DOUBLE-TALK ECHO CANCELING BY NON-FREEZING ADAPTIVE ALGORITHM

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ABSTRACT

The Extended Correlation LMS (ECLMS) algorithm has been introduced to challenge double-talk in the echo canceling system. Here, we propose a new fast implementation of the ECLMS algorithm by using the time-lag of the correlation in FFT kernel to decrease the computational load to only 2.7% of the ECLMS algorithm. Initialization and normalization methods are also defined to increase the convergence in speech signal environment. The computer simulation results support the theoretical findings and verify the robustness of the FECLMS algorithm in the double-talk situation.

1. INTRODUCTION

In hands-free set mobile radio telephone or teleconference system, where acoustic feedback exists between the loudspeaker and microphone, the quality of communication is degraded severely. The degradation of the quality is due to the acoustic echo impulse response of the car or teleconference room. The LMS algorithm and normalized LMS (NLMS) algorithm [1] for adaptive FIR filter are very popular for their simplicity and predictable behavior. The Frequency Domain Adaptive Filter (FDAF) [2] and Frequency Bin Adaptive Filter (FBAF) [3] are well known to have comparatively lower computational complexity than time-domain FIR filter. However, in the double-talk situation when both the near-end and the far-end signal are presented, the performance of echo cancellation is degraded. This is because the error signal used for obtaining the gradient in tap adaptations will be considerably large and therefore, the process of tap adaptation is severely damaged.

The conventional algorithm usually stops adaptation whenever double-talk sensor detects this condition and it keeps freezing the tap coefficient data during the double-talk condition. Stopping the tap adaptation is just a passive action to handle the double-talk condition and it causes lower speed of adaptations. Other works for challenging the problem of the double-talk situation in the echo canceling can be found in [4], [5], [6] and [7] which cause much more complexity adding to a simple LMS algorithm.

To solve the double-talk problem, correlation LMS (CLMS) algorithm [8], [9] and the Expanded CLMS (ECLMS) algorithm [10], which utilize the correlation function of input signal instead of the input signal itself, have been proposed. Therefore, we can continue the tap adaptation (non-freezing) even in the double-talk situation, without misleading the estimation process. However, for large number of tap coefficients, the CLMS and the ECLMS algorithms require heavy computational load for implementation.

In this paper, we have introduced the frequency domain ECLMS (FECLMS) algorithm, by using the time-lag of the correlation in the FFT kernel which gives a lower computational complexity. In order to increase the convergence rate in speech signal environment, we have used an algorithm for normalizing the convergence factor and we have also used initialization in computation of correlation to obtain a better result in the FECLMS algorithm. The computer simulation results verify that the FECLMS algorithm achieves a better result compared with FDAF algorithm and also show the robustness of the FECLMS algorithm in the double-talk condition. Moreover, the required computational load is reduced to only 2.7 % of the ECLMS algorithm.

2. DOUBLE-TALK IN ECHO CANCELER

In the echo canceling system shown in Fig.1, the acoustic impulse response of the teleconference room is estimated by an adaptive algorithm such as the LMS algorithm. The adaptive filter output $\widetilde{y}(n)$ is given by:

$$\widetilde{y}(n) = \sum_{i=0}^{N-1} \widetilde{h}_i x(n-i) \tag{1}$$

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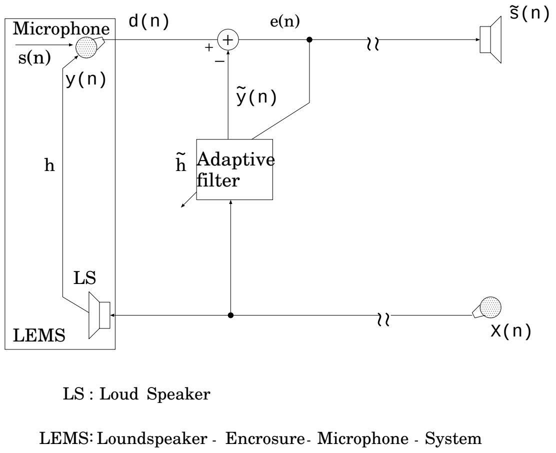

