NON-WIENER PHENOMENON IN OVERSAMPLED SUBBAND ADAPTIVE FILTERS

B. N. M. Laska†, F. Beaucoup‡, and R. A. Goubran†

†Department of Systems and Computer Engineering
Carleton University, Ottawa, Ontario, Canada
{laska,goubran}@sce.carleton.ca

‡Mitel Networks
Ottawa, Ontario, Canada
franck_beaucoup@mitel.com

ABSTRACT

Traditionally held beliefs assert that the minimum mean-square error (MMSE) of subband adaptive filters is higher than that of the equivalent fullband filter. In this paper we show that this is not always the case. Oversampled subband adaptive filters with white and speech-like excitation are observed to produce MMSE levels well below those of the equivalent fullband filter and even below the linear and time-invariant (LTI) Wiener filter. This observation is attributed to non-Wiener behavior, which has previously been observed in adaptive filters with narrowband inputs.

Index Terms — Adaptive filters, Wiener filtering, channel bank filters

1. INTRODUCTION

Subband adaptive filters have previously been proposed for acoustic echo cancellation to overcome the problems of high complexity and slow convergence of long adaptive filters [1]. Figure 1 shows a block diagram of a subband adaptive echo canceller. The far-end speech signal *x(n)*, and the measurement signal *y(n)* containing the echo signal *d(n)* and the near-end speech signal *v(n)*, are decomposed into a set of *M* subbands by a bank of analysis filters. The signals are then downsampled by a factor $D \leq M$, and used as inputs to a bank of adaptive filters which together form an estimate of the echo path vector $\underline{h}(n)$. This downsampling allows the filter adaptation to be carried out at a lower rate, reducing complexity. The subbanded signals are also more spectrally flat, which improves convergence for colored inputs. Subband structures also offer flexibility in terms of double-talk control and post-processing.

Despite the advantages of subband processing, there are also well known limitations. The analysis and synthesis filterbanks increase the delay of the system, which is undesirable for speech communication. Furthermore, while the minimum mean-square error (MMSE) of all adaptive

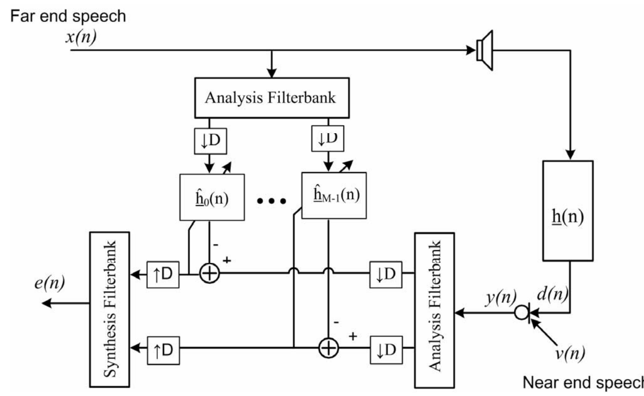


Figure 1. Subband adaptive echo canceller.

filters is limited by factors such as background noise, under-modeling of the echo path impulse responses, and non-linear distortion from hands-free terminals, the performance of subband adaptive filters is further limited by imperfect filterbanks.

Non-ideal filterbanks have finite stopband attenuation and non-zero-width transition bands. As a result, critically sampled subband signals contain aliasing distortion, which increases the MMSE [2]. Subband adaptive filters also need to model non-causal taps in the echo-path impulse response [3]. Under-modeling of the non-causal taps has the same negative impact on MMSE performance as under-modeling the tail of the impulse response.

A known compromise that can compensate for some of the limitations imposed by subbanding is oversampled subband adaptive filtering [4], where the downsampling factor *D* is less than the number of subbands *M*. Non-critical sampling reduces aliasing and leads to relaxed frequency domain requirements on the analysis filters. This allows for shorter filters, which reduces the delay. However, as the oversampling factor increases and the downsampling factor is lowered, the system complexity increases, reducing one of the main benefits of subband adaptive filtering. It is, therefore, generally accepted that there exists a trade-off between MMSE performance, delay, and subband system complexity. It is also believed that subband filters can only approach the MMSE performance of fullband filters under ideal conditions.

* This work was supported in part by the Ontario Centre of Excellence, CITO division, NSERC and Mitel.

