FSF (FREQUENCY SAMPLING FILTER) BANK FOR ADAPTIVE SYSTEM IDENTIFICATION

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ABSTRACT

Some researchers have explored the use of filter banks to improve the performance of the adaptive algorithms. However, some of these approaches have shown degradation in performance. In this paper, we define a new class of filter bank (ideal OFB) which enables us to avoid the degradation, even if we choose critical subsampling in adaptive filtering. Next, we propose a filter bank (pseudo OFB) whose frequency response is the same with the ideal OFB at discrete frequency points. As a result, using the proposed filter bank, we can identify an unknown system at the discrete frequency points.

I. INTRODUCTION

When a adaptive filter is a very long filter, the computational complexity is very large and the performance of the algorithm is unsatisfactory in convergence speed, computational noise, etc. Thus, some researchers have recently explored the use of multirate technique in such applications, by dividing the input and desired signal into subbands[1]-[6]. Using these approaches, the problem of identifying a long filter can be changed to one of identifying several smaller filters in parallel at a lower rate.

However some of these have shown degradation in performance, due to the introduction of either aliasing or spectral gaps in the output[1]-[4]. To avoid this problem, Somayazulu et al. have proposed a new adaptive structure with auxiliary subbands[6][7]. In this structure, we cannot use critical subsampled banks, namely maximally decimated banks, which may achieve the lowest computation complexity. Also, Gilloire et al. have shown that an identified system with factorized cross-terms is effective in solving the problem of the non-cancellation of aliasing terms that appear, when using critical subsampling[5]. However, due to the necessity of the cross-term adaptive filters, this method cannot reduce the number of multiplications per output sample, compared to the structure proposed in [6].

Our purpose in this paper is to propose a new filter bank, which enables us to use an identified system without cross-terms, even if we choose critical subsampling in adaptive filtering. we propose a new class of filter bank and then show that the filter bank is effective in avoiding the degradation. Its performance is compared to those of the conventional QMF bank for adaptive system identification.

II. A NEW CLASS OF FILTER BANK

We will define a new class of filter bank for adaptive system identification. In order to simplify the expression, we assume a two-channel filter bank. Fig. 1 shows an adaptive identification system with filter banks. In this case, we obtain the input-to-output relationship of this structure in the form

$$Y(z)=1/2\{H_0(z)A_0(z^2)F_0(z)+H_1(z)A_1(z^2)F_1(z)\}X(z)$$
$$+1/2\{H_0(-z)A_0(z^2)F_0(z)+H_1(-z)A_1(z^2)F_1(z)\}X(-z). \quad (1)$$

where $A_0(z)$ and $A_1(z)$ are transfer functions of adaptive filters, or identified filters. In general, (1) does not correspond to a linear time invariant system, because the second term of the right side in (1), due to aliasing, cannot be cancelled except for the case of $A_0(z^2)=A_1(z^2)$ namely $U(z)=U(-z)$, even if the filter bank is perfect, that is,

$$H_0(z)F_1(z)+H_1(z)F_0(z)=2Az^{-L} \quad (2)$$
$$H_0(-z)F_0(z)+H_1(-z)F_1(z) = 0 \quad (3)$$

where L is an integer and A is an arbitrary constant. To identify the unknown system U(z), the system shown in Fig.1 must correspond to a linear time invariant system, that is, the aliasing component in (1) must be cancelled. Thus we need a filter bank which meets the following conditions:

$$H_0(z)F_0(z)+H_1(z)F_1(z)=2Az^{-L} \quad (4)$$

$$H_0(-z)F_0(z)=0 \text{ and } H_1(-z)F_1(z)=0 \quad (5)$$

Then we can see that the aliasing component in (1) is exactly canceled and thus the system reduces to a linear time invariant system. Also note that conditions (4) and (5) are stricter than (2) and (3). We term such a filter bank satisfying (4) and (5) "ideal orthogonal filter bank "(ideal OFB), in the sense that each channel signal is independently reconstructed. When using the ideal OFB, we can identify the unknown system U(z) as

$$U(z)=\{H_0(z)A_0(z^2)F_0(z)+H_1(z)A_1(z^2)F_1(z)\}/Az^{-L} \quad (6)$$

For example, the conditions for the ideal OFB, (4) and (5) can be met if

$$F_0(z)=H_1(-z),\ F_1(z)=-H_0(-z) \quad (7)$$

$$H_0(z)=H_1(-z)=H(z) \quad (8)$$

where H(z) has a linear phase and is

$$|H(e^{j\omega})| = |H_{iH}(e^{j\omega})| = \begin{cases} 1 & 0 \leqq \omega < \pi/2 \\ 0 & \pi/2 \leqq \omega < \pi \end{cases} \quad (9)$$

$H_{iH}(e^{j\omega})$ in (8) is referred to as an ideal half-band filter. However, we cannot realize the ideal half-band filter.

