A New Frequency Domain Implementation of ECLMS Algorithm for Double-Talk Echo Canceling

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Abstract: In the echo canceling for hand-free set mobile radiotelephone or teleconference system, the double-talk is occurred when both the near-end and the far-end speakers talk simultaneously. The conventional adaptive digital FIR filter using algorithm such as LMS fails to track the echo path in this condition. Because the error signal is contaminated to the near-end signal to estimate the gradient correctly. The Extended Correlation LMS (ECLMS) algorithm has been introduced by authors to challenge the double-talk in the echo canceling system. Here, the gradient search algorithm is obtained from the correlation function and the error lags between autocorrelation and cross-correlation values of the near and the far end signals. To reduce the computational load of the ECLMS algorithm, the frequency domain version of the algorithm has also previously been defined by the authors but the performance was not appropriate. In this paper, to implement the exact linear convolution in the frequency domain a new fast implementation of the ECLMS algorithm by using zero padding and Half Overlapped Save (HOS) method has been proposed. Also the correlation values are calculated directly in the frequency domain. The computer simulation results support the theoretical findings and verify the robustness of the proposed FBECLMS algorithm in the double-talk situation.

Key words: Signal processing for communication systems, Echo canceling, Adaptive digital filtering, LMS algorithm, Double-talk, Correlation function, Frequency domain.

1 – Introduction

In hand-free mobile radiotelephone or in tele-conference system, where we have acoustic echo feedback from loud speaker to microphone, the quality of communications is degraded severely.

Adaptive FIR filters by using the conventional LMS, BLMS, FDAF or FBAF algorithms (for instance the FBAF algorithm is defined by the first author of this paper, see [1]) are utilized for echo canceling. However, in the double-talk environment when both the near-end and the far-end signals are presented, the error signal used for tap adaptations will be uncorrelated with the echo signal and therefore, tap adaptations processes are severely damaged.

The conventional algorithm usually stops adaptation whenever double-talk sensor detects this condition. Stopping the tap adaptation is just a passive action to handle the double-talk condition and it causes lowering speed of adaptations and/or totally mislead when the echo path changed in the period of halting tap adaptation. Other works for challenging the problem of double-talk situation in the echo canceling can be found in [2], [3], and [4] that cause much more complexity adding to a simple LMS algorithm.

To solve the double-talk problem, correlation LMS (CLMS) algorithm and Extended CLMS (ECLMS) algorithm which utilize the correlation function of input signal instead of the input signal itself, have been proposed [7], [8], and [9]. 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.

To reduce the computational load of the ECLMS algorithm, the frequency domain version of the algorithm (FECLMS) has also previously been defined by the authors [10], [11]. Where, the frequency domain of the correlation function is obtained corresponding to the lag-time of correlation, not sampling time. However, still it is required to calculate the correlation function in the time domain which makes burden to the algorithm. Also, due to utilization of Short Time Fourier Transform (STFT), the exact linear convolution was not implemented. Therefore, the performance was not appropriate.

In this paper, we propose a new implementation of ECLMS algorithm in the frequency domain called frequency bin extended correlation LMS algorithm (FBECLMS), to overcome the previous problems. That is, we use the zero padding and Half Overlapped Save (HOS) method to implement the exact linear convolution in the frequency domain and also the correlation values are calculated directly in the frequency domain, therefore we do not need to make a large amount of calculations in time domain. While, the robustness of the algorithm in the double-talk is the same or even better since we are using the correlation processing and linear convolution.

The computer simulation results verify that the FBECLMS algorithm achieves a better result compared with FECLMS and FDAF [1] algorithms and show the robustness of the FBECLMS algorithm in the double-talk condition.

