

LTJ adaptive filter with fixed reflection coefficients for speech applications

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Abstract— This paper presents a lattice transversal joint (LTJ) adaptive filter whose reflection coefficients are fixed. Reflection coefficients are given by the statistics of speech signal and the proposed order of the lattice predictor is one. The additional computational complexity over the transversal adaptive filter is only 1 addition and 2 multiplications per sample. Simulation results verify that the proposed LTJ adaptive filter achieves fast convergence with negligible additional complexity compared to the transversal adaptive filter.

Keywords— lattice adaptive filter, transversal adaptive filter, reflection coefficients, speech signal, convergence

I. INTRODUCTION

The transversal adaptive filter using least mean square (LMS) algorithm has been widely used in many applications. However, its convergence performance is degraded with highly correlated signal such as speech [1]. The lattice adaptive filter can be an excellent alternative to the adaptive transversal filter since it has fast convergence property [2]. The lattice transversal joint (LTJ) adaptive filter is a modified lattice adaptive filter where lattice predictors after the order of the auto-regressive (AR) process of the input signal are removed [3]. So, it is very suitable for speech signal and its computational complexity is low compared to the lattice adaptive filter. However, both the adaptive lattice filter and the LTJ adaptive filter have poor steady-state performance since the filter coefficients are updated one sample after the reflection coefficients update [4]. This problem can be solved by using the filter coefficients compensation. However, it requires complexity of $2NL$ per sample where N and L represent the length of adaptive filter and the order of the lattice predictor, respectively [4]. This computational complexity is very severe considering that complexity of the transversal filter is $2N$. So, filter coefficients compensation is a big obstacle using the LTJ adaptive filter.

In order to reduce this heavy computational complexity, block compensation methods were proposed [5-6] where filter coefficients compensation is done once a block instead of every sample. So, its average complexity can be reduced. However, it can not reduce the computation complexity with respect to real time implementation because the computational complexity is still $2NL$ at the sample when

filter coefficients compensation is performed. So, it is certain that the block compensation method is not able to be an ultimate solution.

In this paper, the LTJ adaptive filter with fixed reflection coefficients is proposed. It does not require filter coefficients compensation because the reflection coefficients do not vary. So, it has 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 reflection coefficients of speech signal. The excellent performance of the proposed adaptive filter is shown by computer simulation.

II. THE LTJ ADAPTIVE FILTER

Figure 1 shows the LTJ adaptive filter, where $d(n)$, $x(n)$ and $y(n)$ represent the desired input, reference input and filter output, respectively [3]. And L and N mean the order of lattice predictor and the number of filter coefficients, respectively. And $c_m(n)$ is the m -th coefficient of the transversal filter and $f_m(n)$ and $b_m(n)$ represent the m -th forward prediction error and the m -th backward prediction error respectively, and $k_m(n)$ is the m -th reflection coefficient.

Using the fact that speech signal is well modeled as an output of an AR processes, it does not have lattice predictors after L -th stage. So, it is very efficient in computational complexity since it does not require computations for lattice filtering and update of the reflection coefficients after L -th stage.

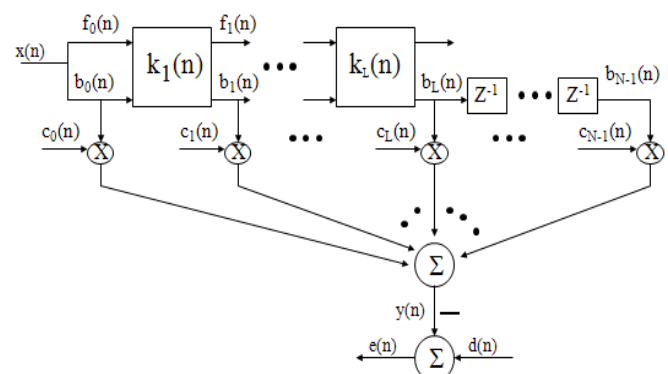


Figure 1. The LTJ adaptive filter

The update equations of the lattice predictor are as follows.

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

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

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

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

$P_m(n)$, β and μ_L denote the estimate power of the m -th backward prediction error, the controlling constant of the power estimation and the step size for the estimate of the reflection coefficients.

The update equations of the transversal filter part are as follows.

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

$$e(n) = d(n) - y(n) \quad (6)$$

$$P_L(n) = \beta P_L(n-1) + (1-\beta)b_L^2(n-1) \quad (7)$$

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

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

μ denotes the step size parameter of the transversal filter.

