34th Annual International Conference of the IEEE EMBS
San Diego, California USA, 28 August - 1 September, 2012

A Modified Log-LMS Adaptive Filter with Low Signal Distortion for Biomedical Applications

Yuzhong Jiao, Rex Y. P. Cheung and Mark P. C. Mok

Abstract— **Life signals from human body, e.g. heartbeat or electrocardiography (ECG), are usually weak and susceptible to external noise and interference. Adaptive filter is a good tool to reduce the influence of ambient noise/interference on the life signals. Least mean squares (LMS) algorithm, as one of most popular adaptive algorithms for active noise cancellation (ANC) by adaptive filtering, has the advantage of easy implementation. In order to further decrease the complexity of LMS algorithm based adaptive filter, a Log-LMS algorithm was proposed, which quantized signals by the function of $\log_2$. The algorithm can replace multipliers by simple shifting. However, both LMS algorithm and Log-LMS algorithm have the disadvantage of serious signal distortion in biomedical applications. In this paper, a modified Log-LMS algorithm is presented, which divides the convergence process into two different stages, and utilizes different quantization method in each stage. Two scenarios of biomedical applications are used for analysis, 1) using stethoscope in emergence medical helicopter and 2) measuring ECG under power line interference. The simulated results show that the modified algorithm can achieve fast convergence and low signal distortion in processing periodic life signals.**

I. Introduction

Biomedical signals generated from human body are often very weak so as to be easily covered by background noise. For example, on the emergency medical helicopter, doctors can hardly hear the heartbeat and respiratory sound of patient but blade flapping noise. Another example is that 50 (or 60) Hz power line interference brings serious influence to electrocardiography (ECG) signal diagnosis and analysis. Common filters are hard to remove the ambient noises without losing useful information because of the spectrum overlapping of biomedical signals and noises [1-2]. Adaptive filter can reduce noises and interferences with the similar spectrum as life signals [1, 3-5]. In numerous adaptive algorithms [6], least mean squares (LMS) algorithm is the most popular and widely used one mainly due to its computational simplicity. In order to further decrease complexity, a Log-LMS adaptive filter was presented [7-10]. The method can reduce the implementation complexity by quantizing signals by the function of $\log_2$ and thus replacing multipliers by shifters. Moreover, it maintains the fast convergence of LMS algorithm. However, similar to LMS algorithm, the Log-LMS algorithm is often suffered from large signal distortion.

* Research supported by Hong Kong Innovation and Technology Fund.

Yuzhong Jiao is with the Hong Kong Applied Science and Technology Research Institute (ASTRI), Hong Kong, China (e-mail: yzjiao@astri.org).

Rex Y. P. Cheung is with the Hong Kong Applied Science and Technology Research Institute (ASTRI), Hong Kong, China (e-mail: rexcheung@astri.org).

Mark P. C. Mok is with the Hong Kong Applied Science and Technology Research Institute (ASTRI), Hong Kong, China (phone: +852-3406-2758; fax: +852-3406-2801; e-mail: markmok@astri.org).

In adaptive filter, output signal is fed back to control the updating of filter coefficients. The way to process the feedback signal can directly affect the characteristics of an adaptive filtering system such as convergence rate, signal integrity and implementation complexity. In this paper, a modified Log-LMS algorithm is presented. It not only keeps the advantages of Log-LMS algorithm including low complexity and fast convergence rate, but also reduces signal distortion. The rest of the paper is organized as follows: Section 2 briefly introduces LMS algorithm and Log-LMS algorithm. Modified Log-LMS algorithm will be described in Section 3. In Section 4, the simulation results of traditional Log-LMS algorithm and modified Log-LMS algorithm in the stethoscope and ECG applications will be presented. Section 5 will further discuss the performances of the two algorithms. A conclusion will be given in the last section.

II. Adaptive Algorithms

The block diagram of adaptive noise cancellation (ANC) system using adaptive filtering is illustrated in Fig. 1 [6]. In this system, the desired response (or primary input) d of the adaptive filter shown in the dashed square is the combination of a source signal s and a noise n_0. The reference input x is equal to noise n_1. n_0 is derived from n_1 after passing an unknown system. n_1 is highly correlated with n_0 as they are derived from the same noise source. The reference input x is filtered to produce an output y. This output is then subtracted from the primary input to produce the system output e, which equals to $s + n_0 - y$. The goal of adaptive algorithm is to estimate the transmission characteristics of the unknown system which n_1 passes. After convergence, the weight vector of digital filter should be close to that of the unknown system, then y becomes a close replica of n_0, and then the system output e approximately equals to the source signal s.

A. *LMS algorithm*

The LMS algorithm is described by the equation [6]:

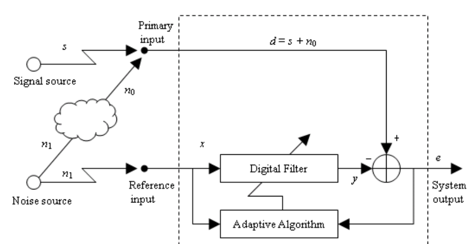

Figure 1. Block diagram of active noise cancellation system

$$W(n+1) = W(n) + \mu(n)e(n)X(n). \tag{1}$$

Where $W(n)$ and $X(n)$ are the weight vector and the input vector of adaptive filter respectively, at time n. $\mu(n)$ and $e(n)$ are the step size and the system output respectively, at time n. The condition of the algorithm achieving convergence is $0 < \mu(n) < 1/\lambda_{\max}$, where $\lambda_{\max}$ is the maximal eigenvalue of the input signal autocorrelation matrix.