2 - Double-Talk and ECLMS Algorithm

In Fig.1, the echo canceling is shown. The output of the FIR filter, $\hat{y}(n)$, is estimating the echo signal, $y(n)$, by adjusting taps, *hi(n)*, to filter the input *x(n)*:

$$\hat{y}(n)=\sum_{i=0}^{N-1} h_i(n).x(n-i) \qquad (1)$$

The echo signal is obtained from echo impulse response, *r*, as follows (*N* is the acoustic impulse response length):

$$y(n)=\sum_{i=0}^{N-1} r_i.x(n-i) \qquad (2)$$

The error signal, *e(n)*, is calculated as below:

$$e(n)=d(n)-\hat{y}(n) \qquad (3)$$

where *d(n)*, is microphone signal that usually contains the echo signal. The LMS algorithm [5] is as follows:

$$h_i(n+1)=h_i(n)+2\mu_0.e(n)x(n-i) \qquad (4)$$

where μ_0, is the step size for tap coefficients adaptation. If the near-end signal, *s(n)*, is also presented during the echo canceling, then the microphone signal contains both the echo and the near-end signals.:

$$d(n)=y(n)+s(n) \qquad (5)$$

We call this condition as double-talk. It is well known that the error signal in this case contains uncorrelated component with input and echo signals . Therefore, the algorithm in (4) is failed to track the correct echo impulse response.

The ECLMS algorithm [9] is defined to handle this condition by computing the autocorrelation of the input as follows:

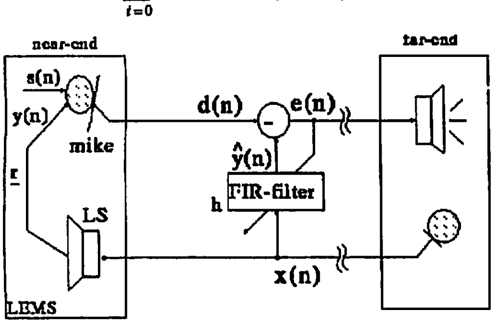
$$\varphi_{xx}(n,k)=\sum_{j=0}^{n} x(j).x(j-k) \qquad (6)$$

Also the cross-correlation between the desired and the input signal is calculated as follows:

$$\varphi_{dx}(n,k)=\sum_{j=0}^{n} d(j).x(j-k) \qquad (7)$$

Substituting from (2), (5) and (6) into (7) and assuming that there is no correlation between the far-end and the near-end signals [6] $\varphi_{sx}(n,k)\approx 0$, the cross-correlation will be obtained as follows:

$$\varphi_{dx}(n,k)=\sum_{i=0}^{N-1} r_i.\varphi_{xx}(n,|k-i|) \qquad (8)$$



LS:Loud Speaker
LEMS:Loudspeaker-Encrosure-Microphone-system

Fig.1. Echo canceler system

To estimate $\varphi_{dx}(n,k)$, we need to process the autocorrelation values of the input by an adaptive filter and find the error, *e(n,k)*, between the real and the estimated values of the cross-correlation.

$$\tilde{\varphi}_{dx}(n,k)=\sum_{i=0}^{N-1}\tilde{h}_i(n).\varphi_{xx}(n,|k-i|) \qquad (9)$$

$$e(n,k)=\varphi_{dx}(n,k)-\tilde{\varphi}_{dx}(n,k) \qquad (10)$$

The gradient for tap coefficients adjustment in the correlation processing filter is obtained (to minimize the MSE) as follows :

$$\nabla_j[MSE]=-2E[e(n,j).\varphi_{xx}(n,j)] \qquad (11)$$

and the ECLMS algorithm [9] is derived as below:

$$h_j(n+1)=h_j(n)+2\mu e(n,j)\varphi_{xx}(n,j) \qquad (12)$$

To achieve to an assure convergence and stability, we have to keep the step size μ in the following range:

$$0\langle\mu\langle 1/\lambda_{max} \qquad (13)$$

where: $\lambda_{max}\langle\sum_{i=0}^{N-1}\varphi^2_{xx}(n,i)$

and the normalized ECLMS algorithm [9] will be:

$$h_j(n+1)=h_j(n)+2\mu_0/(1+\|\bar{\varphi}_{xx}(n)\|^2)e(n,j)\varphi_{xx}(n,j)$$

with: $0\le\mu_0\le 1$ (14)

3 - Frequency Domain Algorithm

In Fig.2, direct implementation of the ECLMS algorithm is obtained in the frequency domain. This implementation is based on short-time Fourier analysis. Also, we need to calculate the correlation values in the time domain. This implementation which was introduced by authors in [10], [11] is called FECLMS.