III. PSEUDO ORTHOGONAL FILTER BANK

Now, we will show that it is possible to identify the unknown system U(z) at discrete frequency points ω_k, without the ideal half band filter. To achieve this purpose, we must search the filters which satisfy (4) and (5) at discrete frequency points ω_k as

$$H_0(e^{j\omega_k})F_1(e^{j\omega_k})+H_1(e^{j\omega_k})F_0(e^{j\omega_k})=2Ae^{-j\omega_k L} \quad (10)$$

$$H_0(-e^{j\omega_k})F_0(e^{j\omega_k})=0 \text{ and } H_1(-e^{j\omega_k})F_1(e^{j\omega_k})=0 \quad (11)$$

When a filter bank meets (10) and (11) at a number of discrete frequency points ω_k and does not meet (4) and (5), we call it "pseudo orthogonal filter bank "(pseudo OFB). Using the pseudo OFB, we could identify the unknown system U(z) at the points ω_k.

Let us discuss how to choose H(z) which meets (10) and (11). For example, replacing H(z) in (8) with the linear-phase half-band filter $H_H(z)$, the filter bank meets (10) and (11). This type of filter may be designed by the Parks and McClellan algorithm, when assigning equal weighting to the passband and stopband[7]. Therefore, as long as the half-band filter $H_H(z)$ is chosen as H(z) under (7) and (8), (10) and (11) yield

$$H_H(e^{j\omega_k})H_H(e^{j\omega_k})-H_H(-e^{j\omega_k})H_H(-e^{j\omega_k})=Ae^{-j\omega_k L} \quad (12)$$

$$H_H(-e^{j\omega_k})H_H(e^{j\omega_k})=0 \text{ and } H_H(e^{j\omega_k})H_H(-e^{j\omega_k})=0 \quad (13)$$

or equivalently

$$H_{iH}(e^{j\omega_k})H_{iH}(e^{j\omega_k})-H_{iH}(-e^{j\omega_k})H_{iH}(-e^{j\omega_k})=Ae^{-j\omega_k L} \quad (14)$$

$$H_{iH}(-e^{j\omega_k})H_{iH}(e^{j\omega_k})=0 \text{ and } H_{iH}(e^{j\omega_k})H_{iH}(-e^{j\omega_k})=0 \quad (15)$$

where

$$H_{iH}(e^{j\omega_k})=H_H(e^{j\omega_k}) \quad (16)$$

Therefore, we can identify the unknown system U(z) as below.

$$U(e^{j\omega_k})=\{H_0(e^{j\omega_k})A_0(e^{j2\omega_k})F_0(e^{j\omega_k}) +H_1(e^{j\omega_k})A_1(e^{j2\omega_k})F_1(e^{j\omega_k})\}/Ae^{-jL\omega_k} \quad (17)$$

IV. FREQUENCY SAMPLING FILTER BANK

Let us consider the frequency sample filters shown in Fig.2, whose frequency responses have N equally spaced points($\omega_k=2\pi k/N$, k: integer) on the ideal half-band filters. Considering the sampling theorem in the frequency domain, the number of the sampling points N should be chosen as a larger integer than the number of impulse response of an unknown system.

Note that this type of filter is one of half-band filters and thus the filter bank composed of the filters, referred to as frequency sampling filter bank (FSF bank), becomes a pseudo OFB. Then, we can identify U(z) at equally spaced points, where U(z) is not identified only at $\omega_k=\pi/2$ in the case of Fig.2(b).

In addition to the property shown in III, the FSF bank has other important properties as follows:

(a) we can easily design it.

(b) we can expand it for M-channel filter banks.

(c) considering the number of frequency points we need for system identification, equally spaced points enable us to use the well-known theory such as the sampling theorem.

(d) we may apply the FFT to execute it

V. SIMULATION RESULTS

We will present a simulation example for the adaptive system identification. In this example, we use the unknown system U(z) having 33 taps shown in Fig. 2 and white noise with a variance of $\sigma^2=1$ as input signal. Also we use the RLS algorithm as adaptive algorithm. Let us consider a tree-strutted four-channel filter bank based on two-channel filter banks.

