3rd European Conference on
Speech Communication and Technology
EUROSPEECH'93
Berlin, Germany, September 19-23, 1993

POLYPHASE ALLPASS IIR STRUCTURES FOR SUB-BAND ACOUSTIC ECHO CANCELLATION

J. E. Hart, P. A. Naylor and O. Tanrikulu

Department of Electrical and Electronic Engineering, Imperial College, London, UK.

10.21437/Eurospeech.1993-414

ABSTRACT

The advantages and current limitations of sub-band approaches to echo cancellation are reviewed. Polyphase allpass IIR half-band decimators and interpolators are presented. Their performance is compared to QMF structures in NLMS sub-band echo cancellation for hands-free telephone signals recorded in a car. The polyphase allpass IIR case is shown to give around 2 dB more ERLE with one fifth of the number of multiplies compared to direct form QMF.

1. INTRODUCTION

The use of adaptive filters in sub-band structures is particularly interesting for applications of acoustic echo cancellation in hands-free terminals such as telephones or video-conferencing systems. In such applications the echo path impulse response can be long thus requiring long adaptive filters to perform the cancellation. Furthermore, the input signal to the system is speech which has a large eigenvalue spread. Both these two factors tend to slow down the convergence of adaptive algorithms such as those based on LMS. However, by treating the signal in a number of sub-bands both the eigenvalue spread and the filter length in each sub-band can be significantly less than in an equivalent full-band case. This leads potentially to faster convergence.

Typically, a signal is divided into sub-bands using QMF filters [1][6]. In FIR implementations, such filters need to be relatively long in order to provide accurate band division without introducing significant aliasing [2]. The introduction of long filters to perform the sub-band decimation increases the computational overhead and, hence, loses some of the initial benefit of the sub-band approach. Furthermore, long FIR filters introduce delays which may be problematic in communication systems.

In general, decimation (or interpolation) requires two operations, lowpass filtering and sub-sampling (or zero padding). In "classical" schemes, the lowpass filtering is performed at the higher of the two rates, i.e. before sub-sampling in decimation and after zero padding in interpolation. Efficient implementations exist [1] in which the filter's multiplications are performed at the lower of the two rates. One of the objectives of the work described here is to reduce the computational overhead arising from the conversion of a full-band signal to a sub-band signal so as to allow more of the potential computational efficiency of sub-band methods to be exploited. In the decimation and interpolation approaches presented, this objective is achieved here as a consequence of three factors, (i) the lowpass filtering processes are performed at the lower of the two sampling rates as in other efficient implementations, (ii) the filters themselves are IIR and hence computationally cheaper and (iii) the same computations are exploited for both highpass and lowpass sections of complimentary half-band filters.

It has been found by several workers, for example [2], that imperfect sub-band decomposition can degrade the performance of adaptive filters used in sub-band echo cancellation schemes. Such imperfections arise in practice since filters used in practical decimation and interpolation are non-ideal in terms of transition-band width and stop-band attenuation. The consequence of the non-ideal filtering is aliasing of the sub-band signals which gives rise to an increased level of misadjustment of the adaptive filters. The use of cross-adaptive filters between bands has been proposed [5] for the removal of cross-terms caused by aliasing. This approach, however, leads to an increased computational complexity. Non-critical sampling of the frequency spectrum is an alternative approach of current interest. A second objective of this work is to investigate the benefit to sub-band acoustic echo cancellation obtained from the relatively high quality filtering, in magnitude response terms, offered by polyphase allpass IIR structures.

2. DECIMATION AND INTERPOLATION

The polyphase decimator structure used in this work is shown in Figure 1. This structure performs a half-band decomposition on the signal $X(z)$ into two half-rate signals, the lower band signal $Y_0(z^2)$ and the upper band signal $Y_1(z^2)$.

