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고정 반사계수를 갖는 격자 트랜스버설 결합 적응필터

A Lattice Transversal Joint Adaptive Filter with Fixed Reflection Coefficients

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요 약 본 논문에서는 적은 계산량으로도 빠른 수렴 속도를 얻을 수 있도록 고정반사계수를 갖는 격자 트랜스버설 결합 적응필터를 제안하였다. 필터의 반사계수는 음성신호의 통계적 특성에 따라 주어지며, 격자예측기의 차수는 1을 제안하였다. 제안한 적응필터가 트랜스버설 적응필터에 비해 무시할 만한 추가 계산량으로도 고속 수렴함이 실험을 통하여 입증되었다. ITU-T G.168 표준안의 대역제한된 유성음신호를 사용한 실험에서 제안된 적응필터는 적응 트랜스버설 필터에 비해 약 6배 빨리 수렴하였다.

Abstract We present a lattice transversal joint (LTJ) adaptive filter with fixed reflection coefficients to achieve fast convergence with low complexity. The reflection coefficients of the filter are given by the statistics of speech signals, and the proposed order of the lattice predictor is one. Experimental results confirm that as compared to the adaptive transversal filter, the proposed adaptive filter achieves fast convergence with a negligible increase in complexity. The proposed adaptive filter converges around six times faster than the adaptive transversal filter in case of the band-limited voiced signal from the ITU-T G.168 standard.

Key Words : Adaptive filter, Lattice filter, Reflection coefficients, Speech signal, Convergence

Ⅰ. Introduction

The adaptive transversal filter employing a least mean square (LMS) algorithm has been extensively used in many applications. However, its convergence performance is degraded in the case of highly correlated signals such as speech signals[1]. The adaptive lattice filter can be an excellent alternative to the adaptive transversal filter since it has the fast convergence property[2]. The lattice transversal joint (LTJ) adaptive filter is a modified adaptive lattice filter in which the lattice predictors of order higher than that of the auto-regressive (AR) process of the input signal are removed[3]. Therefore, the LTJ adaptive filter has a lower complexity than the adaptive lattice filter. However, the steady-state performance of both filters is poor since there is a delay of one sample between the update of the filter coefficient and that of the reflection coefficient[4]. This problem can be solved by filter coefficient compensation[4]. However, this involves a complexity of $2NL$ per sample, where N and L represent the length of the adaptive filter and the order of the lattice predictor, respectively[4]. This complexity is very high, considering that the complexity of the adaptive transversal filter is $2N$. Hence, the high

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complexity of filter coefficient compensation is a major obstacle in the use of LTJ adaptive filters.

To reduce this high complexity, block compensation methods have been proposed[5,6], in which filter coefficient compensation is performed for an entire block at once rather than for every sample. Thus, the average complexity can be reduced. However, when filter coefficient compensation is performed, the complexity of real-time implementation cannot be reduced because the complexity per sample is still 2*NL*.

In this paper, an LTJ adaptive filter with fixed reflection coefficients is proposed. The filter does not require filter coefficient compensation because the reflection coefficients do not vary. Therefore, this filter brings about low computational complexity as well as fast convergence. The order of the lattice predictor and the values of the reflection coefficients are derived by investigating the statistics of the reflection coefficients of speech signals.

II. LTJ Adaptive Filter

The LTJ adaptive filter is shown in Figure 1, where $d(n)$, $x(n)$, and $y(n)$ represent the desired input signal, reference input signal and filter output signal, respectively [3]. L and N denote the order of the lattice predictor and the number of filter coefficients, respectively; $c_m(n)$ is the m-th coefficient of the transversal filter block, and $f_m(n)$ and $b_m(n)$ represent the m-th forward prediction error and the m-th backward prediction error, respectively; $k_m(n)$ is the m-th reflection coefficient. Exploiting the fact that a speech signal is well modeled as an output of AR processes, in this filter, lattice predictors are not used after the L-th stage.

The update equations for the lattice predictors are as follows:

$$f_m(n) = f_{m-1}(n) + k_m(n) b_{m-1}(n-1) \tag{1}$$

$$b_m(n) = b_{m-1}(n-1) + k_m(n) f_{m-1}(n) \tag{2}$$

$$P_{m-1}(n) = \beta P_{m-1}(n-1) + (1-\beta)(f_{m-1}^2(n) + b_{m-1}^2(n-1)) \tag{3}$$

$$k_m(n+1) = k_m(n) - \frac{2\mu_L}{P_{m-1}(n)}(b_m(n) f_{m-1}(n) + b_{m-1}(n-1) f_m(n)) \tag{4}$$

where $P_m(n)$, β, and μ_L denote the estimated power of the m-th backward prediction error, the constant for controlling the power estimation, and the step size in the estimation of the reflection coefficients. The update equations for the transversal filter block are as follows:

$$y(n) = \sum_{m=0}^{N-1} c_m(n) b_m(n) \tag{5}$$

$$e(n) = d(n) - y(n) \tag{6}$$

$$P_L(n) = \beta P_L(n-1) + (1-\beta) b_L^2(n-1) \tag{7}$$

$$c_m(n+1) = c_m(n) + 2\mu e(n) \frac{b_m(n)}{P_m(n)} \quad (0 \le m \le L-1) \tag{8}$$

$$c_m(n+1) = c_m(n) + 2\mu e(n) \frac{b_m(n)}{P_L(n)} \quad (L \le m \le N-1) \tag{9}$$

where μ denotes the step-size parameter of the transversal filter.