First, we consider the conventional filter bank composed of 32-tap filters, referred to as the QMF bank[8]. Fig.3 (a) shows the frequency response of the identified system U'(z), where each adaptive filter has 9 taps, respectively. Note that the error mainly exists in the neighbourhood of $\omega=\pi/4, \pi/2$ and $3\pi/4$, due to the fact that the error is caused by the aliasing component based on the transient band. If we choose a larger number of subbands , the influence of the aliasing will be emphasized. Thus in general, we must choose a very high order filter to obtain a good response.

Next, we apply the proposed FSF bank for identifying the U(z). Using the FSF bank with 32-tap filters, we obtain the identified system U'(z) having the response shown in Fig.3(b). We see that the FSF bank is more effective than the conventional QMF bank. Fig.(c) shows the response where the FSF bank with 64-tap filters is used. From Fig.(c) and Fig.(d), it is shown that we can almost identify the unknown system U(z), if the number of frequency sampling points N is larger than or equal to the length of the U(z).

V. CONCLUSION

We have proposed the FSF bank which enable us to identify an unknown system.

We have defined a new class of filter bank, as referred to as the ideal OFB. Next we have defined the pseudo OFB whose frequency response is the same as that of the ideal OFB at discrete points. The proposed FSF bank is a pseudo OFB. In the example, it has been shown that the FSF bank is more effective than the conventional QMF bank.

REFERENCES

[1] J.Chao, H.Perez, and S.Tsujii:"A 'Jumping Algorithm' for Adaptive Filtering", IEICE Trans. J73-A, 7, pp.1196-1206(July 1990)

[2] H.Yasukawa, S.Shimada, and I.Furukawa:"Acoustic Echo Canceller with High Quality", Proc. ICASSP 87, Dallas, pp.2125-2128(April 1987)

[3] A.Gilloire:"Experiments with Sub-band Acoustic Echo Cancellers for Teleconferencing", Proc. ICASSP 87, Dallas, pp.2141-2144(April 1987)

[4] T.Chinen, H.Honma, and K.Sagawa:"Adaptive Signal Processing Based on LMS Adaptive Filter Bank", IEICE Technical group on circuit and system CAS89-75(Oct. 1990)

[5] A.Gilloire and M.Vetterli:"Adaptive Filtering in Sub-bands", Proc. ICASSP 88, New York, pp.1572-1575(April 1988)

[6] V.S.Somayazulu, S.K.Mitra, and J.J.Shynk:"Adaptive Line Enhancement Using Multirate Techniques", Proc. ICASSP 89, Glasgow, pp.928-931(May 1989)

[7] J.H.McClellan, T.W.Parks and L.R.Rabiner:"A Computer Program for Designing Optimum FIR Linear Phase Digital Filters", IEEE Trans. Audio and Electroacoust., AU-21, 6, pp. 506-526(Dec. 1973)

[8] D.Esteban and C.Galand:"Application of Quadrature Mirror Filters to Split Band Voice Coding Schemes", Proc. of the IEEE International Conf.ASSP, Hartford, Connecticut, pp.191-195(May 1977)

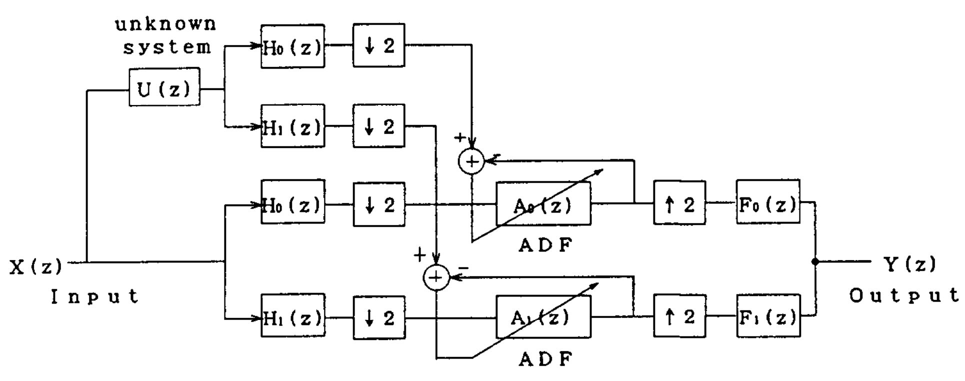
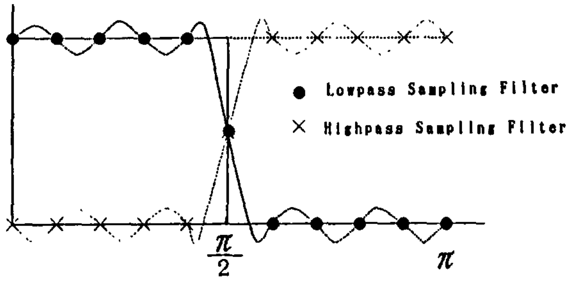
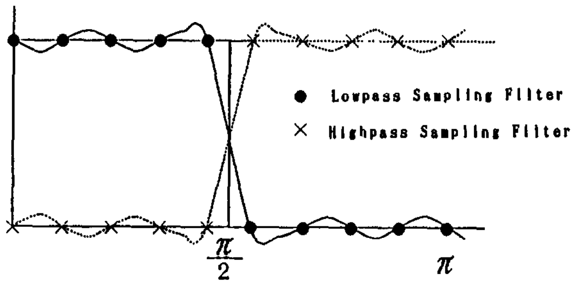