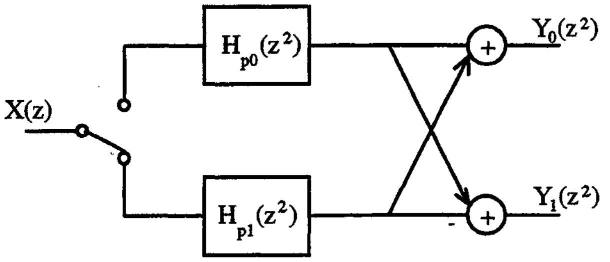

Figure 1. Polyphase Allpass Half-band Decimator

The filters $H_{p0}(z^2)$ and $H_{p1}(z^2)$ are both allpass filters implemented as the cascade of first-order allpass sections such that

$$H_p(z) = \prod_{i=0}^{N_p-1} \frac{\alpha_{p_i} + z^{-1}}{1 + \alpha_{p_i} z^{-1}} \qquad (1)$$

where α_{p_i} is the i^{th} filter coefficient of the p^{th} phase (p = p0, p1, p2, p3 ...) and N_p is the order of the filter in the p^{th} phase. The complementary interpolator is shown in Figure 2.

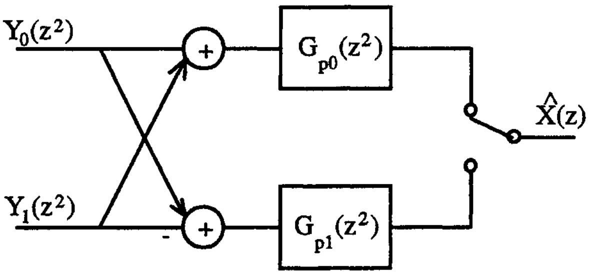

Figure 2. Polyphase Allpass Half-band Interpolator

The transfer functions of the structures can be written as

$$H_0(z^2) = \frac{Y_0(z^2)}{X(z^2)} = H_{p0}(z^2)z^{-1} + H_{p1}(z^2) \qquad (2)$$

$$H_1(z^2) = \frac{Y_1(z^2)}{X(z^2)} = H_{p0}(z^2)z^{-1} - H_{p1}(z^2) \qquad (3)$$

$$\hat{X}(z) = G_{p0}(z^2)\left(Y_0(z^2) + Y_1(z^2)\right) + \qquad (4)$$

$$G_{p1}(z^2)z^{-1}\left(Y_0(z^2) - Y_1(z^2)\right)$$

Combining (2), (3) and (4) and using

$$G_{p0}(z) = 0.5H_{p1}(z) \text{ and } G_{p1}(z) = 0.5H_{p0}(z) \quad (5)$$

the transfer function of the decimator-interpolator is given by

$$H(z) = \frac{\hat{X}(z)}{X(z)} = 2z^{-1}H_{p0}(z^2)H_{p1}(z^2) \qquad (6)$$

Thus it can be seen that when $H_{p0}(z)$ and $H_{p1}(z)$ are allpass,

$$|H(\omega)| = 2 \qquad (7a)$$

$$\angle H(\omega) = -\omega + \angle H_{p0}(2\omega) + \angle H_{p1}(2\omega) \qquad (7b)$$

Equations (7a) and (7b) imply that perfect reconstruction can be obtained in the magnitude spectrum since $|H_{p0}(z)| = |H_{p1}(z)| = 1$, but that phase distortion is introduced since $H_{p0}(z)$ and $H_{p1}(z)$ are not linear phase filters. However, they have approximately linear phase except near their transition-band as shown in Figure 4b.

3. SIMULATIONS

The polyphase, allpass IIR filters employed in the following simulations were of orders 6 and 5 for $H_{p0}(z)$ and $H_{p1}(z)$ respectively. The coefficients α of these half-band filters were obtained using an iterative design procedure [3] and are tabulated below. The coefficient values for the interpolator were trivially obtained from (5).

	α_1	α_2	α_3	α_4	α_5	α_6
Hp0	0.984964	0.798278	0.914901	0.593341	0.297311	0.040409
Hp1	0.865132	0.953132	0.708912	0.452729	0.149350	

Table 1. Polyphase, allpass IIR filter coefficients.

