ADAPTIVE ACOUSTIC ECHO CANCELLATION BASED ON FIR AND IIR FILTER BANKS

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ABSTRACT

In this paper we investigate various subband AEC systems in real handsfree situation, including FIR and IIR analysis and synthesis QMF filterbanks in polyphase structure. The adaptation in the subbands is performed by the Affine Projection Algorithm in comparison to the NLMS algorithm. The IIR filters are superior to FIR filters in the sense that they lead to low signal delay and sharp frequency separation. Furthermore the computational complexity is greatly reduced by the use of IIR filters. The results show that splitting the signal into more subbands has advantages. Both wideband and narrowband speech signals have been evaluated.

1. INTRODUCTION

Acoustic Echo Cancellation (AEC) is one of the challenging applications of adaptive filtering in telecommunications. For a comfortable handsfree communication, the echo generated by the loudspeaker-room-microphone (LRM) system must be suppressed. As the principle is well known, the AEC device provides an adaptive estimate of the room impulse response usually in direct linear transversal form. Hence the echo is compensated by subtraction of its estimate from the near-end input signal.
Several methods have been proposed to overcome the limitations of the classical full-band NLMS-FIR AEC. Subband adaptive filtering has the potential advantage to improve convergence speed and to reduce computational complexity ([4], [3]). Note that the complexity reduction can be substantial as it is roughly proportional to the number of subbands. Also, the convergence speed of the NLMS algorithm can be increased by subband filtering since different parameter values can be assigned to each subband. However, the use of non-ideal critically sampled FIR multirate filterbanks leads to aliasing in the subbands which disturbes convergence and deteriorates the AEC performance. Non-critical subsampling, cross-adaptive filters, oversampling and introduction of gaps between subbands are some of the proposed methods to solve the aliasing problem [3]. A further method is to use IIR filters instead of linear-phase FIR designs for the half-band filter [6].
On the other hand, there has been a significant progress on the adaption algorithm side as well. Besides improvements on the NLMS-scheme, further algorithms have been worked out to improve convergence speed at moderate computational cost [5], [7], [8]. The Fast Affine Projection (APA) seems to be one of the most promising approaches [8].
In our work, we investigate and compare the performance of various AEC schemes using speech signals recorded in a real car environment. We consider both FIR and IIR filters for the analysis/synthesis filterbanks. A various number of subbands is taken into account. We use the NLMS and APA adaption algorithms for FIR-adaptation in the subbands. For the APA, various projection dimensions are investigated. Finally, our work includes the evaluation of both narrowband (standard telephony bandwidth at 8 kHz sampling frequency) and wideband (50 Hz – 7 kHz) speech signals.

2. FILTERBANK DESIGN

A two channel filterbank consists of signal decomposition by highpass and lowpass half-band filters combined with a downsampling process. The data are reconstructed by upsampling and filtering as shown in Fig. 1. Note that signal splitting in more than two subbands can be achieved by the tree structure [9]. Aliasing cancellation at the synthesis

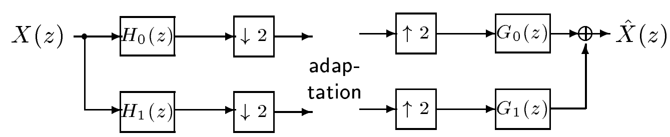

Figure 1: Two channel filter bank

bank output is achieved by [1]

$$G_{00} = 2H_{00} \quad and \quad G_{01} = 2H_{01}. \tag{1}$$

when Quadrature Mirror Filters (QMF) are used with $H_1(z) = H_0(-z)$. Under these conditions, the transfer function becomes allpass independently of the type and design methodology of the $H_0(z)$ and $H_1(z)$ half-band filters. The corresponding polyphase structure [9] is depicted in Fig. 2. The analysis stage is the same in both IIR and FIR filter realization, while the synthesis stages slightly differ for the IIR case as shown in Fig. 3. The dependency between the analysis and synthesis filters (Eq. (1)) holds in both cases.

