Development of Adaptive Noise Cancelling Algorithm for Post Processing of Biomedical Signals

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Abstract

Biomedical signals are ubiquitously contaminated and degraded by background noise which spans nearly all frequency bandwidths. This paper proposes the MADF (multiplication free adaptive digital filter) algorithm to cancel the noise. And the convergence characteristics of the algorithm is analyzed. In the experimental results, the MADF algorithm has the advantage in which has superior to a condition of low-frequency and slow data speed. This application gives an important significance in ensuring the objectivity of clinical information and in promoting the representation and the disease diagnosis.

1. Introduction

Biomedical signals are ubiquitously contaminated and degraded by background noise which spans nearly all frequency bandwidths. In order to cancel these noises and preserve the desired signal, the adaptive filtering techniques have used.

The adaptive approach based on the filtering of background noise adjusts its parameters for each new available sample. The adaptive noise canceling (ANC) method based on the Wiener theory exploits adaptive optimal filtering concepts proven to be useful in many signal processing applications[1]. The goal of the ANC algorithm is to cancel background noise from the main signal, which is composed of the desired signal and background noise that has been correlated with noise from a reference measurement. The technique therefore relies upon access to a reference signal, located at the source of noise fields, as well as the main or primary signal[2].

In 1965, Widrow and Hoff developed adaptive algorithms including least mean squares(LMS), and was realized it to the adaptive noise cancelling system[3]. After this date, the ANC algorithm was successfully employed in many signal processing, seismic, and biomedical areas.

An alternative approach is the recursive least-square (RLS) method. The RLS algorithm has been widely used in real-time system identification applications and fast start-up channel equalization because of its fast convergence rate and stable filter characteristics. However the RLS algorithm may be computationally costly for some application since it requires M^2 operation per time update[4,5].

One of modified LMS algorithm is the Sign algorithm, which can reduce the multiplication operation and obtain the stabilized effects in comparing with another algorithm.

After this date, many efforts to reduce the multiplication number have been by Lee and Un, they proposed the MADF(multiplication free adaptive digital filter) algorithm using the LMS and DPCM(Differential Pulse Code Modulation) method[6]. Also, Park, Youn and Cha[7] were experimented the MADF algorithm using the Sign and DPCM[8].

In this paper, the MADF algorithm to cancel the background noise in biomedical signals is exploited. Also, we analyze the convergence characteristics of MADF algorithm and present it has the performance stabilized

2. Biomedical applications

2.1 The ANC method to enhance ECG monitoring

The adaptive noise cancellation(ANC) technology for the elimination of interference from desired biomedical signals have applied to an electrosurgical unit(ESU) and the detection of desired ECG(electrocardiogram) signals[9].

An ESU, a device widely used in cutting tissue and coagulating severed blood vessels, produces a radiofrequency(RF) signal modulated at 120 Hz. The ECG electrodes used to record the ECG signals can pick up the large ESU voltage present at the patient's skin surface. This nonstationary interference can bury the desired ECG signals by producing a signal-to-noise ratio of approximately. The fig. 1 shows the configuration of an ECU interference rejection system.

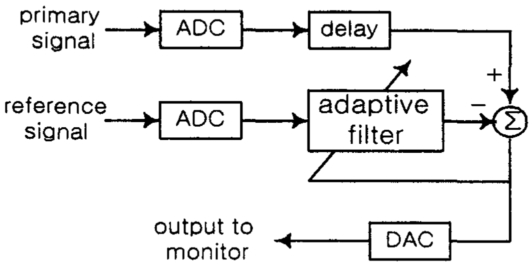

Fig. 1 ECU interference rejection system

The work of Yelderman et al.[9] suggests that processing of the biomedical signals before adaptive filtering may be necessary to eliminate high-frequency interference noise.

In practice, fetal heart rate and number of fetus are detected by recording abdominal electrocardiograms during labor and delivery[10]. However, abdominal ECG are contaminated with background noise due to muscular activity and fetal motion. The adaptive filter has been used to eliminate the background noise and enhance the fetal ECG[2]. The input to the adaptive canceler, consisting of maternal and fetal heartbeats, is recorded from the mother's abdomen. The four electrodes located at the mother's chest are used to record the mother's ECG. Signals from these ECG electrodes are employed as reference inputs to the ANC on the basis of the LMS(least mean square) algorithm.

The LMS algorithm is as follow equation (1).

$$\text{LMS}: H(n+1) = H(n) + \mu\, x(n)\, e(n) \qquad (1)$$

The coefficient equations of the modified LMS algorithms with the additive Sign structure are derived[5].

$$\text{Sign-LMS}: H(n+1) = H(n) + \mu\, x(n)\, \mathrm{sgn}[e(n)] \qquad (2)$$

$$\text{Sign-data LMS}: H(n+1) = H(n) + \mu e(n) \mathrm{sgn}[\, x(n)\,] \qquad (3)$$

where the μ is the convergence constants, the sgn[e(n)] is the 1 during $e(n) \geq 0$ and the -1 during $e(n) < 0$.

