A NEW ADAPTIVE INTERSUBBAND TAP-ASSIGNMENT ALGORITHM FOR SUBBAND ADAPTIVE FILTERS

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ABSTRACT

This paper proposes a new subband adaptive filtering algorithm with intersubband tap assignment. The number of taps for each subband adaptive filter is adaptively controlled based on a sum of squared coefficients to the impulse-response tail of the unknown system. The sum is weighted by the input reference signal power to reflect its power imbalance. For rapid convergence of intersubband tap assignment, a time-varying number is newly introduced as the tap-redistribution step-size. This variable is controlled based on repeated redistribution of the maximum number of taps to the same subband. For colored signals, the proposed algorithm reduces the convergence time for intersubband tap-assignment by more than 60% compared with a fixed tap-redistribution step-size. It is, insensitive to parameter-selection and has good capability to track a path change. It is also shown that the proposed algorithm is effective for real speech signals.

1. INTRODUCTION

Subband adaptive filtering is one of soulutions to increased computation and slow convergence associated with the conventional fullband adaptive filters [1, 2]. Reduced computation due to decimation in subbands is particularly effective for FIR (finite impulse response) filters with a large number of taps. Such FIR filters with increased computaion must be used when the impulse response to be modeled by the FIR filter is long. A long impulse response, sometimes as long as several thousands taps, is often encountered in acoustic echo cancellation for an office and a living room. In addition, necessary computation is also increased when the input signal has a wider bandwidth. There will be more chances to process wideband audio signals as hifi audio coding algorithms have been standardized by ISO (international standard organization) [3]. These facts suggest that the number of taps must further be reduced in some way.

For this purpose, an adaptive intersubband tap-assignment algorithm, *AITAC*, has been proposed [4]. It controls the number of taps for each subband adaptive filter based on a sum of squared coefficients weighted by the subband signal power. An optimum tap distribution among subbands is achieved under a constraint on the total number of taps determined by hardware limitation. After every predetermined number of coefficient adaptations, a constant number of taps are taken from all subbands. At the same time, a cost function to determine the number of redistributed taps to a subband is evaluated. Based on the cost function, redundant taps are redistributed to subbands where they are needed, keeping the total number of taps constant.

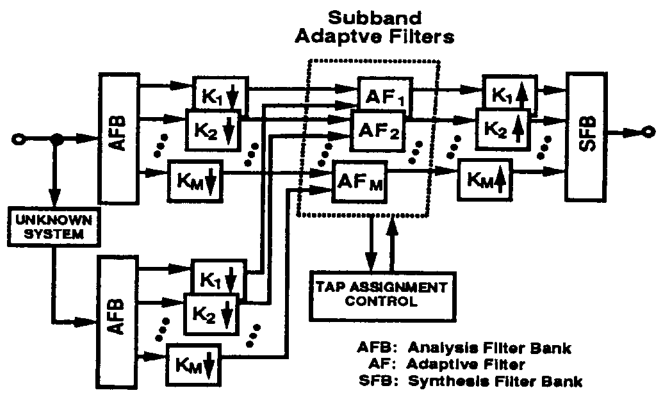

Fig. 1. Subband Adaptive Filtering.

AITAC achieves good adapation for both white and colored signals. However, it redistributes taps among subbands at a constant rate. Therefore, there is a trade-off in the selection of the tap-redistribution constant or the tap-redistribution step-size. A faster convergence of tap assignment is achieved with a large tap-redistribution step-size at the expense of small final error. This error exists both in tap assignment and in coefficient adaptation, as correct tap assignment is essential for smaller coefficient-adaptation error. On the other hand, a small tap-redistribution step-size increases the time reach to an optimum tap assignmnet.

This paper proposes a new intersubband tap assignment algorithm for subband adaptive filters. The proposed algorithm controls the tap-redistribution step-size according to repeated redistribution of the maximum number of taps to the same subband. In the following section, the conventional algorithm, *AITAC*, is reviewd. The new intersubband tap-assignment algorithm is presented in Section 3. Finally, in Section 4, the conventional and the proposed algorithms are compared based on computer simulation results for acoustic echo cancellation.

2. CONVENTIONAL ALGORITHM [4]

Typical system identification by subband adaptive filtering is illustrated in Fig. 1 for an M-band case. The input signal to the unknown system is divided into subbands by an analysis filter

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bank to serve as the input signal to each subband adaptive filter. The output signal from the unknown system is also divided and used as the desired signal in subbands. After these divisions, the subband signals are decimated by a factor of K_i (i=1,2,$\cdots$, M), which is often selected equal to M. The misadjustments or errors in each subband are interpolated and combined by a synthesis filter bank to form a full-band misadjustment. When identification is carried out perfectly, the full-band misadjustment should be zero. Intersubband tap assignment is controlled in TAP ASSIGNMENT CONTROL block based on some criterion.