The FECLMS algorithm has some drawback due to requirement of first calculating correlation values in the time domain and utilization of short-term Fourier analysis. The former makes the algorithm to have burden in computation of correlation and needs a time to converge to real value and the latter causes windowing effect and circular convolution. In the next part, we shall propose our new implementation of the ECLMS algorithm in the frequency domain. This implementation is based on zero padding and Half-Overlapped Saved (HOS) method to implement a real linear convolution. To this end, first we explain about the FDAF (frequency domain adaptive filtering) algorithm [1] which is based on realization of the linear convolution in the frequency domain.

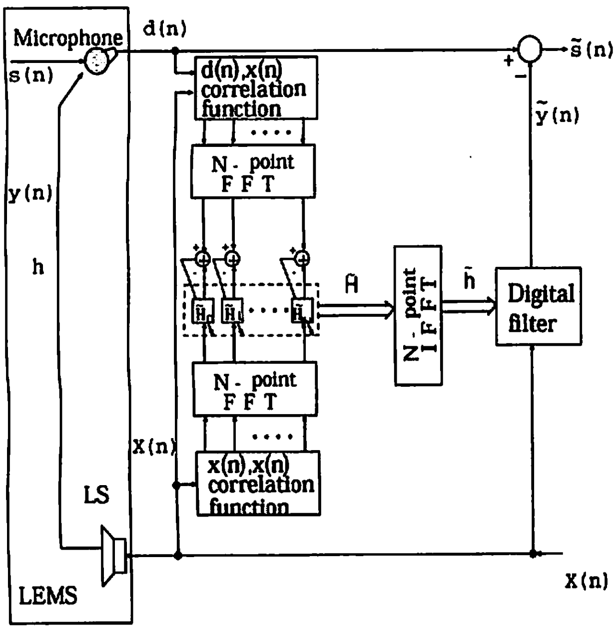


Fig. 2. Echo canceler using FECLMS algorithm

4 - FDAF Algorithm

The LMS algorithm can be implemented in the frequency domain which is called the FDAF algorithm [1]. The input vector $x(n)$ has $2N$ points length which is transform into the frequency domain by the FFT algorithm. Then the weight vector $H(k)$ are multiplied by the input signal to estimate the echo signal. Since we have used of zero padding of the time domain weight vector, the *IFFT* of the output gives the exact linear convolution only on the last N samples. So that, the error is calculated for only the last N samples. Then, by multiplications of complex conjugate of the input and the *FFT* of the error signal, we can implement the cross-correlations for tap adjustment. The cross-correlation must have also zero padding in the time domain. Therefore, we make this to be accomplished by IFFT/ zero padding/ FFT processes. The outcome will be used to adapt the tap coefficients in the frequency domain directly.

