AN OVERSAMPLED SUBBAND ADAPTIVE FILTER WITHOUT CROSS ADAPTIVE FILTERS

M Harteneck, R W Stewart

Signal Processing Division
Dept. of Electrical and Electronic Engineering
University of Strathclyde
Glasgow, G1 1XW
Scotland, U.K.
email: moritz@spd.eee.strath.ac.uk

J M Páez-Borrallo

ETSI Telecomunicación
Universidad Politécnica de Madrid
Ciudad Universitaria s/n
28040 Madrid
Spain
email: paez@gaps.ssr.upm.es

ABSTRACT

Recently a new real-valued oversampled filter bank has been proposed which reduces the "inband" alias. The filterbank consists of at least three channels which are subsampled by different subsampling ratios. In this paper we investigate into the applicability of this filterbank for adaptive subband filtering and compare the setup with existing subband and fullband techniques.

1. INTRODUCTION

One of the main problems in acoustic echo cancelation is the long duration of the impulse response of the echo path which can require adaptive filters with up to several thousand taps to give a good echo reduction. Adaptive filters of such high orders are difficult to implement due to their computational complexity and furthermore they exhibit slow convergence and a high minimum mean squared error which is emphasized by the colored and time varying nature of the input signal in this type of application.

One approach to solve these problems is to use subband architectures in conjunction with adaptive filters. It is anticipated that the adaptive filters in the subbands require less taps due to the subsampling process and that the input signals appear "whiter" as only a narrow frequency band is used. However, a closer analysis of the problem [3] shows that with critically subsampled filterbanks adaptive crossfilters between adjacent subbands are necessary which increase the computational complexity of the adaptive algorithms and reduce the convergence speed as the input signals to the multichannel algorithms, which are used in the subbands, are correlated. These crossfilters are necessary because the subsampling process causes "inband" aliasing and therefore the subband input signal and the subband desired signal cannot be compared directly as information is lost. Efforts to reduce this "inband" aliasing has been made [6] where the analysis and synthesis filters were constructed from IIR filters with a very narrow transition band and a high stopband attenuation. These IIR filter banks however show a poor performance in terms of signal-to-noise ratio in the reconstructed signal.

Another approach to solve these problems is to use oversampled filterbanks with the adaptive filters [2, 5]. In this filterbank structure "inband" aliasing is not present if the analysis filters are complex modulated versions of a lowpass prototype. One drawback of these structures is that the subband signals are then complex valued and therefore the computational complexity of the subband adaptive filters is increased.

In this paper we investigate in the applicability of a real-valued oversampled filterbank which yield subband signals without aliasing [4].

2. FILTERBANK DESCRIPTION

The main problem of subband adaptive filtering is the "inband" alias which occurs if a real-valued analysis filter $H_i(z)$, whose channel is subsampled by S_i, contains the normalized frequency points $\frac{2\pi l}{S_i}$, $l = 1, \ldots, S_i - 1$ [4]. Therefore, to avoid "inband" alias, all the analysis filters need to have spectral nulls at these frequencies, if the same subsampling ratio S_i is to be used for all channels. This however is a contradicting requirement to the demand to have a perfect or near-perfect reconstruction filterbank and so common spectral nulls in all channels are ruled out [7].