For the purposes of comparison, a 32 tap equi-ripple QMF filter (designed using the Parks-McClellan procedure) was also implemented in direct form. The magnitude responses for the QMF and the polyphase allpass IIR structure are given in Figure 3 for the lowpass case. It can be seen that the polyphase allpass IIR structure offers very high quality filtering in magnitude response terms.

The level of distortion in the reconstructed signal is of great importance in multi-rate sub-band systems. For both the polyphase allpass and the QMF filters, the impulse response of cascaded decimator-interpolator sections was computed and the equivalent transfer function calculated in terms of magnitude and phase responses. These responses are given in Figures 4a and 4b. It can be seen that the FIR realisation suffers from the expected in-band ripple and loss around the half-band frequency (~14 dB in this case). The response for the polyphase allpass case almost exactly verifies the statement of perfect reconstruction made in equation (7a). As expected the QMF case gives a linear phase response. The phase reconstruction of the polyphase allpass case is non-linear near the half-band frequency. However, in applications such as echo cancellation of speech signals it can be argued that mild phase distortion does not severely degrade the perceived performance of the system. Furthermore, it can be seen that the delay in the polyphase allpass case is smaller.

To evaluate the polyphase allpass structures in an application, a sub-band acoustic echo canceller was implemented. The echo cancellation performance using the polyphase allpass decimators and interpolators described above was compared to the performance obtained using the QMF filters. Figure 5 shows a sub-band echo canceller for a 2 band case.

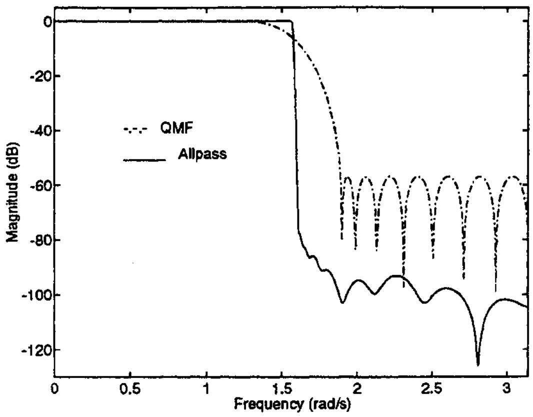

Figure 3. Magnitude Response of the Polyphase Allpass Half-band Filter Compared to 32-tap QMF.

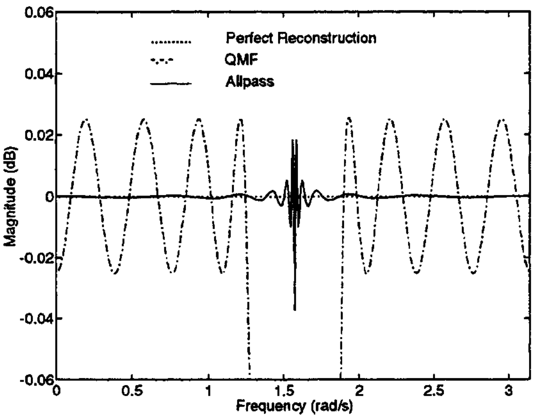

Figure 4a. Magnitude Response of Cascaded Decimator-Interpolator.