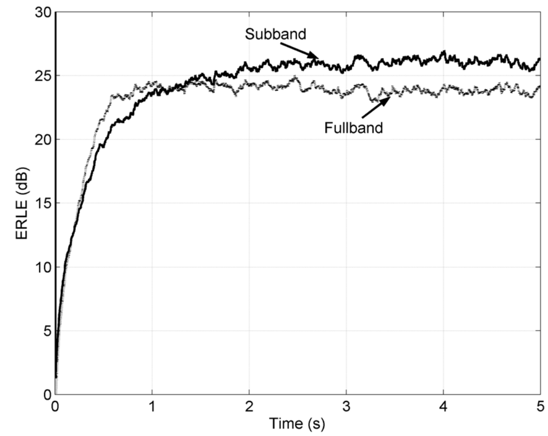
In this paper we show that the MMSE of oversampled subband filters can actually be much lower than that of fullband adaptive filters in general scenarios. We then show how this increased performance is due to non-Wiener effects that have previously been demonstrated to occur in adaptive filters with narrowband input signals.

2. MMSE PERFORMANCE OF OVERSAMPLED SUBBAND ECHO CANCELLERS

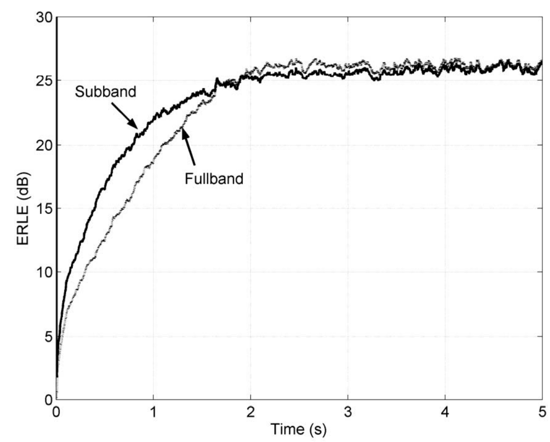
An experiment was performed to compare the echo cancellation capabilities of fullband and oversampled subband normalized least-mean-squares (NLMS) adaptive filters for white noise excitation and echo recorded in a 20' by 40' conference room using a commercial hands-free terminal. The subband system used an 8-channel 2-times-oversampled polyphase filterbank with 64-tap analysis filters designed using the approach in [4]. This length of analysis filter provided a stopband attenuation of 90 dB, ensuring that aliasing would not factor into the results. The fullband and subband filter lengths were chosen to enable modeling of a 150-ms echo path at 16-kHz sampling rate, and the NLMS step-size was $\mu = 1.0$.

Figure 2(a) shows the echo return loss enhancement (ERLE) of the two systems. The slower asymptotic convergence of the subband adaptive filter was expected, and is known to relate to slowly converging modes of the subband filters near the band edges [5]. More notable was the fact that the steady-state ERLE of the subband system is higher than that of the fullband system. Reducing the step-size to $\mu = 0.25$ equalizes the MMSE performance, as can be seen in Fig. 2(b). The MMSE of the fullband structure decreases, which is expected due to decreasing gradient noise, while the subband MMSE actually increased marginally. To investigate the impact of step size on subband and fullband steady-state ERLE, we ran tests under more controlled conditions.

Simulations were run using white noise excitation and a measured 250-ms acoustic-echo-path impulse response. Figure 3 shows steady-state ERLE as a function of step-size for a fullband system and an 8-channel subband system with varying degrees of oversampling. No noise was added; therefore the achievable MMSE was limited by system under-modeling. The Wiener ERLE is indicated by the dashed black line. The fullband structure behaves as expected: for small step sizes the ERLE is close to the Wiener level, and as the step size increases, the ERLE decreases due to increasing gradient noise. In the 3/2 times and 2 times oversampled subband systems, however, the ERLE remains relatively constant, and in the 4-times oversampled system it actually increases until a maximum is reached and the gradient noise begins to dominate. Also remarkable is the fact that the maximum ERLE in the 4-times oversampled case exceeds the Wiener level.



(a)



(b)

Figure 2. ERLE for white inputs. a) $\mu = 1.0$ b) $\mu = 0.25$.