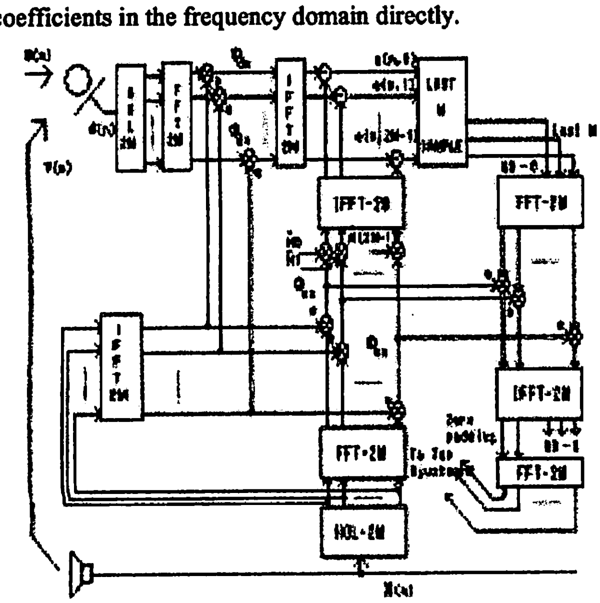


Fig.3. Construction of proposed FBECLMS algorithm

5 - FBECLMS Algorithm

To avoid the problems mentioned in the FECLMS algorithm, we propose a new frequency domain implementation of the ECLMS algorithm. First, we implement the autocorrelation and the cross-correlation directly in the frequency domain as below:

$$\Phi_{xx}(n,k)=\sum_{m=0}^{2N-1}\varphi_{xx}(n,m)W^{mk}=\sum_{m=0}^{2N-1}\Big[\sum_{j=n}^{n+2N-1}x(j)x(j-m)\Big]W^{mk}$$

$$=\sum_{j=n}^{n+2N-1}x(j)\sum_{m=0}^{2N-1}x(j-m)W^{mk}=X(k)\sum_{m=n}^{2N-1}x(j-m)W^{-(j-m)k}$$

$$\Phi_{xx}(n,k)=X(k)X^{*}(k) \tag{15}$$

by the same procedure, we can find the cross-correlation:

$$\Phi_{dx}(n,k)=D(k)X^{*}(k) \tag{16}$$

where $X(k)$ and $D(k)$ are $2N$ points *FFT* of the input and the desired response, respectively. These vectors of the input and the desired signal are Half-Overlapped data sequences. We also make zero padding for the tap coefficients in the time domain as like as the FDAF algorithm explained in the previous section. Also the error is only calculated for the last N samples which are corresponding to the correct linear convolution as so is it for HOS method. The rest of processes are the same as in the FDAF algorithm. That is for tap coefficients adaptation, we make the cross- correlation between the error and the input signal. Of course, here in our algorithm, the input signal is autocorrelation of the far-end signal. Next, we should make zero padding in the time domain to match the gradient part with zero-padding of the tap coefficients. This is done by IFFT/ zero-padding/ FFT processes. Fig. 3 shows the whole process of the new proposed implementation of the ECLMS algorithm in the frequency domain. We choose to name this method as Frequency Bin Extended Correlation LMS (FBECLMS) algorithm due to the basic similarity with the past FBAF algorithm [1] invented by the first author.

6 - Simulation Results

The acoustic echo impulse response, r_i, of the room is assumed to have exponential decaying shape that decreases to -60 dB after N samples as follows [1]:

$$r_i = Randn[\exp(-8i/N)] \tag{17}$$

To measure the performance of the convergence of the algorithm, we use the ratio of distance of weight and impulse response, Dw(n), which is defined as follows:

$$Dw(n)=10\log_{10}\Big[\sum_{i=0}^{N-1}\|r_i-\tilde{h}_i(n)\|^2 \Big/ \sum_{i=0}^{N-1}\|r_i\|^2\Big] \tag{18}$$