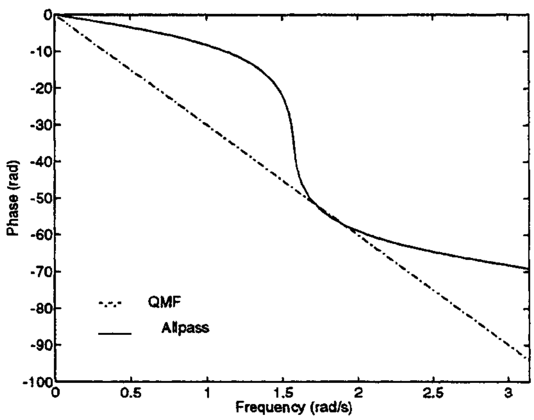

Figure 4b. Phase Response of Cascaded Decimator-Interpolator

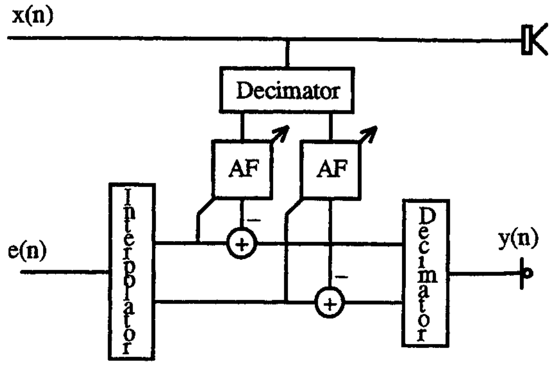

Figure 5. Two Band Adaptive Echo Canceller

Here, the two blocks marked AF are standard NLMS adaptive filters [4] having a step-size of 0.1 and each one having an impulse response length equal to half the length of the equivalent full-band case. In this work we have considered 2, 4 and 8 band systems where the 4 and 8 band systems are simple extensions of the system in Figure 5.

The signals x(n) and y(n) were recordings of USASI noise (a stationary signal with noise-like spectrum) made for a hands-free acoustic front-end in a stationary Renault 25 car. The front-end was of the integrated loudspeaker and microphone type and is a current hands-free telephone product of Matra (France). An equivalent full-band impulse response length of 256 taps was used and the sampling frequency was 8 kHz.

Three different decimation/interpolation schemes were focused on, (a) 4 band QMF, (b) 4 band polyphase allpass IIR and (c) full-band. The 4 equal bands were obtained by sub-dividing each of 2 half-band signals as a binary tree. In all schemes the adaptive filtering used was identical. The ERLE was computed during the convergence and is plotted in Figures 6a, 6b and 6c for cases (a), (b) and (c) respectively. It should be noted that the choice of 0.1 for the step-size was made arbitrarily and does not represent the optimal case for convergence speed. It does, however, give a true comparison of the convergence of the 3 methods. Finally, Figure 7 shows the ERLE after convergence for the full- and sub-band cancellers for 2, 4 and 8 equal bands.

4. COMPUTATIONAL COMPLEXITY

The number of multiply operations per sample required to perform the decomposition of a signal into 4 sub-bands using the QMF and allpass polyphase IIR filters is as follows.

QMF:	POLYPHASE IIR:
2 x 32 + 4 x 16 = 128	11 + 2 x 5.5 = 22

It should be noted that the direct form implementation has been used for the QMF filters in this work. The complexity of the QMF case can be reduced by a factor of 2 or more by using efficient implementations such as described in [1].

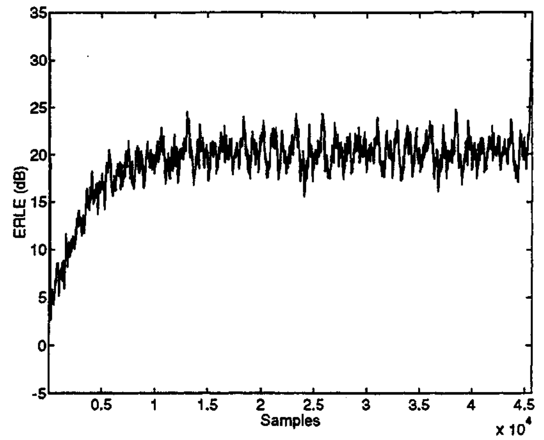

Figure 6a. Convergence of the 4 Band QMF Based Echo Canceller

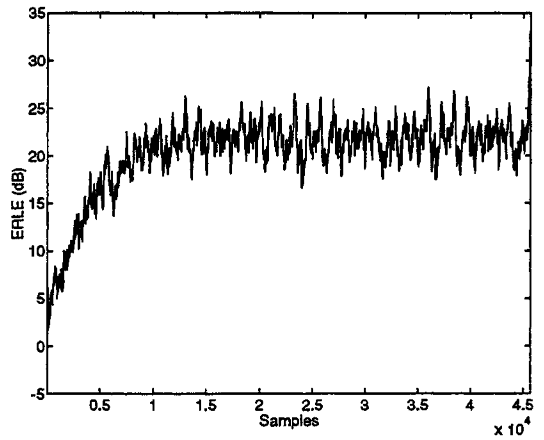

Figure 6b. Convergence of the 4 Band Polyphase Allpass IIR Echo Canceller.