But, in order to assure the objectivity of biomedical signal, there must be groped for new methods with analysis and clinical diagnosis. Although the signal is nonstationary, the adaptive digital filter is designed by optimal coefficients in time varying. However the LMS algorithm is inappropriate for the fast-varying signals due to its slow convergence property.

2.2 The ANC method with the MADF algorithm

The MADF algorithm for canceling the noise is proper for highly correlative input signal. In the MADF structure of fig.2, the DPCM is used to the reference input of the Sign algorithm. The reference inputs signal is used to the update equation of adaptive filter coefficients. Also, the $\hat{X}(n)$ is the predicted inputs vector of the DPCM, the $\overline{X}(n)$ is the reconstructed input vector and the B(n) is the outputs vector.

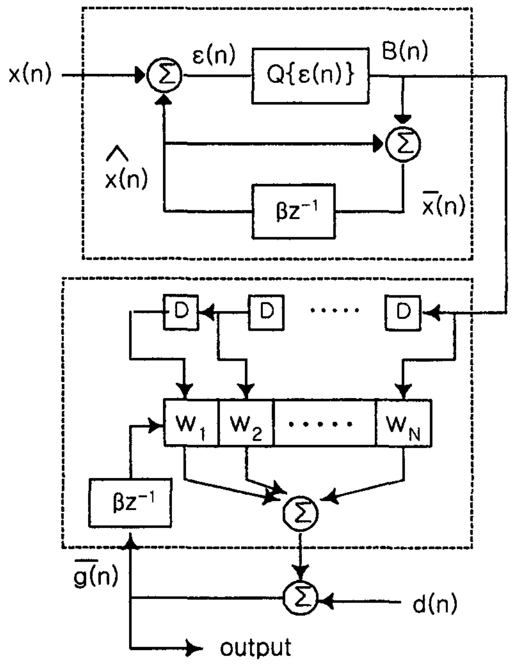

Fig. 2 The MADF structure

In Fig. 2, the predicted input $\hat{X}(n)$ is presented by the 1_{st} coefficients $\beta(0 < \beta < 1)$ predicted of the DPCM and $\overline{X}(n)$. In order to obtain the update equation for the coefficients of an adaptive filter H(n), we use the f(n), g(n), $\overline{g}(n)$ and e(n)[5, 6].

In order to analyze out the performance, we define the autocorrelation of the signal X(n) and $\hat{X}(n)$, and the cross-correlation of the signal d(n) and X(n) . Let the input signal assume wide-sense stationary,.the zero mean of the Gaussian noise, and the input pair {d(n), X(n) }, n $\neq$ k, is independent, then we obtain the covariance for the error vector of the DPCM.

When the input signal is zero-mean and Gaussian random process, the mean square error of LMS is as follows[7] ;

$$\sigma_e^2(\infty) \approx \zeta_{\min} + \frac{\mu}{2}\zeta_{\min}[tr(R_{xx})] \quad (4)$$

Where $\zeta_{\min} = E\{e^2_{\min}(n)\}$.

Assuming the small μ of Sign algorithm, the mean square error of the Sign algorithm is

$$\sigma_e^2(\infty) \approx \zeta_{\min} + \frac{\mu}{2}\sqrt{\frac{\pi\zeta_{\min}}{2}}[tr(R_{xx})] \quad (5)$$

The Sign algorithm in a identical stationary state have slowly converged in comparing the LMS, but in the case of small μ, there is fast converge.

In compare with equation (4) and (5), the estimated error in stationary state is more largely. They have the follow relations.

$$\mu_{Sign} = \mu_{LMS}\sqrt{\frac{2\xi_{\min}}{\pi}} \quad (6)$$

So, the modified Sign algorithm is the MADF structure, and their coefficient updating equation is as follows;

$$H(n+1) = H(n) + \mu\overline{X}(n)\operatorname{sgn}\{e(n)\} \quad (7)$$

Then the mean square error of the MADF algorithm is

$$\sigma_e^2(\infty) \approx \zeta_{\min} + \frac{\mu}{2}\sqrt{\frac{\pi\zeta_{\min}}{2}}[tr(R_{xx}) + N\sigma_e^2] \quad (8)$$

Comparing the equations (5) and (8), the stationary error estimated of the MADF algorithm is the more large. But if we can reduce the noise quantized, the difference of two algorithms is very small.

Another method to cancel the noise, we use the fractionally spaced equalizer(FSE). Fig. 3 shows the structure of fractionally spaced equalizer.