In order to show the capability and robustness of the frequency domain proposed FBECLMS algorithm, we have performed several computer simulations. In Fig. 4, the convergence characteristics for the frequency domain proposed FBECLMS algorithm is compared with those of FECLMS, CLMS, FDAF, and LMS algorithms. The double-talk condition is assumed for all algorithms. That is, the input (the far-end) is white noise and the near-end signal is assumed to be a color noise signal made by passing a white noise through a first order IIR filter. The echo impulse response has been selected to have 32 samples before it vanished to -60 dB. Therefore, the FFT size is 32 for FECLMS and 64 for FDAF and FBECLMS algorithms. We have run the simulations for 50 times and calculated the average over obtained data. The step sizes for all frequency domain algorithms are chosen to be 0.007 and for normalized LMS and CLMS algorithms is 1. As we see from Fig 4, The LMS and FDAF remain at -2 dB and CLMS reaches to -6 dB while FECLMS converge to -10 dB and even our new proposed algorithm, FBECLMS, achieves to -13 dB which is the best among all algorithms. In Fig. 5, weakness of FDAF algorithm is shown when double-talk condition is imposed to single-talk. That is, the FDAF works well in single–talk, but fails to do the job in the double-talk. The average was obtained over 100 runs and the step size is 0.01. Other condition is the same as in the previous simulation. In the next simulation in Fig. 6, we have started with FDAF and in the single-talk condition. Then, at 192-nd iteration, we have changed to double-talk condition. The simulation shows that if we continue with the FDAF and remain step size unchanged at 0.007, then, it will return to -2 dB from a converged value of -12 dB. This effect is due to existence of uncorrelated double-talk signal in error signal. That is the echo path can not be estimated correctly. If we detect the double-talk and freeze the step size (zero step size), then of course, it will remain at -12 dB. However, in this period there is no ability of echo path estimation if it would be changed. Now, if we select the proposed FBECLMS algorithm, even with non-freezing step size (0.007), the echo canceler converges to a better value even in the double-talk. This simulation clearly approves the robustness of our algorithm in the double-talk. Here, we have used the same parameters as in the first simulation, that is, input is white noise and near-end is color noise. In the last simulation, we start with FDAF algorithm and we impose the double-talk condition using white and color noise. At 10000-iteration, we switch the algorithm to FBECLMS. The step size is 0.007 for both algorithms. As it is shown in Fig. 7, at the beginning when we use FDAF algorithm, the enough convergence is not achieved, but after 10000-iteration when we use FBECLMS algorithm, the echo canceler converges to -15 dB. Thanks to robustness of the proposed algorithm in double-talk.

7 - Conclusion

A new implementation of the extended correlation LMS algorithm in the frequency domain was proposed. The frequency domain is obtained based on using the fast Fourier transform where in its kernel the lag time of the correlation function was used as the time variable. Therefore, the correlation values could be calculated in the frequency domain directly. The zero padding and half overlapped saved method are utilized in order to implement the linear convolution. An improvement of more than 2 dB has been obtained in performance of the proposed algorithm over the past algorithm. Also, a 15 dB convergence has been obtained as compared with other conventional algorithm which totally does not converge in double-talk. Therefore, the proposed FBECLMS algorithm is more robust than previously proposed ECLMS, FECLMS, and FDAF algorithms. The robustness of the proposed algorithm for the echo canceling in double-talk situation was shown by computer simulation.

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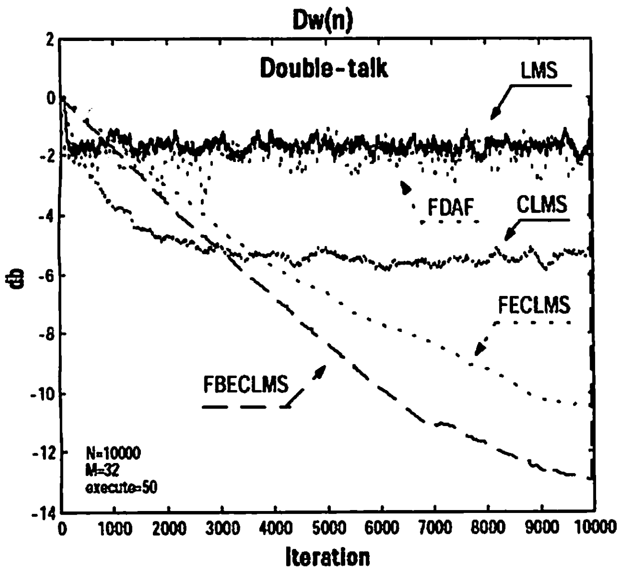

Fig. 4. Comparison between LMS, CLMS, FDAF,FECLMS, and proposed FBECLMS in double-talk.