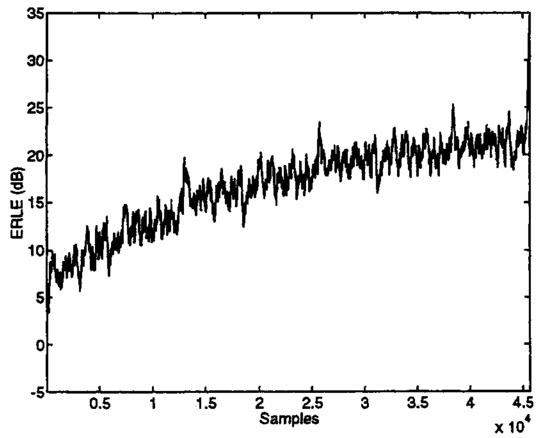

Figure 6c. Convergence of the Full-Band Echo Canceller

This work was funded in part by the CEC (ESPRIT Project 6166 "FREETEL") and the UK Science and Engineering Research Council (Studentship Award).

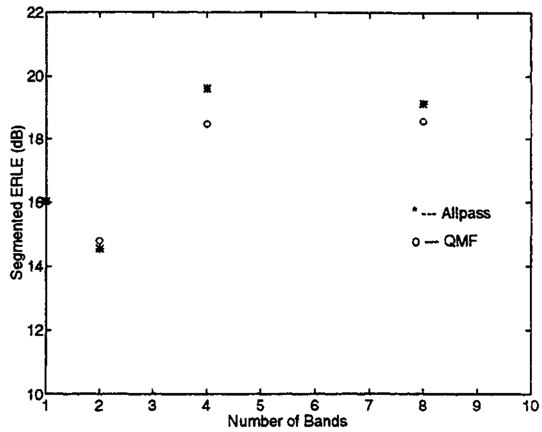

Figure 7. Comparison of ERLE After Convergence

6. CONCLUSIONS

It has been seen that sub-band approaches to acoustic echo cancellation have significant potential advantages over full-band methods. However, two factors that can limit the performance of sub-band cancellers have be highlighted - the computational cost of the decimation/interpolation and the aliasing between bands which arises from the non-ideal characteristics of the filters employed. The use of polyphase allpass IIR decimator and interpolator structures has been proposed and investigated. It has been shown that these filters have very narrow transition bands, very high stop-band attenuation and very low computational cost compared to QMF-type filters. Tests of sub-band acoustic echo cancellers using polyphase allpass IIR structures have shown convergence speed and ERLE performance as good as, or better than, QMF-based cancellers with a significantly smaller computation cost.

7. REFERENCES

[1] Proakis, J.G., Rader, C.M., Fuyun Ling, Nikias, C.L., *Advanced Digital Signal Processing*, Macmillan, 1992.

[2] Vetterli, M., "A Theory of Multirate Filter Banks", IEEE Trans. on Acoust., Speech and Sig. Proc., Vol. ASSP-35, No. 3, 1987.

[3] Harris, F., d'Oreye de Lantremange, M., Constantinides, A.G., "Design and Implementation of Efficient Resampling Filters using Polyphase recursive Allpass Filters", 25th Asilomar Conf. on Signals, Systems and Computers, pp. 1031-6, Vol. 2.

[4] Haykin, S., *Adaptive Filter Theory*, Prentice Hall, 1991.

[5] Gilloire, A., and Vetterli, M., "Adaptive Filtering in Sub-bands", Proc. ICASSP, pp. 1572-5, 1988.

[6] Chu, P.L., "Quadrature Mirror Filter Design for an Arbitrary Number of Equal Bandwidth Channels", IEEE Trans. on Acoust., Speech and Sig. Proc., Vol. ASSP-33, No. 1, 1985.