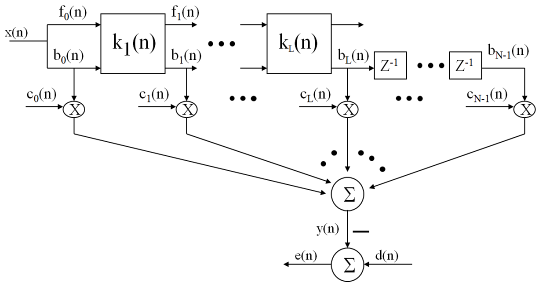

그림 1. LTJ 적응필터
Fig. 1. The LTJ adaptive filter.

Ⅲ. The Proposed Adaptive Filter

Both the LTJ adaptive filter and the lattice adaptive filter have poor steady-state performance because of the one-sample delay between the updates of the reflection coefficients and the coefficients of the transversal filter block. The analysis of the cause of the steady-state problem and its solution is described in detail in[4,5]. The solution is known as filter coefficient compensation, and it requires $2NL$ multiplications and additions per sample[4]. This degree of complexity is very high since the complexity of the adaptive transversal filter is $2N$. In other words, the complexity of filter coefficient compensation is L times that of the adaptive transversal filter.

Filter coefficient compensation is necessary for dealing with time-varying reflection coefficients. Therefore, fixed reflection coefficients are employed. Since these coefficients are time-invariant, (4) need not be implemented and the power estimation is only needed for the first stage; the estimated powers for the other stages are given by the following equations:

$$P_m(n) = P_{m-1}(n) \cdot (1 - k_m^2) \tag{10}$$

$$P_m(n) = P_0(n) \cdot \prod_{j=1}^{m} (1 - k_j^2) \tag{11}$$

Defining qm as

$$q_m = \prod_{j=1}^{m} (1 - k_j^2) \quad (1 \le m \le L)\ (q_0 = 1) \tag{12}$$

(11) is expressed as

$$P_m(n) = P_0(n) \cdot q_m \tag{13}$$

where q_m is calculated in advance. Therefore, the power estimation is required only for the input signal $x(n)$ as follows:

$$P_0(n) = \beta P_0(n-1) + (1-\beta) x^2(n) \tag{14}$$

The implementation of (14) is required in the adaptive transversal filter as well. Substituting (13) in the second term on the right-hand side of (9), we obtain

$$2\mu e(n) \frac{b_m(n)}{P_0(n) q_m} = 2 \frac{\mu}{P_0(n)} \frac{1}{q_m} e(n) b_m(n) \tag{15}$$

If we define $\mu'(n)$ and v_m as

$$\mu'(n) = \mu / P_0(n) \tag{16}$$

$$v_m = 1/q_m \tag{17}$$

and substitute (16) and (17) in (15), (8) can be expressed as

$$c_m(n+1) = c_m(n) + 2\mu'(n) v_m e(n) b_m(n) \tag{18}$$

where v_m acts as a factor for power normalization, and the normalized step-size parameter can be written as

$$\tilde{\mu}_m(n) = \mu'(n) \quad (m = 0) \tag{19}$$

$$\tilde{\mu}_m(n) = \mu'(n) \cdot v_m \quad (1 \le m \le L) \tag{20}$$

$$\tilde{\mu}_m(n) = \tilde{\mu}_L(n) \quad (L+1 \le m \le N-1) \tag{21}$$

Substituting (19), (20) and (21) in (18), we obtain

$$c_m(n+1) = c_m(n) + 2\tilde{\mu}_m(n) e(n) b_m(n) \tag{22}$$

The complexity of the proposed adaptive filter is higher than that of the adaptive transversal filter because of the lattice filtering of (1) and (2) and the step-size normalization in (20). The implementation of (1) requires L-1 additions and L-1 multiplications, as shown in Figure 1, since the L-*th* forward prediction error signal $f_L(n)$ is not required for the lattice filtering. The implementation of (2) requires L additions and L multiplications and that of (20) requires L multiplications. Therefore, the total additional complexity is $2L$-1 additions and $3L$-1 multiplications.

In order to ensure fast convergence of the proposed LTJ adaptive filter, it is important to obtain the appropriate fixed reflection coefficients for efficiently de-correlating the speech signal. To examine the statistical distribution of the reflection coefficients, sentences from actual speeches were collected and analyzed by using the computer program VOICEBOX [7]. The histogram of the reflection coefficients is shown in Figure 2. The sampling rate for the speech was 8 kHz, and the order of the lattice predictor was ten.