Let us define some important vectors. The coefficient vector $c_{i,k}$ and the input signal vector $x_{i,k}$ in the i-th subband at the k-th iteration are determined by

$$c_{i,k} = [\, c_{i,1,k}\ c_{i,2,k} \cdots\cdots c_{i,N_{i,k},k}\,]^T, \tag{1}$$

$$x_{i,k} = [\, x_{i,k-N_{i,k}+1}\ x_{i,k-N_{i,k}+2} \cdots\cdots x_{i,k}\,]^T, \tag{2}$$

where $[\cdot]^T$ denotes vector transpose. $N_{i,k}$ stands for the number of taps in the i-th subband at the k-th coefficient-adaptation. The subband misadjustment $e_{i,k}$ is then given by

$$e_{i,k} = y_{i,k} - c_{i,k}^T x_{i,k}, \tag{3}$$

where $y_{i,k}$ is the desired signal in the i-th subband, or in other words, the unknown system output decomposed into the i-th subband.

The conventional algorithm [4] redistributes taps to subbands based on the input subband-signal power multiplied by a sum of coefficients which correspond to the impulse-response tail of the unknown system. The larger the above product of a subband, the larger the number of taps in the subband. For an M-band case, the number of taps in the i-th subband is adapted by the following equations.

$$N_{i,mS} = N_{i,(m-1)S} - R + R \cdot \Phi_{i,mS} \tag{4}$$

$$\Phi_{i,mS} = INT\left[M \times \frac{\sum_{p=(m-1)S+1}^{mS} v_{i,p}\bar{c}_{i,p}^T\bar{c}_{i,p}}{\sum_{p=(m-1)S+1}^{mS} trace\{v_p \tilde{c}_p\}} \right] \tag{5}$$

$$\tilde{c}_k = [\, \bar{c}_{1,k}^T\bar{c}_{1,k}\bar{c}_{2,k}^T\bar{c}_{2,k} \cdots \bar{c}_{M,k}^T\bar{c}_{M,k}\,] \tag{6}$$

$$\bar{c}_{i,k} = [c_{i,N_{i,k}-P+1,k}\ c_{i,N_{i,k}-P+2,k} \cdots\cdots c_{i,N_{i,k},k}]^T \tag{7}$$

$N_{i,mS}$ is the number of taps in the i-th subband at the m-th tap redistribution, where S is a predetermined nonnegative integer. This means that a new number of taps for the i-th subband is calculated at every S coefficient adaptations. $\bar{c}_{i,k}$ is a partial coefficient vector which corresponds to the last P elements of the coefficient vector $c_{i,k}$. Therefore, P coefficients from the tail are squared, summed and multiplied by the input-signal power in each subband, and then, accumulated over S iterations. The input-signal power vector v_k is given by

$$v_k = [\, v_{1,k}\ v_{2,k} \cdots\cdots v_{M,k}\,]^T, \tag{8}$$

where

$$v_{i,k} = x_{i,k}^T x_{i,k}. \tag{9}$$

R is the fixed tap-redistribution step-size which is the unit number of taps collected from each subband at a single tap-redistribution. Normalized LMS (NLMS) algorithm [5] may be employed for coefficient adaptation in each subband. It is given by

$$c_{i,k+1} = c_{i,k} + \frac{\mu}{\delta + x_{i,k}^T x_{i,k}} e_{i,k} x_{i,k}. \tag{10}$$

μ is the step size for coefficient adaptation and δ is a small positive number to avoid division by zero when the reference input signal samples are zero. Other algorithms may also be used for coefficient adaptation.

By (1) and (4) - (9), R taps are removed from each subband at a single modification of subband filter taps. The obtained MR taps are redistributed to subbands according to the value of $\Phi_{i,mS}$. Therefore, excess taps in some subbands are repeatedly redistributed to others with tap-shortage at every S iterations. Because of a constant R as the tap-redistribution step-size, there is a trade-off in tap assignment between the convergence speed and the final misadjustment. A large R speeds-up convergence with a degraded final misadjustment compared to a small R and vice versa.

3. PROPOSED ALGORITHM

The proposed algorithm adaptively controls the redistribution step-size R_{mS} to achieve fast convergence and small final misadjustment simultaneously. When the maximum number of taps are redistributed to the same subband repeatedly, R_{mS} is increased by R_0, where R_0 is a positive integer. Otherwise, R_{mS} is decreased by R_0. It should be noted that R_{mS} must be kept nonnegative.