Figure 1: Echo canceling system

where N is the number of tap, $\widetilde{h}_i$ is the tap coefficient of the FIR filter and $x(n)$ is the far-end signal. Then, the error signal between the desired signal $d(n)$ and the output signal $\widetilde{y}(n)$ is as follows:

$$e(n) = d(n) - \widetilde{y}(n) \tag{2}$$

which is used to adapt the tap coefficients of the adaptive filter by the LMS algorithm as follows:

$$\widetilde{h}_i(n+1) = \widetilde{h}_i(n) + 2\mu e(n)x(n-i) \tag{3}$$

where μ is the step size for the tap coefficients adaptation. The condition of double-talk in the echo canceler is occurred when the near-end signal $s(n)$ comes to exist simultaneously with the far-end signal $x(n)$. In this condition the microphone signal consists of both the echo returned signal $y(n)$, and the near-end signal, $s(n)$, that is:

$$d(n) = y(n) + s(n) \tag{4}$$

where:

$$y(n) = \sum_{i=0}^{N-1} h_i x(n-i) \tag{5}$$

where h and N are the echo impulse response and its length. Thus, the error signal $e(n)$ contains also the near-end $s(n)$, which is uncorrelated with the input signal $x(n)$, and therefore, the gradient search is misled to estimate the correct echo path impulse response. In this situation, we say, the double-talk has been occurred and the efficiency of the the conventional algorithm such as the LMS algorithm will be destroyed.

To avoid this problem, the extended correlation LMS (ECLMS) algorithm has been proposed [10]. In the following section, we propose the new proposed frequency-domain ECLMS (FECLMS) algorithm.

3. FECLMS ALGORITHM

To reduce the complexity in computation, we propose the FECLMS algorithm that has been shown in Fig.2. First, we take the fast Fourier transform (FFT) of the input signal correlation based on time-lag k in the fast Fourier transform kernel.

$$\Phi_{xx}(n,p) = \sum_{k=0}^{N-1}\sum_{j=0}^{n} x(j)x(j-k)W^{-kp} \tag{6}$$

where W shows complex exponential $e^{j(2\pi/N)}$, Φ_{xx} shows the FFT of the correlation and p is the frequency variable of FFT. Similarly, take FFT of the cross correlation between $x(n)$ and $d(n)$ based on time-lag k.

$$\Phi_{dx}(n,p) = \sum_{k=0}^{N-1}\sum_{j=0}^{n} d(j)x(j-k)W^{-kp} \tag{7}$$

where Φ_{dx} shows the FFT of the cross correlation. After the necessary substitutions from (4) and (5) into (7) using (6), the FFT of the cross correlation function Φ_{dx} between d(n) and x(n) signals will be as follows:

$$\Phi_{dx}(n,p) \simeq H_p \Phi_{xx}(n,p) \tag{8}$$

where H_p is pth element of the FFT of the echo impulse response vector $\mathbf{h} = [h_0 h_1 \cdots h_{N-1}]$. Then, on the basis of eqn.(8), the adaptive filter in which the input signal is the FFT of the correlation function of the far-end signals is defined by:

$$\widetilde{\Phi}_{dx}(n,p) = \widetilde{H}_p(n)\Phi_{xx}(n,p) \tag{9}$$

where $\widetilde{H}_p(n)$ is the adaptive filter tap coefficient and $\widetilde{\Phi}_{dx}(n,p)$ is the estimation value of $\Phi_{dx}(n,p)$. Next, we define the cost function for adapting the adaptive filter tap coefficients. This function is defined as follows:

$$J(n,p) = E\left[\mathcal{E}^*(n,p)\mathcal{E}(n,p)\right] \tag{10}$$

where

$$\mathcal{E}(n,p) = \Phi_{dx}(n,p) - \widetilde{\Phi}_{dx}(n,p) \tag{11}$$

The superscript * shows the Hermitian transposition. To obtain the gradient value of eqn.(10), we differentiate (10) with respect to tap coefficient $\widetilde{H}_p$:

$$\begin{aligned} \nabla J(n,p) &= \frac{\partial}{\partial \widetilde{H}_p(n)} E[\mathcal{E}^*(n,p)\mathcal{E}(n,p)] \\ &= -2E[\mathcal{E}(n,p)\Phi^*_{xx}(n,p)] \end{aligned} \tag{12}$$