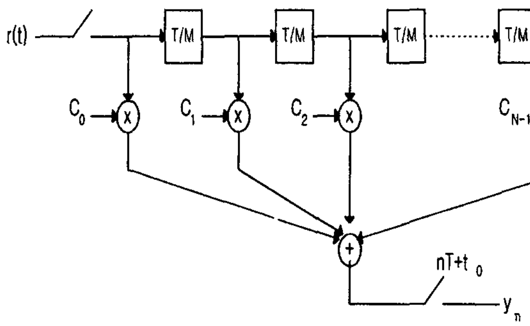

Fig. 3 The FSE structure

The signal sampled in kT + t_0 is as follows.

$$r(kT+t_0) = a_k + \sum_{i\neq k} a_i f(kT+t_0-iT) + N(kT+t_0) \quad (9)$$

where the a_i is a informative signal, the f(t) is the impulse response of a channel and the N(t) is the white Gaussian noise. The a_k is the original signal and the 2_{nd} term is the ISI term. The FSE structure can compensate for delay distortion occurring in ISI. Then the output Z_k is calculated by filter coefficient h_i.

$$Z_k = \sum_{i=0}^{N} h_i r(kT+t_0-iT) \quad (10)$$

The N is the order of an equalizer in equation (10), the h_i is the coefficients. The delay tap of the FSE in implementing the digital filter became τ = kT/M [6, 7].

3. Experimental results

Table 1 compares the operation number of each algorithm. Where we can remove the multiplier, but the addition increase. So, in additive equipment, it is necessary for the A/D converter and memory element including the M resolution.

Table 1. Comparison of each algorithm

Algorithm	multiplication	Addition
LMS	2N	2N
Sign	N	2N
MADF	0	3N + 5

In order to experiment the algorithm proposed, the reference signal d(n) make use of the channel input signal s(n) having the 12-tap delayed. The quantization step of the MADF is the 13 and applicable for the 1_{st} predictor.
The sample number of the input signal are 4000, we estimate the total average of the iterative experiments for 70 number.

The Fig. 4 presents the convergence characteristics of the Sign and MADF algorithm that have the SNR of the 30 dB, $w = 2.9$, $\beta = 0.5$, $\Delta = 0.25$ and $\mu = 0.008$.

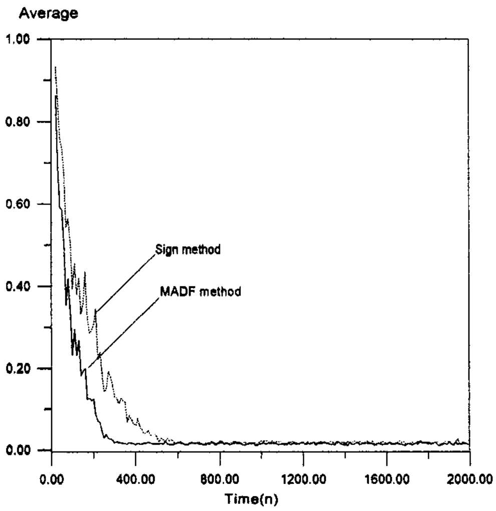

Fig. 4 Comparison of MADF and Sign algorithm

The performance of each algorithm improves the convergence characteristics in case of increasing the SNR. But the MADF algorithm has no multiplier and the computational number reduced. So, this algorithm shows the convergence characteristics stabilized applying for fractionally spaced equalizer.

4. Conclusion

In order to cancel the background noise in biomedical signals, the MADF algorithm is proposed and analyzed the convergence characteristics of the one. In the experimental results, the MADF algorithm has no multiplier and the computational number reduced. So, this algorithm shows the convergence characteristics stabilized applying for fractionally spaced equalizer

Reference

[1] Wiener, Extrapolation and Smoothing of Stationary Time Series and Engineering Applications, Wiely, N, Y. 1949

[2] Widrow. B. et all, “Adaptive Noise Cancelling : Principles and Applications,” Proc. Of IEEE 63(12), pp. 1692-1716, 1975

[3] B.Widrow and S.D.Stearns, Adaptive Signal processing, Englewood Cliffs, NJ: Prentice-Hall, 1985.

[4] J.W.Lee, C.K.Un, and J.C.Lee, "Adaptive digital filtering of differentially coded signals," Proc. of the IEEE ICASSP., Tampa, FL, PP.1257-1260, March 1985.

[5] V. J. Matheuws, Sung Ho Cho, "Improved Convergence Analysis of Stochastic Gradient Adaptive Filters Using the Sign Algorithm" IEEE Trans. on ASSP, Vol. 35, No. 4, April, 1987

[6] V. J. Mathews, "An Efficient FIR Adaptive Filter Using DPCM and the Sign Algorithm", IEEE Trans. on ASSP, Vol. 37, No. 1 Jan. 1989.

[7] T.H. Park, D. H. Youn and I. W. Cha, "Multiplication-Free Adaptive Filter," *Proc. of the IEEE*, Vol.76, No.5, pp.632-634, May,1988.

[8] Sung Ho Cho, "Convergence Analysis for Efficient Adaptive Digital Filtering Algorithm and Structures," A dissertion submitted to the University of Utah, Aug. 1989

[9] Yelderman. M., Widrow, B., Cioffi., J. M. Hesler and Leddy, “ECG Enhancement by Adaptive Noise Cancellation of Electrosurgical Interference,” IEEE Trans. Biomed. Eng. BME7 : 392-398, 1983

[10] Roche, J. Hsu, and Barter, “The fetal Electrocardiography,” Obstet. Gynecol. 92 : 1149-1159, 1963