Therefore, in [4], a filterbank design is proposed which

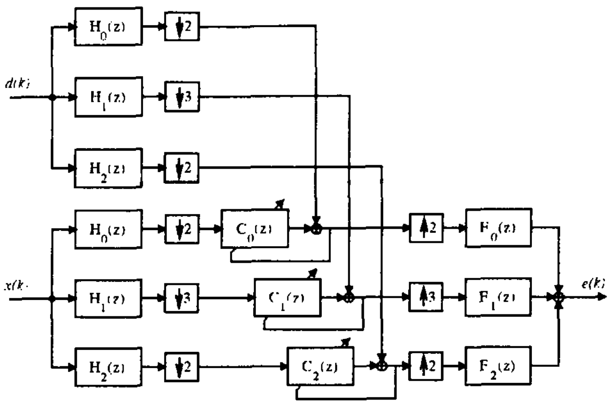

Figure 1: Block Diagram of Proposed 3 Channel Structure

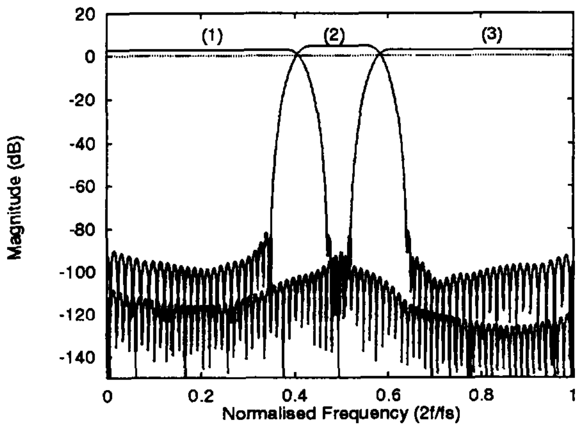

Figure 2: Analysis Bank of Proposed Structure: (a) $H_0(z)$, (b) $H_1(z)$, (c) $H_2(z)$

circumvents the inband aliasing by choosing different subsampling ratios S_i for different channels i. The simplest filterbank presented consists of 3 channels where two channels are subsampled by 2 and one channel by 3, which gives an oversampling ratio of 133%.

Fig. 1 shows such a filterbank in an adaptive subband filter setup, i.e. two analysis banks which analyze the input signal $x(k)$ and the desired signal $d(k)$ and one synthesis bank which reconstructs the error signal $e(k)$ from its subband signals. In this case channel 0 and 2 are subsampled by 2 and channel 1 is subsampled by 3. Fig. 2 shows the corresponding set of analysis filters. Note that due to the choice of the passbands the subband signals of $x(k)$ and $d(k)$ do not contain any aliasing components and therefore no crossterm adaptive filters between adjacent subbands are necessary.

At the moment typical designs [4] of such filterbanks yield a reconstruction error and "inband" alias levels

Filterbank	$O(N)$ Algorithm	$O(N^2)$ Algorithm	OSR
Fullband	$100\% \cdot l_e$	$100\% \cdot l_e^2$	100%
2-3-2	$61\% \cdot l_e$	$28.7\% \cdot l_e^2$	133%
2-7-2	$52\% \cdot l_e$	$25.3\% \cdot l_e^2$	114%
9 Channel	$37.7\% \cdot l_e$	$8.4\% \cdot l_e^2$	178%

Table 1: Computational Complexity of Proposed Structure

of around -100 dB. The set of analysis filters as shown in Fig. 2 result in a reconstruction error of -125 dB, a delay of 151 samples and inband alias levels between -102 dB and -115 dB.

3. COMPUTATIONAL COMPLEXITY

Tab. 1 shows the computational complexity and oversampling ratio (OSR) of the proposed structure for various oversampled filterbanks and for different adaptive algorithms with computational complexities of $O(N)$ (e.g. LMS/NLMS) and $O(N^2)$ (e.g. RLS). The computational complexity is given as a ratio of the complexity between a fullband implementation and a subband implementation as functions of equivalent fullband lengths l_e. It is assumed that the calculations necessary in the filterbanks are much less than the calculations necessary for the adaptive filters and can therefore be omitted in the evaluation.

The filterbanks, which are compared, are a 2-3-2 filterbank [4], where two channels are subsampled by 2 and one by 3, which is shown in Fig. 2, a 2-7-2 filterbank, where two channels are subsampled by 2 and one by 7 and a filterbank consisting of nine channels with different subsampling ratios [4].

