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Automatic Tap Assignment in Sub-Band Adaptive Filter

Zhiqiang MA†, Kenji NAKAYAMA† *and* Akihiko SUGIYAMA††, *Members*

SUMMARY An automatic tap assignment method in sub-band adaptive filter is proposed in this letter. The number of taps of the adaptive filter in each band is controlled by the mean-squared error. The numbers of taps increase in the bands which have large errors, while they decrease in the bands having small errors, until residual errors in all the bands become the same. In this way, the number of taps in a band is roughly proportional to the length of the impulse response of the unknown system in this band. The convergence rate and the residual error are improved, in comparison with existing uniform tap assignment. Effectiveness of the proposed method has been confirmed through computer simulation.
key words: *sub-band, adaptive filter, filter bank, number of taps, system identification, noise canceller, echo canceller*

1. Introduction

Sub-band adaptive filters are efficient techniques to reduce computation requirement and to improve convergence rate. So far, the sub-band adaptive filters have employed uniform tap assignment,[(1)-(5)] i.e., the adaptive filter in each band has the same number of taps. However, the uniform tap assignment is not always the optimum assignment; tap assignment may depend on the characteristics of an unknown system in each band. The problem arises on how to assign suitable number of taps in each band.

In this letter, an automatic tap assignment method in sub-band adaptive filter is proposed. Effectiveness of the proposed method is confirmed through computer simulation.

2. Automatic Tap Assignment in Sub-Band Adaptive Filter

The optimum impulse response length of a sub-band adaptive filter depends on the characteristics of an unknown system. In order to obtain a small residual error and a fast convergence rate, the number of taps in a sub-band adaptive filter should be determined by the length of the impulse response of an unknown system in this band.

To describe how the proposed automatic tap assignment method works, we discuss a two bands case shown in Fig. 1. In this figure, for simplicity, U.S. denotes an unknown system. A_1 and A_2 are analysis filters which split the full-band input signal $\boldsymbol{x}(n)$ and desired response $\boldsymbol{d}(n)$ into two band signals. $\boldsymbol{x}_1(n)$, $\boldsymbol{x}_2(n)$, $\boldsymbol{d}_1(n)$, and $\boldsymbol{d}_2(n)$ denote the components of $\boldsymbol{x}(n)$ and $\boldsymbol{d}(n)$ in the low and high bands, respective-

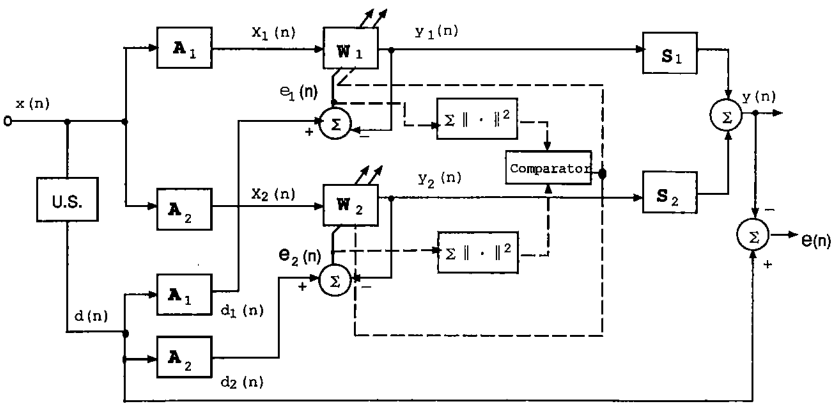

Fig. 1 Automatic tap assignment in a two-bands adaptive filter.

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† The authors are with the Faculty of Technology, Kanazawa University, Kanazawa-shi, 920 Japan.
†† The author is with the C & C Systems Research Laboratories, NEC Corporation, Kawasaki-shi, 216 Japan.

ly. In order to avoid the influence of aliasing components, over-sampling is used. W_1 and W_2 denote the tap weights in low and high bands, $y_1(n)$ and $y_2(n)$, the outputs of the adaptive filters, and $e_1(n)$ and $e_2(n)$, the residual errors. S_1 and S_2 are synthesis filters which synthesize $y_1(n)$ and $y_2(n)$ into the full-band signal $y(n)$. The difference between $d(n)$ and $y(n)$ is $e(n)$, the residual error in the full-band.