The proposed algorithm is given by (5) - (9) and the following recursive equations for $N_{i,mS}$ and R_{mS}.

$$N_{i,mS} = N_{i,(m-1)S} - R_{(m-1)S} + R_{(m-1)S} \cdot \Phi_{i,mS} \tag{11}$$

$$R_{mS} = R_{(m-1)S} + R_0 \times \theta_{mS} \tag{12}$$

$$\theta_{mS} = \begin{cases} +1 & \text{for } \gamma_{mS} = \gamma_{(m-1)S} = \cdots = \gamma_{(m-r)S} \\ -1 & otherwise \end{cases} \tag{13}$$

γ_{mS} is the subband index i which gives the maximum value of $\Phi_{i,mS}$. Therefore, $\theta_{mS} = +1$ when the same subband keeps the maximum $\Phi_{i,mS}$ for r consecutive tap redistributions, where $r > 0$. As S taps from the tail of the filter are evaluated, (14) should naturally be satisfied.

$$1 \le R_{mS} < S \tag{14}$$

For a two-band case, the number of taps in the first and the second subbands are adjusted by the following equations which correspond to (5) and (11)-(13) for an M-band case.

$$\begin{cases} N_{1,mS} = N_{1,(m-1)S} + R_{(m-1)S} \cdot sign(\Delta\bar{c}_{mS}) \\ N_{2,mS} = N_{2,(m-1)S} - R_{(m-1)S} \cdot sign(\Delta\bar{c}_{mS}) \end{cases} \tag{15}$$

$$\Delta\bar{c}_{mS} = \sum_{p=(m-1)S+1}^{mS} (v_{1,p} \cdot \bar{c}_{1,p}^T\bar{c}_{1,p} - v_{2,p} \cdot \bar{c}_{2,p}^T\bar{c}_{2,p}) \tag{16}$$

Tab. 1. Parameter for Simulations.

Parameter	Value	Parameter	Value
r	1	R_0	1
P	32	M	2
$N_{i,0}$	750	μ	0.8

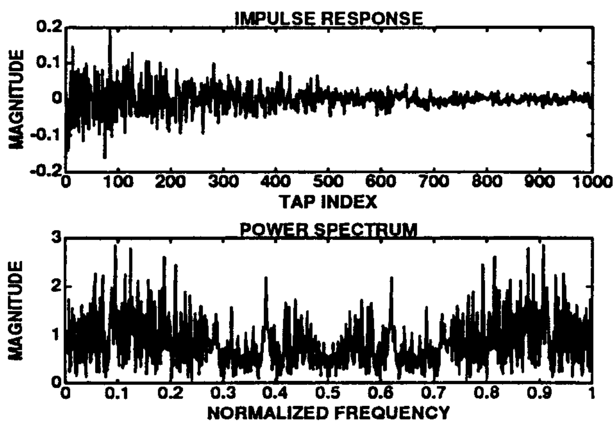

Fig. 2. Impulse Response of the Unknown System and Its Power Spectrum.

$$R_{mS} = R_{(m-1)S} + R_0 \cdot sign\left(\left\{\sum_{p=(m-r)S}^{mS} sign(\Delta\bar{c}_p)\right\}^2 - r(r+2)\right) \quad (17)$$

$sign\,(\cdot)$ is equal to the sign of a nonzero argument and zero for a zero argument. The number of taps in one of the subbands, whose power-weighted sum of the squared tail-coefficients is larger than the other, is increased by R_{mS}. R_{mS} is increased by R_0 when taps are redistributed to the same subband repeatedly, otherwise, decreased by R_0.

4. SIMULATION RESULTS

Simulations have been carried out assuming acoustic echo cancellation. White *Gaussian* signals, colored signals and a speech signal were used as the input signal. The colored signal was generated by filtering a white *Gaussian* signal with a third-order recursive filter. The transfer function A(z) of the filter is given by

$$A(z) = \frac{0.25}{1 - 1.5z^{-1} + z^{-2} - 0.25z^{-3}} \quad (18)$$

where $z = e^{j\omega}$. The number of taps for the impulse response to be identified was 1000. The impulse response and its power spectrum are depicted in Fig. 2. The total number of taps was 1500. Other simulation parameters are tabulated in Tab. 1. All the curves for simulation results except for a real speech signal are an ensemble average of fifty independent runs.

Figure 3 compares tap-assignment behavior by the proposed and the conventional algorithms for white signals. Thanks to the time-varying tap-redistribution step-size, convergence by the proposed algorithm is reduced by more than 60% compared with the algorithm with a fixed tap-redistribution step-size [4].