In the linear-phase FIR implementation of the polyphase structure, the half-band filter was designed by the standard

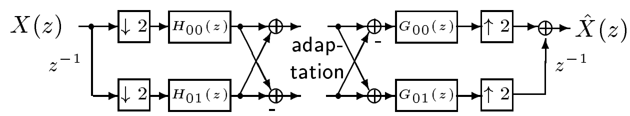

Figure 2: Two channel QMF FIR filter bank in polyphase structure

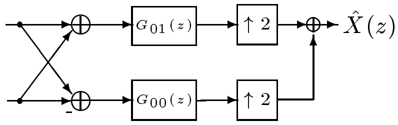

Figure 3: QMF IIR synthesis stage in polyphase structure

Parks-McClellan method. Note that FIR filters lead to a signal delay depending on the number of filter coefficients used. However, the most disturbing effect for the AEC is the aliasing caused by flat filter slopes. To eliminate aliasing, we used a bandstop filter between the subbands.
Unlike FIR filters, IIR filters are able to provide a much sharper cutoff slope. We adopted the design method presented in [6]. The IIR half-band filters are arranged as a cascade of first-order allpass filters. To produce the filter coefficients, an odd-order elliptic lowpass filter is designed by conventional bilinear transformation. If the passband and stopband ripples are kept symmetrical, the poles of the elliptic filter become imaginary and so the allpass coefficients $d_k, k = 1, 2, ..., N$ can easily be derived [9] from the imaginary poles of the lowpass transfer function using the pole interlace property [2]. Note that aliasing still occurs when using an IIR filter but it is more concentrated to the band edge. Therefore, a biquadratic notch can be added for aliasing cancellation.

3. ADAPTATION ALGORITHMS

The NLMS algorithm is one of the algorithms we applied in our experiments with subband AEC. It is a stochastic gradient type algorithm which iteratively seeks the minimimum of the mean square error surface. Normalization is essential for instationary signals, e.g. speech. We recall the defining equations for the sake of completeness: [1]

$$e(t) = y(t) - \mathbf{h}(t-1)^T \mathbf{x}(t) \tag{2}$$

$$\mathbf{h}(t) = \mathbf{h}(t-1) + \mu \frac{\mathbf{x}(t)}{\mathbf{x}(t)^T \mathbf{x}(t)} e(t) \tag{3}$$

where $\mathbf{x}(t)$ represents the input signal vector (far-end signal), $y(t)$ the desired response (near-end signal), $\mathbf{h}(t)$ the adaptive filter coefficients, μ the step size and $e(t)$ the error signal (AEC output).
The second algorithm in our experiments was the APA. It can be viewed as a generalization of the NLMS algorithm [8]. Each coefficient update can be interpreted as an one-dimensional projection [4]. Generally, the APA uses p-dimensional projection. Recently, several efforts have been made (e.g. [8]) towards fast versions which reduce the high computational complexity of [7] $(p+1)L + O(p^3)$. The original APA is given by

[1]Bold printed lower case letters indicate vectors, higher case letters denote matrices.

	NLMS	Fast APA
full	$2L$	$2L + 20p$
sub FIR	$\frac{2L}{n} + log_2 nb + nk$	$\frac{1}{n}(2L + 20p) + log_2 nb + nk$
sub IIR	$\frac{n}{2}i + \frac{2L}{n} + log_2 nc$	$\frac{n}{2}i + \frac{1}{n}(2L + 20p) + log_2 nc$

Table 1: Comparison of computational complexity between NLMS and Fast APA in subband and in fullband mode

$$\mathbf{e}(t) = \mathbf{y}(t) - \mathbf{X}(t)^T \mathbf{h}(t-1) \tag{4}$$

$$\mathbf{h}(t) = \mathbf{h}(t-1) + \mu \mathbf{X}(t)(\mathbf{X}(t)^T \mathbf{X}(t))^{-1} \mathbf{e}(t) \tag{5}$$

The matrix $\mathbf{X}(t)$ is an $(L \times p)$ matrix of the last $L + p - 1$ input values, $\mathbf{e}(t)$ the error vector and $\mathbf{y}(t)$ the vector containing the desired signal. Note that $\mathbf{e}(t)$ and $\mathbf{y}(t)$ are vectors in contrast to the NLMS case and they contain past $p - 1$ values besides the current value. Table 1 compares the computational complexity of the NLMS and fast AP Algorithm, where n denotes the number of used subbands. Furthermore, k indicates the number of FIR filter coefficients, b the number coefficients of the bandstop filter used in the FIR case, c the notch filter coefficients in the IIR case and i the overall number of allpass coefficients.