The numbers of taps in two bands are initially set to the same. The mean-squared errors (MSE) of two bands are compared. The output of the comparator controls the tap assignment. The number of taps increases in the band which has a larger error, but decreases in the other, until residual errors in two bands become the same. The number of taps is controlled one by one.

3. Simulation

Several unknown systems which have different lengths of impulse response in each band have been used in simulation, in order to confirm the effectiveness of the automatic tap assignment. One example is shown in the following.

3. 1 Unknown System

A 10th-order all poles IIR filter is used as the unknown system. Its transfer function can be expressed as

Table 1 Poles of unknown system.

p	r_p	θ_p (π)
1	0.99	0.1
2	0.95	0.3
3	0.9	0.5
4	0.8	0.6
5	0.8	0.7

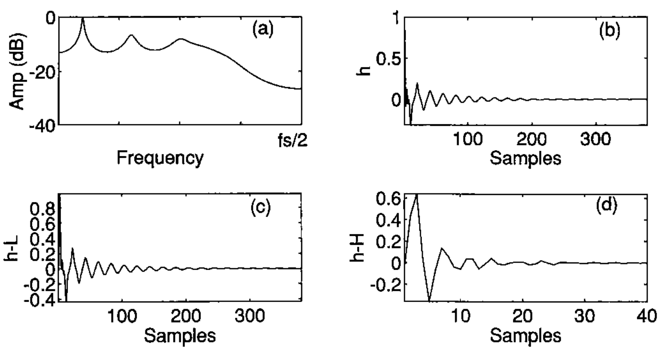

Fig. 2 Amplitude and impulse response of the unknown system.
(a) Amplitude-frequency response.
(b) Impulse response in full-band (h).
(c) Impulse response in low-band (h-L).
(d) Impulse response in high-band (h-H).

$$H(z)=K\sum_{p=1}^{5}\frac{1}{1-2r_p\cos\theta_p z^{-1}+r_p^2 z^{-2}} \quad (1)$$

where K is a scaling factor. The values of r_p and θ_p are shown in Table 1. The amplitude and impulse response h of the unknown system are shown in Fig. 2. h-L and h-H denote the impulse responses in low and high bands, respectively. Figure 2 shows that the length ratio of h-L and h-H is in the order of 10.

3. 2 Analysis and Synthesis Filter Banks

Polyphase structures are used in analysis and synthesis filter banks.[6] Prototype filter is a 41-tap quadrature mirror filter (QMF). Amplitude-frequency response of the prototype filter $|H(e^{j\omega})|$ is shown in Fig. 3(a). Reconstruction error of this analysis/synthesis system is less than ± 0.11 dB as shown in Fig. 3(b). The amplitude-frequency responses of A_1 and A_2 are $|H(e^{j\omega})|$ and $|H(e^{j\omega+\pi})|$, respectively. The amplitude-frequency responses of S_1 and S_2 are the same as A_1 and A_2, respectively.

3. 3 Automatic Tap Assignment

In the simulation, the input signal is white noise and the normalized LMS algorithm[7] is used in the adaptive filter.

$$e(n)=d(n)-W^H x(n) \quad (2)$$

$$W(n+1)=W(n)+\frac{\alpha}{\varepsilon+|x(n)|^2}x(n)e^H(n) \quad (3)$$

where $[\cdot]^H$ denotes the Hermitian transposition, $\alpha=0.05$, and $\varepsilon=10^{-10}$.