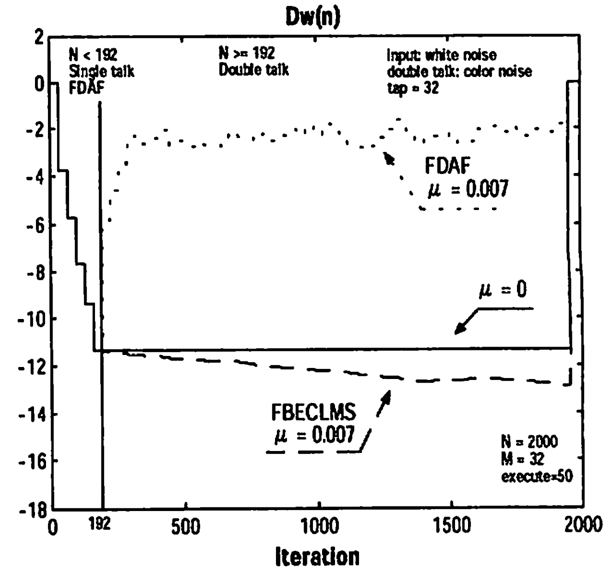

Fig.6. Switching from single to double talk and comparison of various performances.

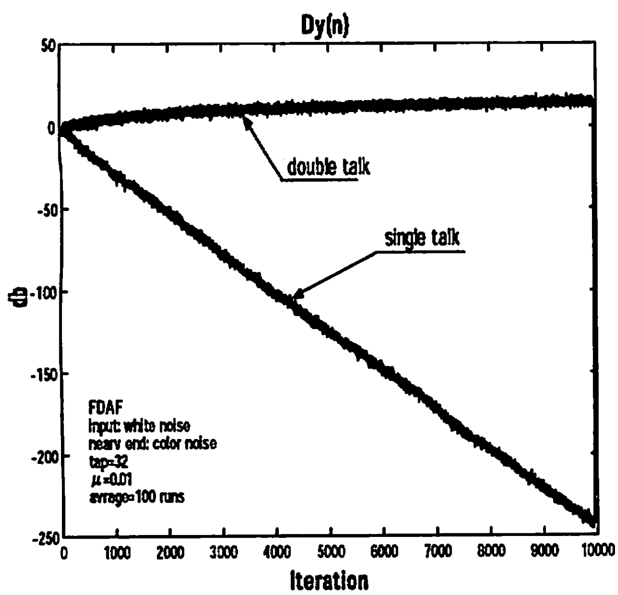

Fig.5. Convergence characteristics of FDAF algorithm in single and double-talk conditions.

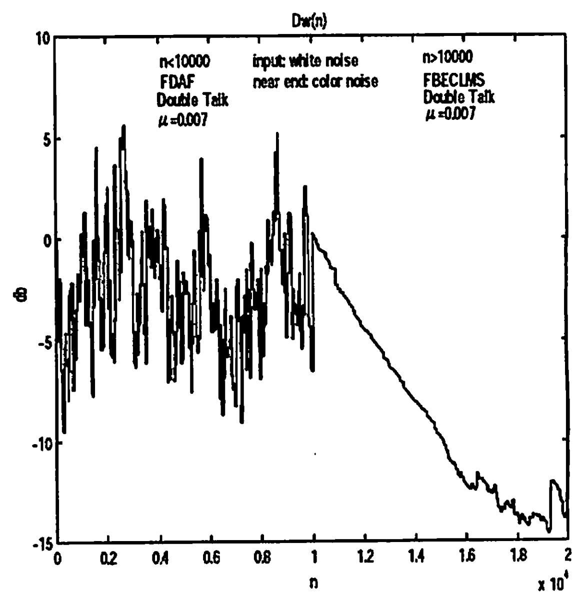

Fig.7. Switching from FDAF to FBECLMS under double talk condition.