4. EXPERIMENTAL RESULTS

Speech signals recorded in a real car environment were used in the experiments. The free dimensions of the experiments were: FIR or IIR filterbank, number of subbands, NLMS or APA adaption algorithm, order of APA projection dimension, narrowband (standard telephony bandwidth at 8 kHz sampling frequency) and wideband (50 Hz – 7 kHz) speech signals. Both male and female speech samples were evaluated. The results of the experiments were first compared by the Echo Return Loss Enhancement (ERLE) calculated over a sliding window as follows:

$$\mathrm{ERLE}_k[dB] = 10 \log \frac{\sum\limits_{i=k-\lambda}^{k} y_i^2}{\sum\limits_{i=k-\lambda}^{k} e_i^2} \tag{6}$$

where y_i denotes the near-end signal and e_i is the AEC output.
Wideband signals turned out to have generally about 10dB better compensation results than narrowband signals regardless of the choice of the half-band filter type while the complexity was kept constant. The increase from 4 to 8 subbands produces about 5 dB better compensation obviously due to the more selective frequency splitting. However, an increasing number of subbands causes a larger signal delay in each of the analysis/synthesis stages which is especially critical when FIR filters with a high number of filter coefficients are used. On the other hand we have found that in combination with a matched bandstop even an FIR filter with 8 coefficients can yield satisfactory adaptation. The

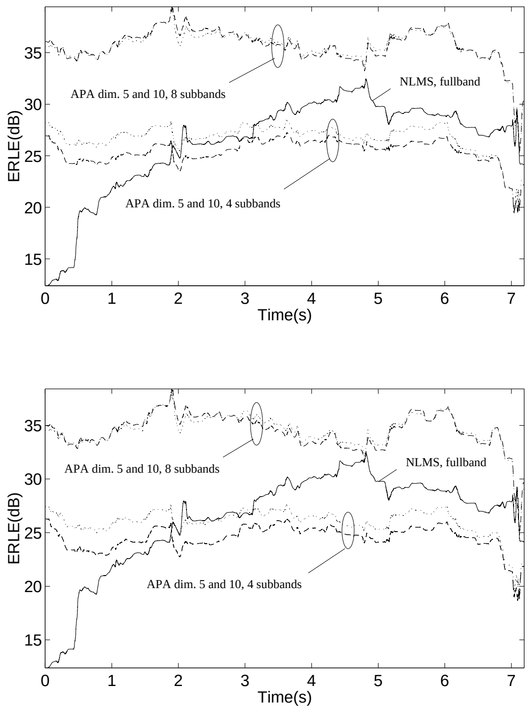

Figure 4: ERLE curves for various order of projection dimensions in fullband and subband mode for an FIR (top) and IIR QMF bank (bottom), narrowband signal

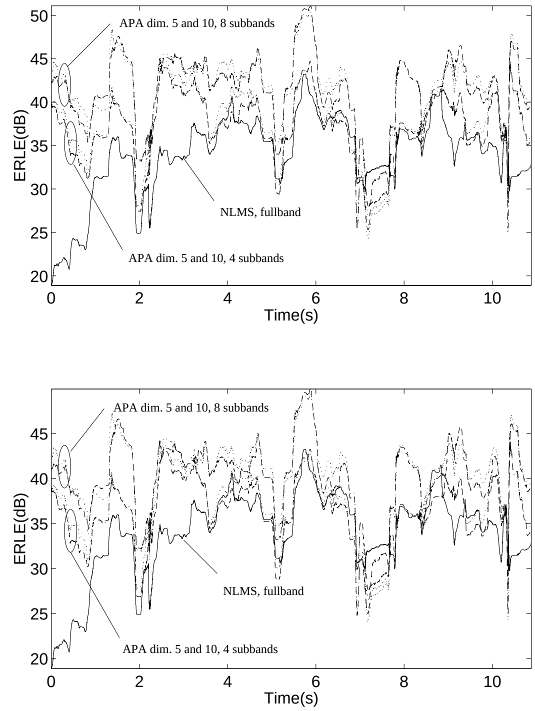

Figure 5: ERLE curves for various order of projection dimensions in fullband and subband mode for an FIR (top) and IIR QMF bank (bottom), wideband signal

inclusion of spectral gaps by the bandstop filter was found to be indispensable to obtain useful AEC performance.