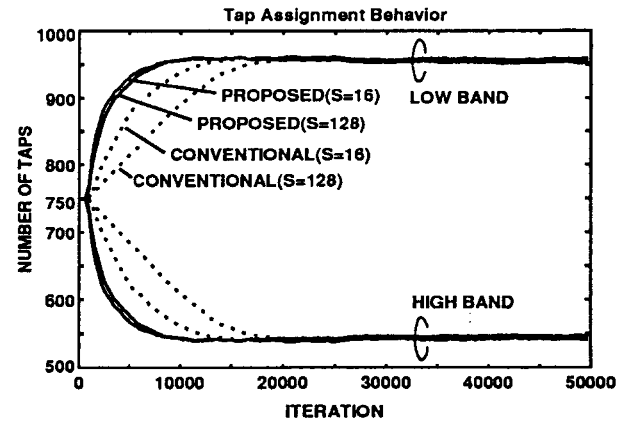

Fig. 3. Tap-Assignment Behavior (White Signal).

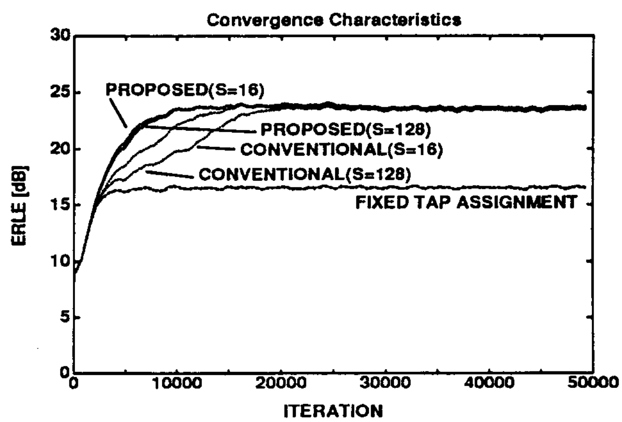

Fig. 4. Convergence Characteristics (White Signal).

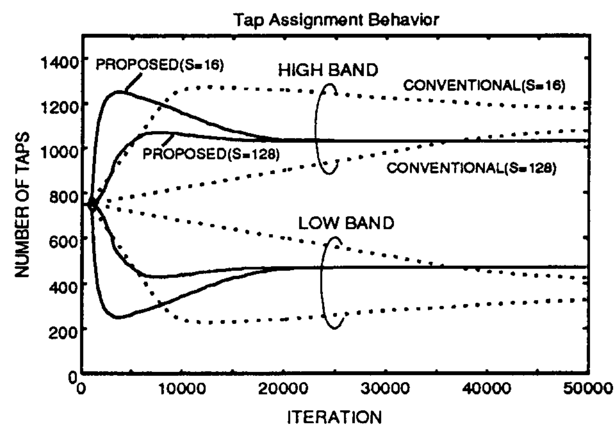

Fig. 5. Tap-Assignment Behavior (Colored Signal).

Shown in Fig. 4 are the full-band ERLE (echo return loss enhancement) curves for the conventional and the proposed algorithms as well as the classic subband algorithm with fixed tap-assignment. The proposed algorithm converges to the final full-band ERLE, which is almost 10dB better than that by the fixed tap-assignment, as much as 60% faster than [4].

Figure 5 exhibits tap-assignment behavior for colored signals. Convergence time for tap assignment by the proposed

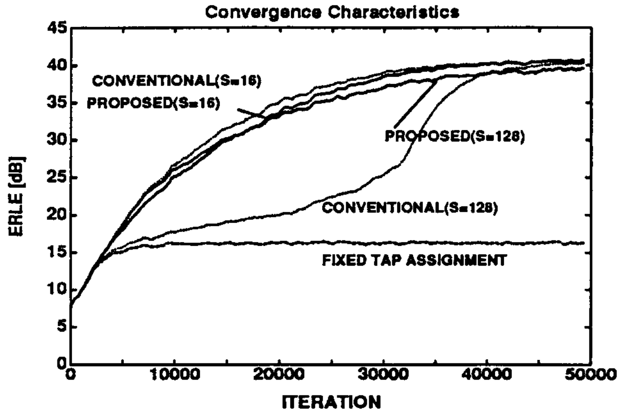

Fig. 6. Convergence Characteristics (Colored Signal).

algorithm is shorter than 40% of that by the fixed tap-redistribution step-size algorithm. Convergence of the ERLE is slightly faster as shown in Fig. 6. However, it is insensitive to selection of parameter S.