From (12) we derive the steepest-descent Frequency-Domain ECLMS (FECLMS) algorithm as follows:

$$\widetilde{H}_p(n+N) = \widetilde{H}_p(n) + 2\mu_f \mathcal{E}(n,p)\Phi^*_{xx}(n,p) \tag{13}$$

where μ_f is convergence parameter. To improve the

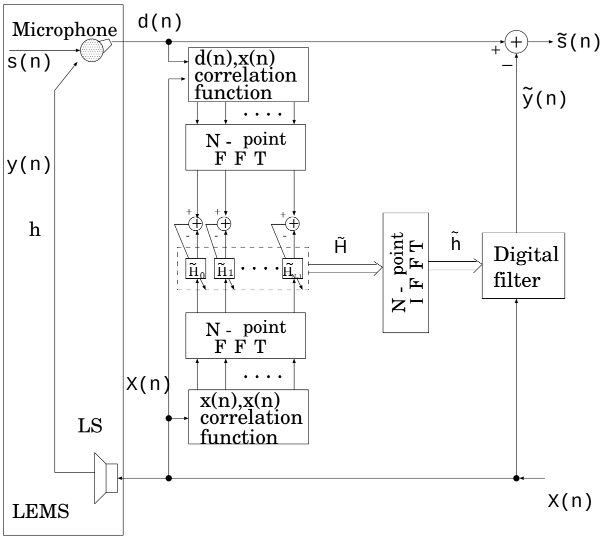

Figure 2: Echo canceler using FECLMS algorithm

convergence speed and tracking of the FECLMS algorithm for correlated input signal, such as the speech signal, we have adaptively normalized the convergence factor $\mu_p(n)$ to the power of the signal in each frequency bin, $\sigma_p^2(n)$. The power of each bin is calculated using the following recursive equation:

$$\sigma_p^2(n) = \frac{n}{n+1}\sigma_p^2(n-1) + \frac{1}{n+1} \mid \Phi_{xx}(n,p) \mid^2 \quad (14)$$

and the convergence factor is normalized according to the following equation:

$$\mu_p(n) = \frac{\gamma}{\sigma_p^2(n)} \quad (15)$$

where $0 \leq \gamma \leq 1$ is a constant factor.

4. COMPUTATIONAL COMPLEXITY

In order to show one of the merits of the proposed FECLMS algorithm, we compare the computational complexity of the proposed algorithm with the one of the conventional ECLMS algorithm [10]. In the ECLMS algorithm in order to extract N (N is number of tap coefficients) output samples, we need $2N^2$ multiplications. On the other hand, in the proposed FECLMS algorithm we need three N-point FFTs and only $2N$ multiplications are required in the rest of processing, that is:

$$3 \times \frac{N}{2}\log_2 N + 2N \quad (16)$$

In the Table 1, the ratios of the computational load for FECLMS to ECLMS algorithm are given with respect to various number of tap coefficients N. So, for instance in $N = 256$ the proposed FECLMS algorithm requires only 2.7% of computational load for the ECLMS algorithm. This makes the hardware implementation of the FECLMS algorithm a realistic matter using fewer chips of DSP or in considering of the mass production, it requires less LSI area.

Table 1: Computational load ratio

N	FECLMS/ECLMS
64	0.085
128	0.049
256	0.027

5. SIMULATION RESULTS

In order to show the capability and robustness of the proposed FECLMS algorithm, we have performed several computer simulations. To measure the performance of the algorithm, we have used $IRER$(Impulse Response Estimation Ratio) which is defined by the following equation:

$$IRER = 10\log_{10}\left[\frac{|\mathbf{h} - \widetilde{\mathbf{h}}(n)|^2}{|\mathbf{h}|^2}\right] \quad (17)$$

where $\mathbf{h}$ is the vector of the echo impulse response. $\widetilde{\mathbf{h}}(n)$ is the vector of the adaptive filter tap coefficients.