III. THE LTJ -AF : FIXED REFLECTION COEFFICIENTS

Not only the LTJ adaptive filter but also the lattice adaptive filter have poor steady-state performance because of one sample mismatch between the update of the reflection coefficients and the coefficients of the transversal filter part. The analysis of the reason and the solution for overcoming the steady-state problem are described in detail in [4, 5]. This solution is called filter coefficients compensation and it requires 2NL multiplications and additions [4]. This amount of computational complexity is very heavy since the computation complexity of transversal filter is 2N. Namely, the computational complexity of the filter coefficients compensation is L times bigger than that of the transversal adaptive filter. Therefore, the LTJ adaptive filter requiring the filter coefficients compensation has a problem in real-time implementation.

It is time-varying reflection coefficients that require filter coefficients compensation. So, the fixed reflection coefficients are employed in order not to require the filter coefficients compensation. Since the reflection coefficients are time invariant, (4) is not needed and the power estimation is only needed in the first stage and those of the other stages are given as the following equations.

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

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

Defining q_m as

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

(11) is described as

$$P_m(n) = P_0(n) \cdot q_m \quad (13)$$

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

$$P_0(n) = \beta P_0(n-1) + (1-\beta)x^2(n) \quad (14)$$

(14) is also required in the transversal adaptive filter using NLMS adaptive algorithm. Substituting (13) in the second term on the right-hand side of (9), we obtain

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

If we define $\mu^1(n)$ and q_m^1 as

$$\mu^1(n) = \mu / P_0(n) \quad (16)$$

$$q_m^1 = 1/q_m \quad (17)$$

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

$$c_m(n+1) = c_m(n) + 2\mu^1(n)q_m^1 e(n)b_m(n) \quad (18)$$

q_m^1 acts as a power normalization and the normalized step-size parameter can be written as

$$\mu_m^2(n) = \mu_m^1(n) \quad (m = 0) \quad (19)$$

$$\mu_m^2(n) = \mu_m^1(n) \cdot q_m^1 \quad (1 \leq m \leq L) \quad (20)$$

$$\mu_m^2(n) = \mu_L^2(n) \quad (L+1 \leq m \leq N-1) \quad (21)$$

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

$$c_m(n+1) = c_m(n) + 2\mu_m^2(n)e(n)b_m(n) \quad (0 \leq m \leq N-1) \quad (22)$$

The additional complexity of the proposed LTJ adaptive filter over the transversal adaptive filter results from the lattice filtering of (1), (2) and step-size normalization in (20). (1) requires L-1 additions and L-1 multiplications since L-th forward prediction error signal $f_L(n)$ is not required as shown in fig 1. (2) requires L additions and L multiplications and (20)

requires L multiplications. So, additional complexity in total is $2L-1$ additions and $3L-1$ multiplications.

In order to ensure fast convergence of the proposed LTJ adaptive filter, it is important to find proper fixed reflection coefficients which efficiently de-correlate the speech signal. To examine the statistical distribution of reflection coefficients [7,8], Korean, Japanese and English sentences are used as speech signal and [9] was used as an computer program. The histogram of the reflection coefficients is shown in Figure 2. Sampling rate of the speech was 8 kHz and the order of the lattice predictor was ten.

Among ten reflection coefficients, the statistical distribution of the first reflection coefficient is very asymmetric and its mean value is far from zero. So, we can expect that the first reflection coefficients can be helpful to improve the convergence speed. And our simulation results according to the change of the number of the lattice stage confirmed this expectation. In this paper, we recommend that the order of the lattice predictor is one which can give fast convergence as well as efficient complexity. Therefore, the additional computational complexity of the proposed LTJ adaptive filter is only 1 addition and 2 multiplications per sample comparing with the transversal adaptive filter.

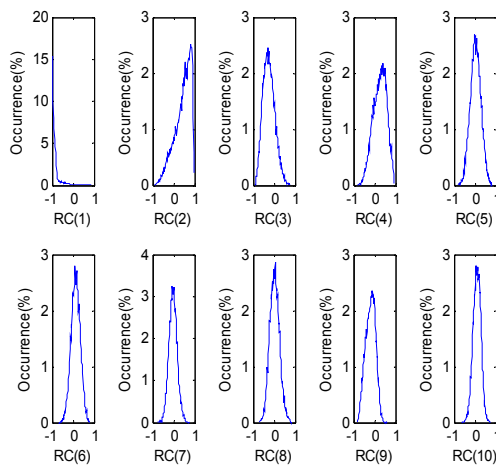


Figure 2. A Histogram of reflection coefficients

IV. SIMULATIONS

For the computer simulation, we have considered echo cancellation environment. The convergence performance of the proposed adaptive filter is compared with those of the transversal adaptive filter.

