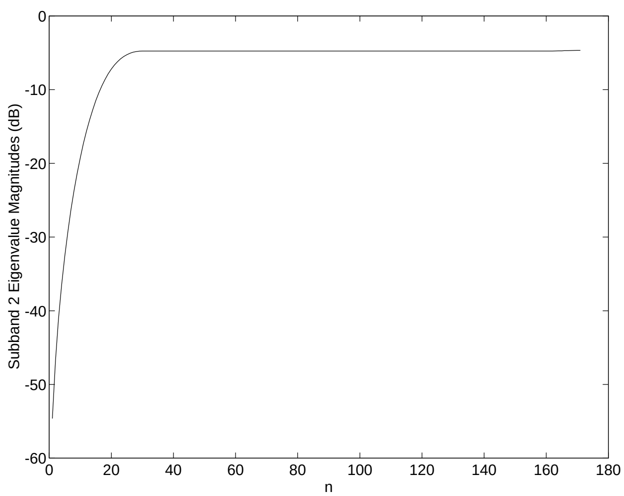

Figure 5: Subband 2 input correlation matrix eigenvalues.

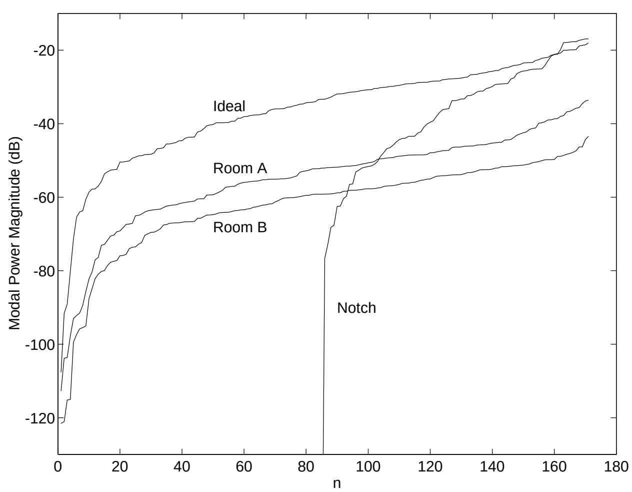

Figure 6: Subband 2 modal power magnitudes for the various echo paths.

Given the above results, it is clear for subband 2, that the modal powers associated with the band edge eigenvalues (which lead to the slowest decay rates), are greatest for the ideal echo path followed by Room A, Room B, and finally the notch echo path. Since these modal powers along with the decay rates determine the magnitude of the subband MSE, we expect the subband MSE to be smallest for the ideal echo path, followed by Room B, Room A, and finally the ideal echo path should have the largest subband MSE. Figure 7 confirms these expectations.

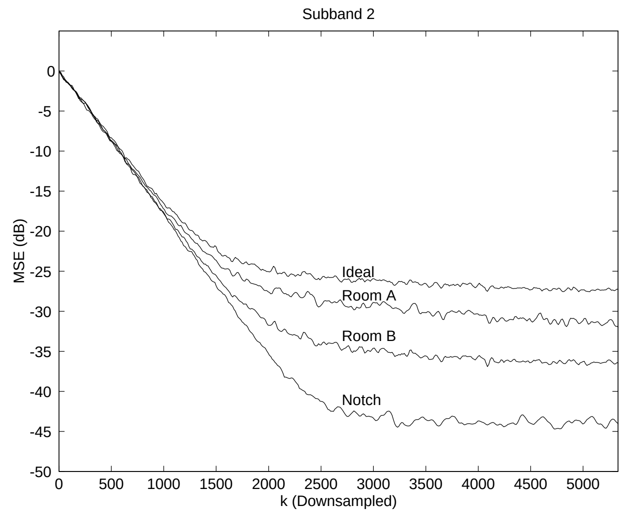

Figure 7: Subband 2 MSE curves various echo paths.

The reconstructed MSE (defined as $\xi_k = \mathrm{E}\left[\left|e_k\right|^2\right]$) curves for the various simulations are illustrated in Figure 8. As speculated the system with the notch echo path has the fastest convergence because of the smallest overall band edge modal powers for the subbands. The system with notch echo path has smallest MSE, followed room B and then room A (after k = 13000). The largest MSE (after k = 13000) is for the system with the ideal echo path.

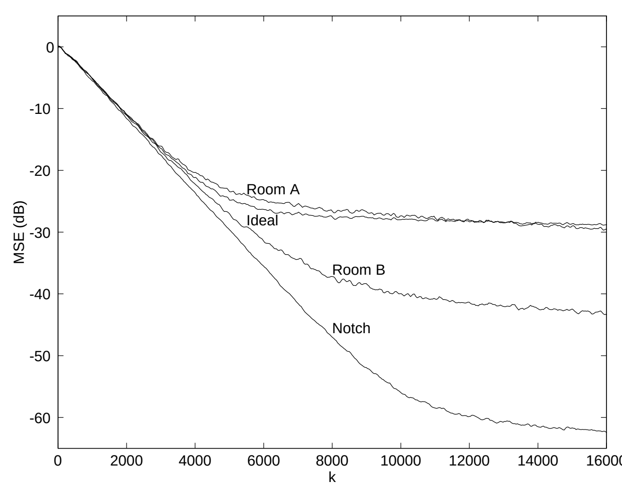

Figure 8: MSE curves for subband adaptive filter systems with various echo paths.

4: Conclusions

In this paper we have examined the effects of various room responses on the convergence of oversampled, subband acoustic echo cancelers. As a general characteristic, room responses with spectral nulls near the band edges of the analysis filters lead to smaller band edge modal powers. Smaller band edge modal powers have the fastest convergence due to a minimization of the effects of slow converging, band edge modes.

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