In the automatic tap assignment, 50 taps are initially assigned to each band. During adaptive process, the mean-squared errors of two bands are calculated and compared. The output of the comparator controls the tap assignment. The number of taps increases in the band which has a larger error, but decreases in the

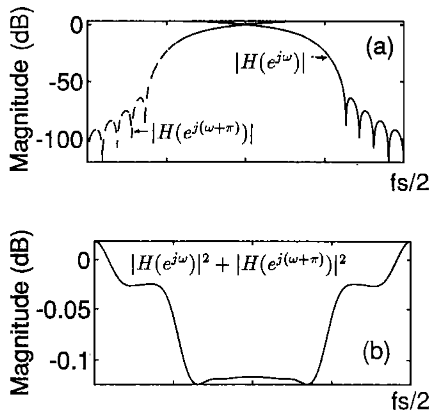

Fig. 3 (a) Amplitude-frequency response of prototype filter.
(b) Analysis/synthesis reconstruction error.

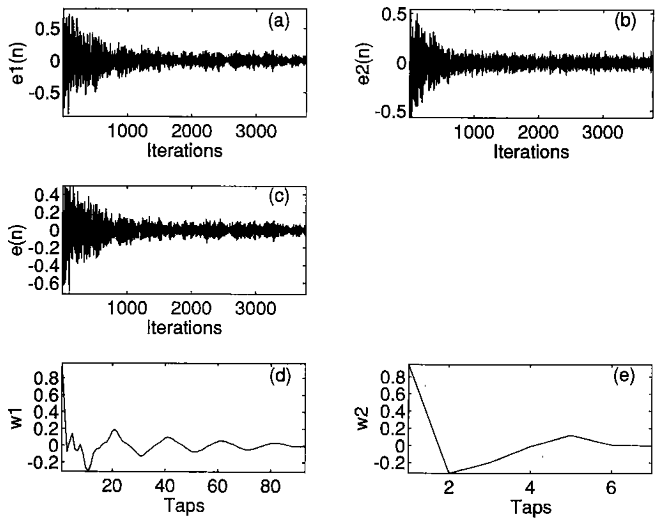

Fig. 4 Residual errors and tap weights in automatic tap assignment.
(a) Residual error of low-band adaptive filter ($e_1(n)$).
(b) Residual error of high-band adaptive filter ($e_2(n)$).
(c) Total residual error ($e(n)$).
(e) Tap weights of low-band adaptive filter (W_1).
(d) Tap weights of high-band adaptive filter (W_2).

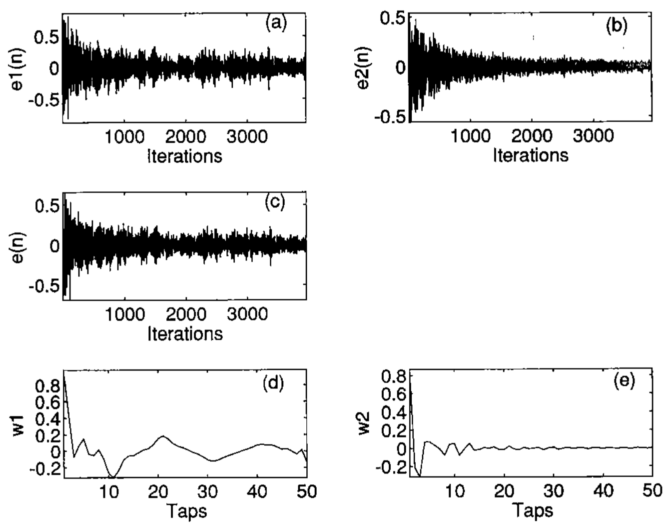

Fig. 5 Residual errors and tap weights in uniform tap assignment.
(a) Residual error of low-band adaptive filter ($e_1(n)$).
(b) Residual error of high-band adaptive filter ($e_2(n)$).
(c) Total residual error ($e(n)$).
(e) Tap weights of low-band adaptive filter (W_1).
(d) Tap weights of high-band adaptive filter (W_2).

other. The number of taps is controlled one by one. After convergence, the number of taps is 93 in the low band and 7 in the high band, respectively. The simulation results are shown in Fig. 4. The residual errors in the two bands become almost the same after convergence. The length ratio of W_1 and W_2 is also in the