4. SIMULATIONS

Floating point computer simulations were carried out using the proposed structure with least squares (LS) algorithms in a system identification setup where the unknown system was an IIR model of an acoustic transfer function with an impulse response of a duration of about 200 taps. The other algorithms used were a fullband algorithm and a 2 band structure as proposed in [3] which needs adaptive crossfilters. The filterbank used for the 2 band structure was a QMF bank using the 32C filter from [1].

4.1. Same Computational Complexity

In this set of simulations, the three algorithms were compared while maintaining the same computational

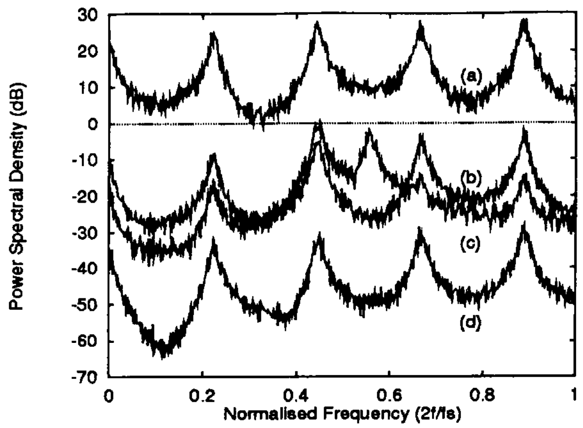

Figure 3: Residual Error for Same Computational Complexity: (a) Desired Signal $d(k)$, (b) Fullband QR Algorithm, (c) 2 Band Structure with Crossterms [3], (d) Proposed Structure

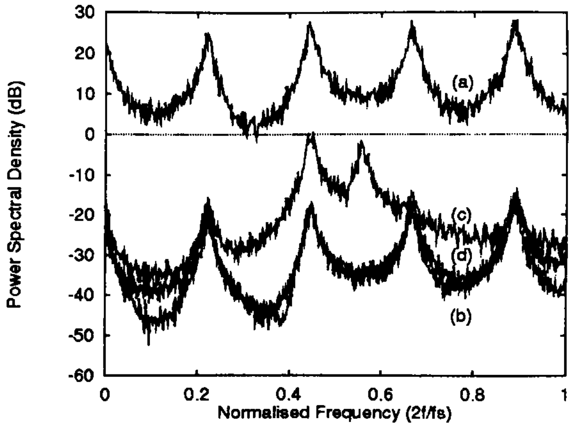

Figure 4: Residual Error for Same Equivalent Fullband Length: (a) Desired Signal $d(k)$, (b) Fullband QR Algorithm, (c) 2 Band Structure with Crossterms [3], (d) Proposed Structure

complexity for each structure using LS adaptive algorithms. The fullband adaptive filter had a length of 140 taps, the 2 band structure had 100 taps in the main adaptive filters and 40 taps in the cross adaptive filters and the proposed structure had adaptive filters with 130 taps in the channels subsampled by 2 and 87 taps in the channel subsampled by 3.

It can be seen that the proposed structure (d) outperforms the fullband (b) and the 2 band structure (c). This however was expected as the equivalent fullband length l_e is 140 taps for the fullband structure, 200 taps for the 2 band structure and 260 taps for the proposed structure and therefore a much better model could be found. Note the additional spectral peak and the degraded performance in the error signal of the 2 band adaptive filter (c), which is caused by the aliasing produced by the filterbank.

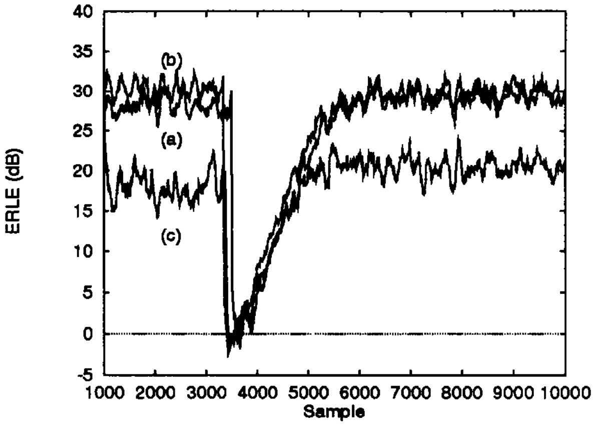

Figure 5: Simulation Results with System Change after 3333 samples: (a) Proposed Structure, (b) Fullband QR Algorithm, (c) 2 Band Structure with Crossterms