Ten reflection coefficients were investigated, among which the statistical distribution of the first reflection coefficient was very asymmetric and its mean value was far from zero. Therefore, it is expected that the first reflection coefficient can be helpful to de-correlate the speech signal. This expectation is experimentally confirmed to be true. The order of the lattice predictor should be selected such that fast convergence and low complexity are realized. We propose that the order of the lattice predictor is one. Therefore, the additional complexity of the proposed adaptive filter over the adaptive transversal filter is only one addition and two multiplications per iteration.

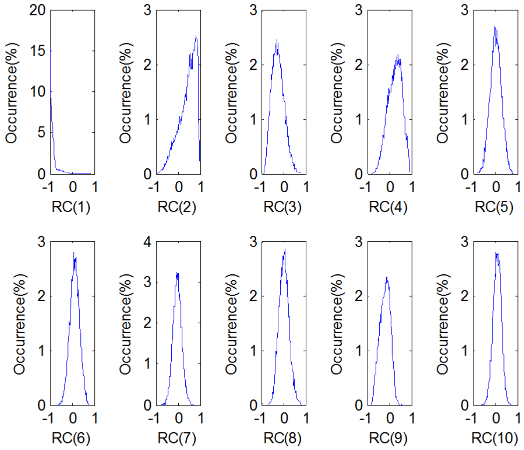

그림 2. 반사계수의 히스토그램
Fig. 2. Histogram of reflection coefficients

Ⅳ. Experimental Results

The convergence performance of the proposed adaptive filter is compared with that of the adaptive transversal filter. We conducted our experiment under an echo cancellation environment. The signals were sampled at 8-kHz. For the echo path, we used the impulse response of echo path model 5 of the ITU-T G.168 standard[8]. The lengths of the echo path and the adaptive filter are 128 samples. For the input signal *x(n)*, the band-limited voiced signal from the ITU-T G.168 standard and a real speech signal were used. Echo return loss enhancement (ERLE)[3] was used as the performance index. The fixed reflection coefficients was k_1 = -0.8225.

1. Voiced signal from the ITU-T G.168

The convergence performances of both adaptive filters in the case where the input signal is the band-limited voiced signal are shown in Figure 3. White Gaussian background noise was added to obtain an echo-to-noise ratio of 40 dB. The ERLE was calculated from the results of 100 independent trials. The proposed adaptive filter converges around six times faster than the adaptive transversal filter.

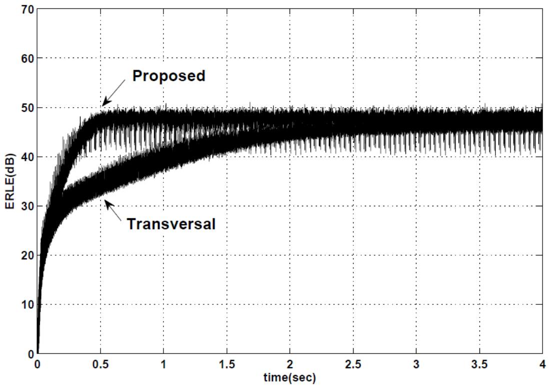

그림 3. ERLE(dB) : 유성음 ITU-T G.168
Fig. 3. ERLE(dB) : ITU-T G.168 Voiced Signal

2. Speech signal

Speech signals were obtained from ten speakers (five male and five female speakers). For each speaker, the speech was divided into ten intervals to which each cancellation was performed. And each interval covers 4 seconds. The echo-to-noise ratio was 20, 30, or 40 dB. A comparison of the overall ERLE of both filters is presented in Table 1. It is evident that the proposed adaptive filter outperformed the adaptive transversal filter; when the proposed adaptive filter is used for speech signals, we can achieve an average performance improvement of approximately 3 dB in ERLE.

표 1. ERLE(dB) : 음성신호
Table 1. ERLE(dB) : for speech signals

Echo to Noise Ratio (dB)	Transversal	Proposed
20	16.5	19.3
30	19.9	22.8
40	21.2	24.4

Ⅴ. Conclusion

We have proposed an LTJ adaptive filter with fixed reflection coefficients. The proposed order of the lattice predictor is one, and the reflection coefficients are given by the statistics of the reflection coefficients for the speech signal. The complexity of the proposed adaptive filter involves only one more addition and two more multiplications per iteration, compared to that of the adaptive transversal filter. Experimental results confirm that the proposed adaptive filter converges six times faster than the transversal adaptive filter for the voiced signal from the ITU-T G.168 standard and that for real speech signals, an average improvement of approximately 3 dB in ERLE over the adaptive transversal filter was achieved. Therefore, compared to the adaptive transversal filter, the proposed adaptive filter can achieve a faster convergence rate with a negligible increase in complexity.

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