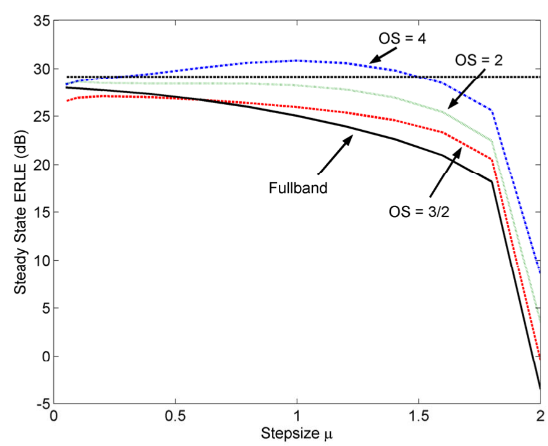


Figure 3. ERLE vs. step-size for white noise input, oversampled subband NLMS adaptive filters.

To verify that the results were not isolated to under-modeling noise, we performed another experiment (not shown) where white noise was added to the measurement signal. While the absolute positioning of the curves changed, the shape and relative placement remained the same.

3. SUB-WIENER MMSE IN ADAPTIVE FILTERS

Fullband adaptive filters have been previously shown to achieve MMSE levels below those of the linear time invariant (LTI) Wiener filter in adaptive equalization, linear prediction, and noise cancellation applications where the input signals are narrowband and the adaptive step-size is large. This behavior has been termed "non-Wiener performance" and a thorough examination of the topic can be found in [6], where it is demonstrated that the performance gain is caused by the adaptive filters exploiting correlation information not available to the LTI Wiener filter.

Consider the operation of an NLMS adaptive filter as an example. In NLMS, the error signal sample at time n is given by:

$$e(n) = d(n) - \underline{\hat{h}}^T(n)\underline{x}(n)$$

where $\underline{x}(n) = [x(n),...,x(n-N-1)]^T$ is the excitation vector and N is the adaptive filter length. At each iteration, the filter weights are adapted to minimize the *a-posteriori* error, the error that would have been produced at that iteration if the modified tap weight vector were used. If consecutive reference data vectors and desired signal samples are strongly correlated, then minimizing the *a-posteriori* error of the current iteration will, on average, reduce the *a-priori* error of the next iteration.

In [6] it is demonstrated that the performance bound for an adaptive filter is not given by the LTI Wiener filter, but rather by a Wiener filter with access to the infinite past of both the reference and desired signals. This infinite-horizon Wiener filter can be equivalently modeled by a time-varying finite-horizon Wiener filter that takes signal correlation into account. For sufficiently large step sizes and correlated inputs, adaptive filters can track the input signal and approximate the behavior of the time-varying optimal filter to reduce the output error.

Figure 4 shows the steady-state ERLE as a function of step size for fullband filters with increasingly correlated excitation signals. To generate the colored inputs, we filtered white noise by lowpass filters with cutoff frequencies of ω_c = 3/4π, 1/2π and 1/4π respectively, and stopband attenuation of 90 dB. As expected, the more highly colored the input signal is, the higher the steady-state ERLE.

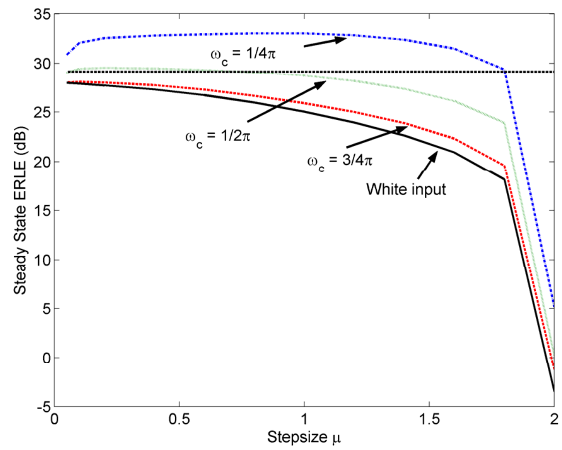


Figure 4. ERLE vs. step-size for lowpass-filtered inputs, fullband NLMS adaptive filters.

4. NON-WIENER BEHAVIOR IN OVERSAMPLED SUBBAND ADAPTIVE FILTERS

In oversampled subband adaptive filters, signal correlation is introduced by the subbanding process. Bandpass filtering by the analysis filterbank creates a narrowband, highly correlated signal. When that signal is downsampled by a non-critical factor, the resulting subband signal is still colored. Consequently, oversampled subband filters receive correlated inputs and can therefore exhibit non-Wiener behavior for a broader class of input signals than fullband filters do. This can be seen by re-examining Figure 3. The ERLE performance of the increasingly oversampled subband systems for white inputs corresponds almost exactly to the performance of the fullband systems with increasingly lowpass inputs in Figure 4.