In Fig.3, the convergence characteristics for the proposed FECLMS and the conventional FDAF algorithms [2] are given in the single-talk situation. As we can see the proposed algorithm has a better performance compared with the conventional algorithm. In Fig.4, the same simulation as was performed before but under double-talk situation is shown. As we can see the performance of the proposed algorithm is better than the conventional one. The only thing to be mentioned here, is that, in the proposed algorithm the normalization process is performed in tap adaptations and also we utilize the initialization to compute the correlation values. This is because in the recursion formulas for computation of correlation, the exact values are not given at the beginning, so we had better to use the correlation function after convergence. Therefore, they were initialized in the proposed algorithm. At any rate, the proposed algorithm is robust in the double-talk situation. In the last simulation results which are given in Fig.5, we have used the real speech data for both the near-end and the far-end signal. Counting from one to three are pronounced in Japanese by two different speakers. So, this simulation is in double-talk situation. The performance of the normalized FECLMS compared with FDAF algorithm is shown and we can see the robustness of our algorithm.

6. CONCLUSION

A new implementation of the extended correlation LMS algorithm in the frequency domain was proposed. The frequency domain is obtained using the fast Fourier

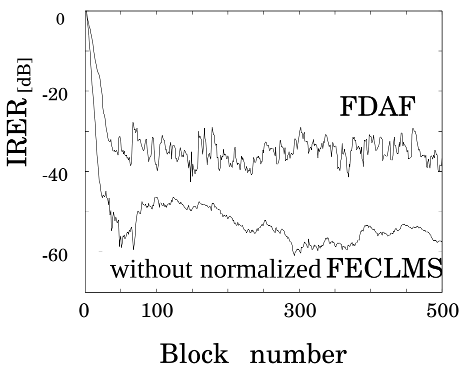

Figure 3: IRER with white-noise N=16, μ_f=0.9

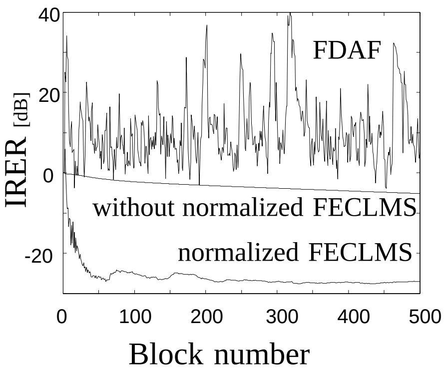

Figure 4: Normalized effect in the FECLMS algorithm with color noise N=16, μ_f =0.03, γ =1

transform where in its kernel the lag time of the correlation function was used as the time variable. This is in a sense different from the conventional approach where sample time is used. We have shown that the proposed algorithm required only 2.7% of the multiplications required in the conventional algorithm. So the hardware size becomes small and realistic as one considers the practical usage of echo canceling in hand free-set mobile telephone. The robustness of the proposed algorithm in the double-talk situation was shown by computer simulation results in which we used two new techniques for normalization of tap coefficient adjustment and initialization of the correlation function. Meanwhile for real implementation, we have also used the real speech signal in order to evaluate the proposed algorithm and the result was quite satisfactory.

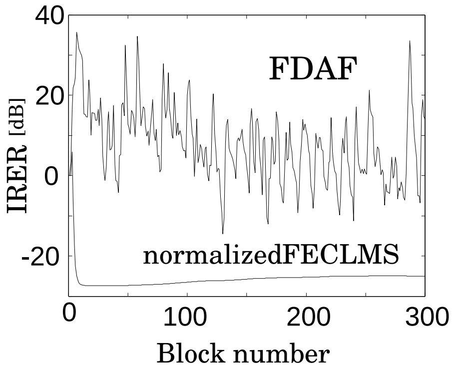

Figure 5: IRER with speech signal N=16, γ=1

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