4.2. Same Equivalent Fullband Length

In Fig. 4 the same simulation was performed, however the equivalent fullband length was kept the same for all structures. The fullband adaptive filter had 200 taps, the 2 band structure 100 taps in its main adaptive filters and 40 taps in its cross adaptive filters and the proposed structure had adaptive filters with 100 taps in channel 0 and channel 2 and 66 taps in channel 1.

It can be seen that the fullband structure and the proposed structure give the same performance whereas the 2 band structure performs worse which is due to the aliasing encountered in the subbands. Here the near-perfect reconstruction property of the filterbank is quite important as it would show a higher minimum mean squared error in the simulation.

4.3. Time-Varying System

Fig. 5 shows echo-return-loss-enhancement (ERLE) plots against time. ERLE is defined as the ratio of the power of the echo signal over the power of the residual signal and is a measure of the echo suppression of the algorithm. The ERLE is estimated over a time window of 100 samples. After 3333 samples the unknown system is changed abruptly to a different IIR model which yields an impulse response of about 150 taps. The parameters for the adaptive structures were set such that the same equivalent fullband length l_e of 100 taps was achieved and that the same memory lengths for the LS estimation were obtained.

Echo Level	2 3 2		Fullband	
	SNR	ERLE	SNR	ERLE
14.12 dB	13.3 dB	24.7 dB	3.9 dB	23.0 dB
0.1 dB	15.9 dB	16.1 dB	9.9 dB	10.0 dB
-5.8 dB	16.1 dB	10.0 dB	9.9 dB	4.0 dB
-9.4 dB	16.1 dB	6.7 dB	9.9 dB	0.5 dB
-11.9 dB	16.1 dB	4.2 dB	9.9 dB	-1.9 dB

Table 2: Noisy System Identification results

It can be seen that the fullband (b) and the proposed structure (a) behave the same in terms of achievable ERLE and convergence speed. The only difference is the delay caused by the filterbank. The 2 band structure again suffers under the influence of the "inband" aliasing and therefore gives a worse ERLE level when adapted. The convergence speed is unaffected from this effect.

4.4. Simulations with Observation Noise

The next set of simulations was done to investigate the robustness towards noise and the achievable ERLE and SNR when compared to a fullband architecture. Tab. 2 shows the results of this set of simulations for different levels of echo where echo is defined as the ratio of power in the output from the unknown system to observation noise power. For the calculation of the SNR and the ERLE it was assumed that the observation noise is the signal of interest as it would be in an acoustic echo cancelation setup.

It can be seen that generally the proposed structure outperforms a fullband implementation in terms of SNR and ERLE. The achieved improvement in ERLE is about 7 dB and in SNR is about 6 dB. The improvements are due to the reduced order of the adaptive filters and to the small reconstruction error of the used three channel filterbank.

5. CONCLUSIONS

In this paper we have investigated the applicability of the recently proposed filterbank [4] for subband adaptive filtering. The main feature of this filterbank is that the different channels operate on different subsampling ratios and thereby a filterbank design can be achieved which reduces the "inband" aliasing significantly.

It has been shown that the proposed structure yields a lower computational complexity then a fullband implementation or traditional critically subsampled filterbanks which is due to the fact that lower order adaptive filters can be used at a lower sampling frequency.

In a noise-free environment, the proposed structure matches the minimum mean squared error of a fullband implementation. In a noisy environment however, it outperforms a fullband implementation which is due to the lower filter order of the adaptive filters.

6. ACKNOWLEDGMENTS

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