Fig. 1 Adaptive identification system with two-channel filter banks.



(a) N = 4M



(b) N ≠ 4M

Fig. 2 Frequency sampling filters.

N : number of sampling points

M : integer

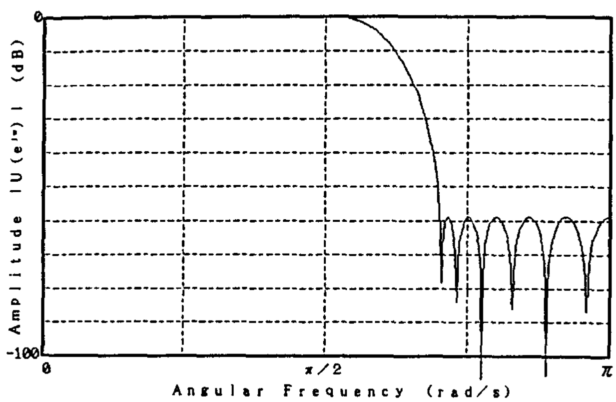
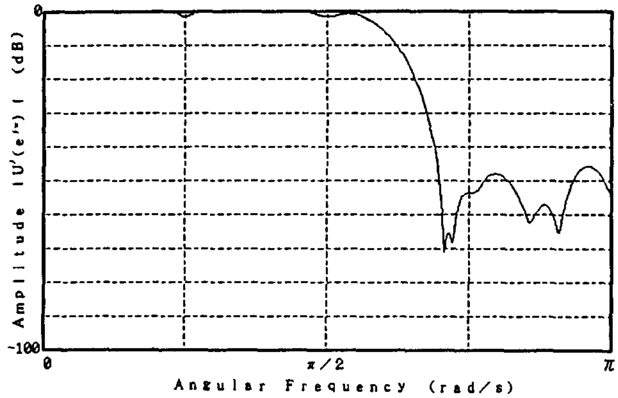
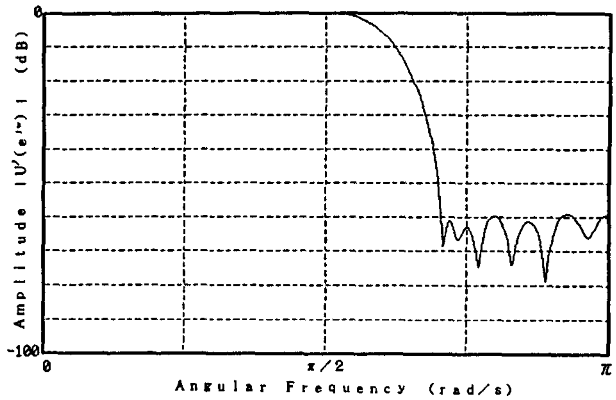


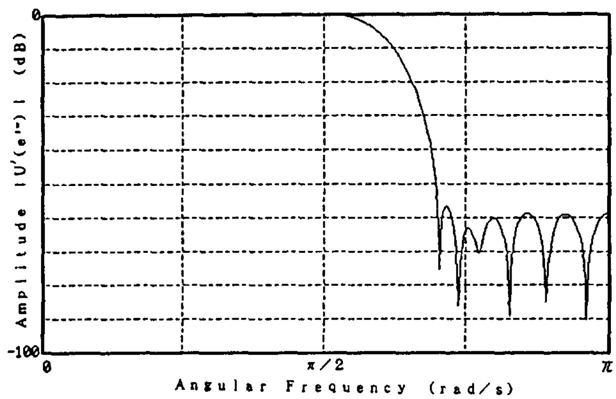
Fig. 3 The unknown system U(z) used in this example.



(a) The conventional QMF bank composed of 32-tap filters.



(b) The proposed FSF bank composed of 32-tap filters.



(c) The proposed FSF bank composed of 64-tap filters.

Fig. 4 The identified systems U'(z), where we use the adaptive filters with 9 taps and the RLS algorism.