Figures 7 and 8 show tracking capability of the proposed algorithm for a path change with colored inputs. Both tap assignment in Fig. 7 and ERLE convergence in Fig. 8 show good tracking capability. Finally, in Fig. 9, tap assignment behavior for a real speech signal is depicted. The upper graph is the speech signal used to obtain the tap-assignment behavior in the lower graph. It is obvious from Fig. 9 that the proposed algorithm achieves reasonable tap assignment for a real speech signal.

5. CONCLUSION

A new adaptive intersubband tap-assignment algorithm has been proposed. A variable number of taps are collected from all subbands and redistributed based on the power-weighted sum of squared coefficients. The variable is adaptively controlled according to repeated redistribution of the maximum number of taps to the same subband. An optimum tap assignment has been reached within 40% of the time required for the conventional algorithm even for colored signals. Insensitivity to parameter selection has been shown to be an additional advantage of the proposed algorithm. Almost 25dB better final ERLE compared with the classical fixed tap-assignment algorithm is achieved for the same total number of taps. No problem has been encountered for a path change and real speech input either. With all these characteristics, the proposed algorithm is promising for wide-band acoustic echo cancellation, especially in a mid- to a large-size room.

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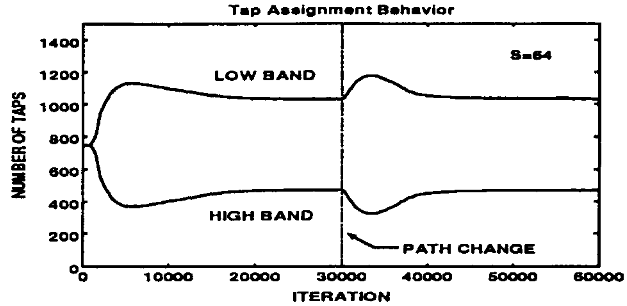

Fig. 7. Tap-Assignment Behavior for a Path Change.

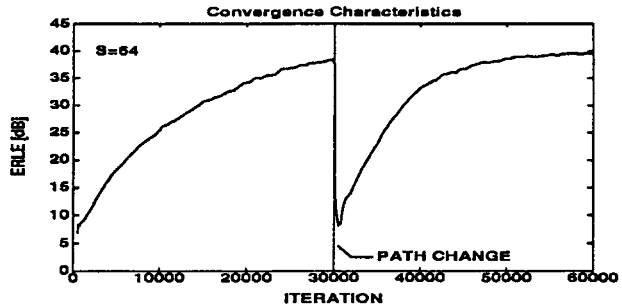

Fig. 8. Convergence Characteristics for a Path Change.

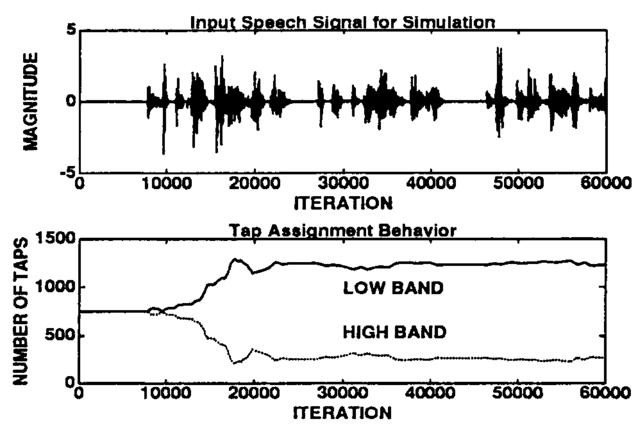

Fig. 9. Tap-Assignment Behavior for a Real Speech.

REFERENCES

[1] E. Hänsler, "The Hands-Free Telephone Problem - An Annotated Bibliography," Signal Processing, vol.27, No.3, pp.259-271, Jun. 1992.

[2] J. J. Shynk, "Frequency-Domain and Multirate Adaptive Filtering ," IEEE SP Magazine, pp.14-37, Jan. 1992.

[3] "Coding of Moving Pictures and Associated Audio for Digital Storage Media at up to about 1.5 Mb/s," ISO/IEC 11172-3, Aug. 1993.

[4] A. Sugiyama and A. Hirano, "A Subband Adaptive Filtering Algorithm with Adaptive Intersubband Tap Assignment," Trans. IEICE (Institute of Electronics, Info. and Commun. Engrs.), vol.E77-A, No.9, pp.1432-1438, Sep. 1994.

[5] G. C. Goodwin and K. S. Sin, "Adaptive Filtering, Prediction and Control ," Prentice-Hall, 1984.