The input signal and the echo path were given with a sampling rate of 8-kHz. For the echo path, we used the impulse response of the echo path model of [10]. The length of the echo path and the adaptive filter are 128 samples. For the input signal $x(n)$, real speech signal were used. The white Gaussian background noise is added to make the echo to noise ratio 40 dB. The echo return loss enhancement (ERLE) [7] was used as performance index. The same value of step

size was given to the transversal adaptive filter and the proposed adaptive filter. The order of the lattice predictor of the proposed adaptive filter is 1. So, the additional complexity of the proposed adaptive filter over the transversal adaptive filter is 2 multiplications and 1 addition per sample. This complexity is very small considering the complexity of the transversal adaptive filter is $2N$ multiplications per sample.

Speech signals of ten speakers were used. For each speaker, echo cancellation for 4 seconds about 10 different speech intervals were performed. Figure 3 shows a result of a certain 4 seconds period. (a) represents the input speech signal. (b) and (c) show the ERLE of the transversal filter and the proposed adaptive filter, respectively.

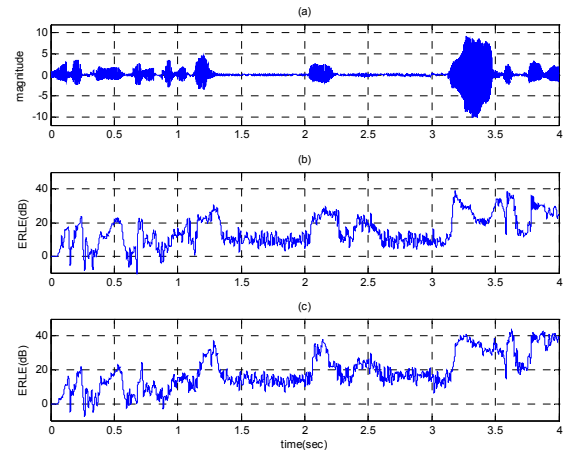


Figure 3. ERLE(speech signal)

The performance improvement of the proposed adaptive filter over the transversal adaptive filter is clear. Although the amount of the performance improvement varies according to person, it is evident that the proposed adaptive filter outperformed the transversal adaptive filter and we can expect 3.6 dB performance improvements in average with the proposed adaptive filter for speech signal.

V. CONCLUSIONS

We have proposed the LTJ adaptive filter with fixed reflection coefficients for speech applications. The proposed order of the lattice predictor is one and values of the reflection coefficients are given by measuring the statistics of reflection coefficients of speech signals. The additional complexity of the proposed adaptive filter over the transversal adaptive filter is only 1 addition and 2 multiplications.

Computer simulations showed that the proposed adaptive filter achieved 3.6 dB improvements in average over the transversal adaptive filter for the real speech signals. Therefore, the proposed adaptive filter can achieve faster convergence rate with negligible additional complexity compared to the transversal adaptive filter.

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REFERENCES

- [1] A. H. Sayed, *Fundamentals of Adaptive Filtering*, John Wiley & Sons, 2003.
- [2] E. H. Satorious and S. T. Alexander, "Channel equalization using adaptive lattice algorithm," *IEEE Transactions on Communication*, vol. 27, pp. 899-905, June, 1979.
- [3] Jae-Ha Yoo, Sung-Ho Cho and Dae-Hee Youn, "An acoustic echo cancellation based on the adaptive lattice-transversal joint (LTJ) filter structure," *IEICE Transactions on Fundamentals of Electronics, Communications and Computer Sciences*, vol. E81-A (9), pp. 1951-1954, September, 1998.
- [4] Naoki Tokui, Kenji Nakayama and Akihiro Hirano, "A synchronized learning algorithm for reflection coefficients and tap weight in a joint lattice predictor and transversal filter," *Proceedings of International Conference on Acoustics, Speech, and Signal Processing*, 2001.
- [5] Jae-Ha Yoo, "A general analysis and complexity reduction for the lattice transversal joint adaptive filter," *International Technical Conference on Circuits/Systems, Computers and Communications*, 2002.
- [6] Naoki Tokui, Kenji Nakayama and Akihiro Hirano, "Block implementation of a synchronized learning algorithm in adaptive lattice filters," *Proceedings of International Conference on Acoustics, Speech, and Signal Processing*, 2003.
- [7] B. Farhang-Boroujeny, *Adaptive Filters Theory and Applications*, John Wiley & Sons, 1998.
- [8] Peter Vary and Rainer Martin, *Digital Speech Transmission : enhancement, coding and error concealment*, John Wiley & Sons, 2006.
- [9] <http://www.ee.ic.ac.uk/hp/staff/dmb/voicebox/voicebox.html>
- [10] ITU-T G.168: *Recommendations for Digital Network Echo Cancellers*, International Telecommunication Union Telecommunication Standardization Sector, Aug, 2004.