The behaviour of the AEC did not actually change when an IIR filter replaces the FIR+bandstop combination. Furthermore, the combination of the IIR filter with a biquadratic IIR notch was found to be sufficient instead of the more complex bandstop. Note that the FIR version generally would not converge with a notch filter. An interesting observation when using IIR filters as compared to FIR filters in the filterbank is that, above certain limits, the choice of the filter order or the attenuation level of the filter has no significant effect on the AEC performance. In addition, we found that a group delay equalization has no advantages which might be expected to be the case because of the highly nonlinear IIR group delay characteristics. In other words, the linear-phase property of FIR half-bands cannot really be exploited in the AEC application. In conclusion, the use of IIR half-bands and biquadratic notch filters for the polyphase filterbank is preferred over the FIR solution.

The use of different projection dimensions for the APA in each subband reduces the computational complexity. However, for projection orders greater than 5, the adaptation achieved by this structure is not as effective as when assigning equal projection dimensions to all subbands.

Figs. 4 to 7 show several narrowband and wideband results for various projection dimensions. For comparison, a reference NLMS fullband AEC is also included. The two subfigures present the ERLE corresponding to FIR and IIR analysis/synthesis filters, respectively. The use of APA improves the convergence speed over NLMS adaptation in the subbands. The dotted curves represent the APA in dimension 10, the dashed ones the APA dimension 5.

Finally, we note that our informal listening test results fully correlate with the above results based on ERLE.

5. CONCLUSION

The performance of various subband AEC systems has been experimentally investigated and compared in a real handsfree environment. Both quality (ERLE, listening tests) and complexity aspects and signal delay have been taken into account. The use of IIR polyphase multirate filterbanks for AEC has proven to be an effective approach. It is clearly superior to the FIR design. The inclusion of a simple notch filter at the half-band was found to be advantageous. AEC adaption by affine projection generally outperforms the NLMS scheme, especially if the projection dimension varies over subbands. Splitting into more subbands had advantage over fewer subbands. Finally, a greater echo compensation could be achieved for wideband speech than for narrowband speech.

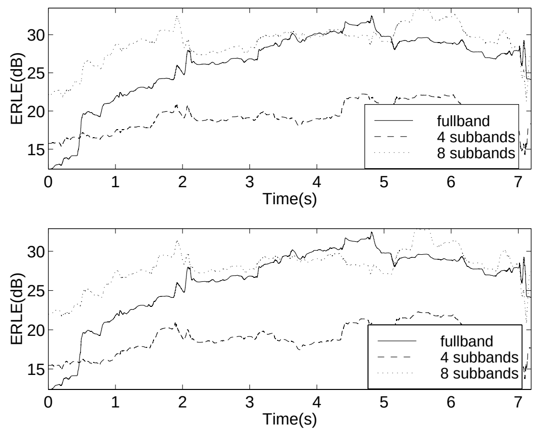

Figure 6: ERLE curves for NLMS in fullband and subband mode for an FIR (top) and IIR QMF bank (bottom), narrowband signal

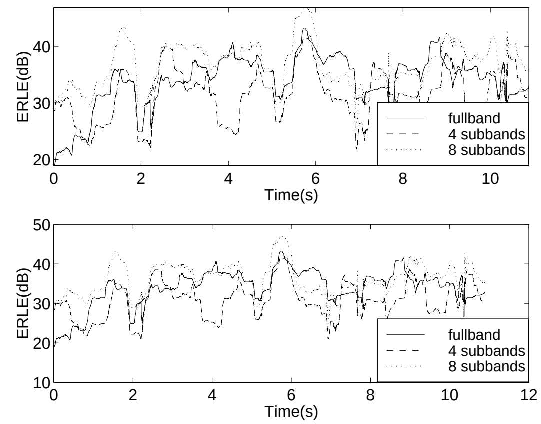

Figure 7: ERLE curves for NLMS in fullband and subband mode for an FIR (top) and IIR QMF bank (bottom), wideband signal

6. REFERENCES

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