In [7] it is observed that oversampling subband adaptive echo cancellation filters helps mitigate the effect of non-causal taps on the MMSE. The phenomenon is attributed to the low-energy regions corresponding to the analysis filter stopband offering extra degrees of freedom to the allowable subband impulse response. Although no link was made in [7] to the step size or the Wiener performance, it appears that the phenomenon is the same as the one we observed with truncated impulse responses. In both cases, the under-modelling of the subband filters is compensated for by the non-Wiener behavior.

In [8], near-end signal cancellation is observed in an oversampled subband echo canceller using the affine projection algorithm (APA). This cancellation is attributed to non-Wiener behavior triggered by the narrowband reference signals produced by oversampling. This behavior is seen as negative as it distorts the near-end signal, and a regularization parameter is developed to prevent its occurrence. It is also noted that the amount of near-end signal distortion increases not only with the degree of oversampling, but also with the affine projection order. This observation fits with the theory of non-Wiener performance: since APA has direct access to more distant past inputs, it should be better able to exploit correlation.

Figure 5 plots ERLE as a function of step size for white inputs to oversampled subband adaptive filters with varying degrees of oversampling using APA with a projection order of 3. For step sizes around $\mu = 1.5$ the steady-state ERLE of the 4 times oversampled structure is almost 20 dB in excess of the Wiener level.

Our observations show that subband filters are capable of producing sub-Wiener MMSE levels for white inputs. To examine the phenomenon with more realistic signals that are non-stationary and more instantaneously band-limited, we used the ITU composite source signal (CSS) as the excitation for a fullband and a 2-times oversampled 8-channel subband echo canceller. Both structures used APA with a projection order of 3, a step size of $\mu = 1.0$, and 150ms. adaptive filters. The ERLE results are presented in Fig. 6 where the LTI Wiener level is also indicated. The oversampled subband structure still achieves ERLE levels over 5 dB higher than the Wiener filter.

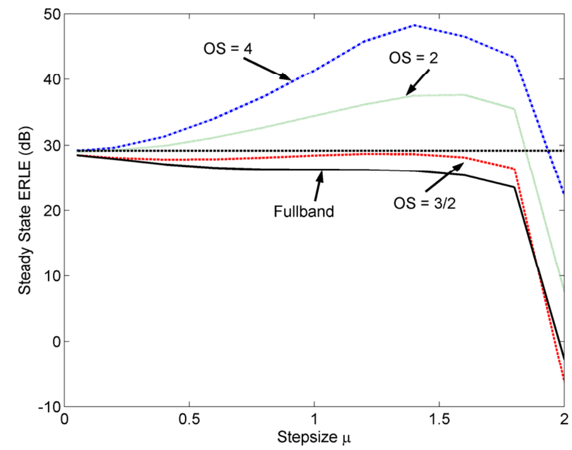


Figure 5. ERLE vs step-size for white noise input, oversampled subband APA adaptive filters.

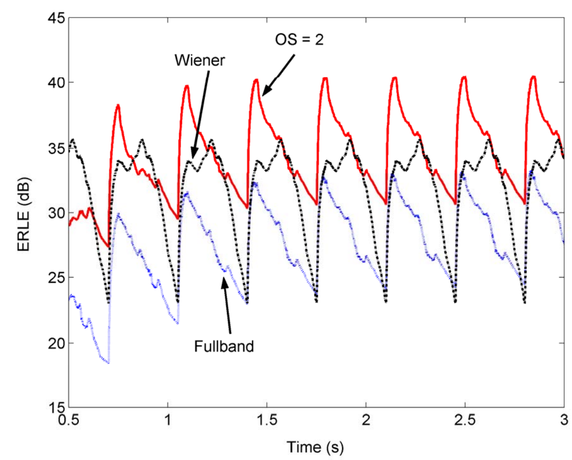


Figure 6. ERLE for CSS input.

5. CONCLUSIONS

When designing subband adaptive filters, it is known that there are compromises with respect to signal delay, computational complexity, and MMSE performance. This work presents yet another factor to consider. While traditionally held beliefs assert that the MMSE of subband adaptive structures is higher than that of the equivalent fullband filter, we show that this is not always the case. Oversampled subband adaptive filters with white and speech-like excitation are observed to produce MMSE levels well below those of the equivalent fullband filter, and even below those of the LTI Wiener filter. This observation is attributed to non-Wiener behavior, which has previously been observed in adaptive filters with narrowband inputs.

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