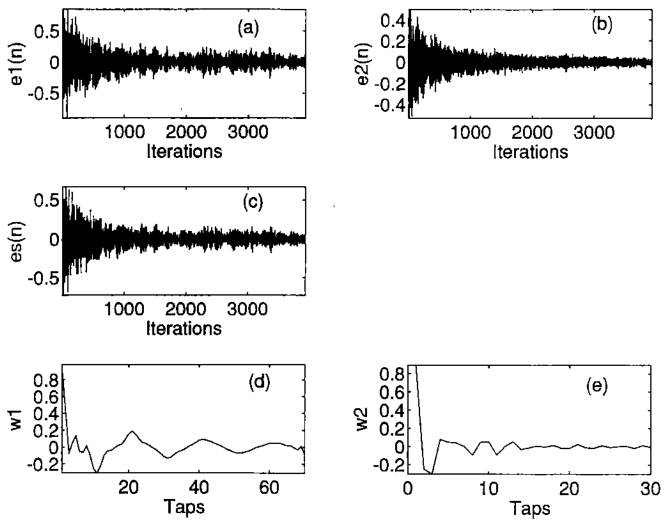

Fig. 6 Residual errors and tap weights in fixed non-uniform tap assignment, with 70 taps in low band and 30 taps in high band.
(a) Residual error of low-band adaptive filter ($e_1(n)$).
(b) Residual error of high-band adaptive filter ($e_2(n)$).
(c) Total residual error ($e(n)$).
(e) Tap weights of low-band adaptive filter (W_1).
(d) Tap weights of high-band adaptive filter (W_2).

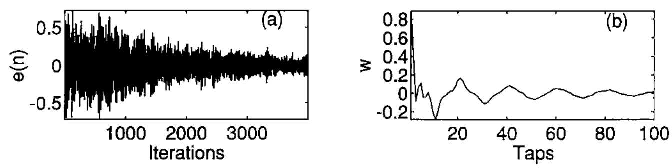

Fig. 7 Residual error and tap weights in full-band adaptive filter.
(a) Residual error ($e(n)$).
(b) Tap weights (W).

order of 10 and consistent with the length ratio of h-L and h-H.

The simulation results of uniform tap assignment, with 50 taps in each band, are shown in Fig. 5. $e_1(n)$ and $e(n)$ are larger than those in automatic tap assignment.

The simulation results of fixed and non-uniform tap assignment, for example, with 70 taps in the low band and 30 taps in the high band, are shown in Fig. 6. $e_1(n)$ and $e(n)$ are larger than those in automatic tap assignment and smaller than those in uniform tap assignment. In application, it is difficult to assign the suitable number of taps in each band for an unknown system in advance. It is also impossible to fix the suitable number of taps in each band for a time varying unknown system. These problems can be solved by the proposed automatic tap assignment method. When the lengths of the impulse responses in the bands are unequal, the convergence rate and the residual error can be improved by using the automatic tap assignment method, in comparison with the fixed uniform or non-uniform tap assignment.

The residual error and tap weights of the full-band adaptive filter, with 100 taps and $\alpha=0.05$, is shown in Fig. 7. This result is inferior to that obtained by the automatic tap assignment method.

We have also done simulations with unknown systems which have longer impulse response in the high band, or have same lengths of impulse responses in all bands. The results show that, when the lengths of impulse responses in all bands are unequal, the automatic tap assignment is evidently superior to the uniform tap assignment. When the lengths of impulse responses in all bands are equal, the same results are obtained.

4. Conclusions

An automatic tap assignment method in sub-band adaptive filter has been proposed. The number of taps in each band is controlled by the mean-squared error. So that, the number of taps in a band is almost proportional to the length of the impulse response of the unknown systems in this band. When the lengths of the impulse responses in the bands are unequal, the convergence rate and the residual error can be improved by using the automatic tap assignment, in comparison with using the uniform tap assignment.

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