B. Log-LMS algorithm

The log-LMS algorithm is described as below [9]:

$$W(n+1) = W(n) + \mu(n)Q(e(n))X(n). \tag{2}$$

Where $Q(\cdot)$ means logarithmic quantizer which simply converts its input into the power-of-two value. The quantization operation is defined by:

$$Q(z) = 2^{\left\lfloor \log_2(|z|) \right\rfloor} \operatorname{sgn}(z). \tag{3}$$

Where $\lfloor v \rfloor$ is the largest integer less than v, $\operatorname{sgn}(\cdot)$ means the sign of a real number.

As for biomedical applications, the source signal s is usually a periodic life signal with large repeated peaks like QRS complexes of ECG. In LMS and Log-LMS algorithms, the system output e is directly used to control the updating of filter weight vector. Therefore, a steady state after convergence is easily destroyed when a sudden large peak in the system output e appears. The continuously enlarged difference in transmission characteristics (or weight vectors) between the unknown system and the digital filter shown in Fig. 1, will lead to serious signal distortion.

III. Modified Log-LMS Algorithm

In order to reduce signal distortion, the whole processing of adaptive filtering will be divided into two stages. The first stage is called converging stage, in which adaptive filter iterates with a fast convergence rate. After a primary convergence arrives, the system enters into the second stage, extracting stage. In this extracting stage, adaptive filter will iterate to extract the source signal with low distortion.

The iteration function of weight vector in the modified Log-LMS algorithm is the same as (2). But the quantization operation is changed, which is shown as follows:

$$Q(z) = \alpha \cdot 2^{\theta(z)} \operatorname{sgn}(z). \tag{4}$$

$$\theta(z) = \begin{cases} 0 & \text{when } \left| z/\alpha \right| < 1 \\ \left\lfloor \log_2(\left| z/\alpha \right|) \right\rfloor & \text{at converging stage.} \\ -\left\lfloor \log_2(\left| z/\alpha \right|) \right\rfloor & \text{at extracting stage} \end{cases} \tag{5}$$

Where α is a small power-of-two value and less than 1. The role of z/α is to amplify the processed signal to integer type by shift operation. At converging stage, the quantization operation is similar with that of Log-LMS algorithm, while at extracting stage, the processed signal is replaced with its reciprocal. Fig. 2 shows the relation of $|z/\alpha|$ and $\theta(z)$.

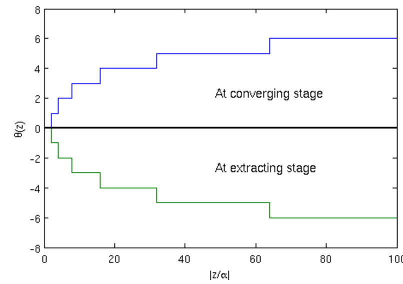

Figure 2. The relation of $|z/\alpha|$ and $\theta(z)$

In order to switch between the two different stages, a threshold ε related to the system output is introduced. When the system output becomes stable, which means $|e(n)-e(n-1)|$ is less than ε for a certain number of continuous iterations, we can determine that the converging stage is done and the iteration begins to enter into the extracting stage.

IV. Simulations

Two cases are given in this section to fully compare the performance of traditional Log-LMS algorithm and modified Log-LMS algorithm in biomedical applications. One case is to extract the heartbeat of patient by using a stethoscope on emergency medical helicopter. The other one is to extract ECG signal from 50 (or 60) Hz power line interference. For a fair comparison, we will firstly ensure the signal integrity of system output, and then investigate which algorithm converges quicker. This is because it is meaningless to merely compare convergence rate without a consideration of signal distortion.

A. Using Stethoscope on Emergency Medical Helicopter

Conventional passive acoustic stethoscopes are ineffective in noisy environments that exceed levels of 80 dB. However, the power level of helicopter sound is usually at 100~110 dB [10]. So a digital stethoscope must be capable of more than 30 dB gain in the extreme environment.

There are usually two microphones installed in the digital stethoscope with active noise cancellation. One microphone is placed outside of stethoscope head to capture the external noise, which is mainly composed of helicopter sound. The other is installed inside of the stethoscope head to receive the acoustic signal of heartbeat and internal noise. The unknown system, which external helicopter sound passes to form the internal noise, is modeled by a 100~1,000 Hz bandpass filter. The running helicopter sound comes from a wav file sampled at 11,025 Hz. The two algorithms for comparison are traditional Log-LMS algorithm with $\mu = 2^{-6}$ and our modified algorithm with $\mu = 2^{-4}$, $\alpha = 2^{-12}$, and $\varepsilon = 2^{-10}$. In order to obtain a good heartbeat signal, a smaller step size is used in the traditional algorithm. The orders of digital filters for both algorithms are 64.

Fig. 3 shows the original heartbeat signal of a patient and the simulation results of two algorithms in the scenario of using stethoscope on emergency medical helicopter.

Compared with traditional Log-LMS algorithm, our modified algorithm can produce an output signal with lower distortion. The heartbeat signal after traditional Log-LMS algorithm is clearly distorted. If the same step size is used in the traditional algorithm, the signal distortion would be larger. A small green spot in Fig. 3(c) shows the switching line from converging stage to extracting stage.

B. ECG Measurement under Power Line Interference

ECG is the recording of heart's electrical potential versus time. The measurement of ECG is often interfered by 50 (or 60) Hz power line noise. Cables carrying ECG signals from the patients to the monitoring equipment are susceptible to electromagnetic interference from the 50 (or 60) Hz power line noise.

In the adaptive power line interference canceling system, the primary input is a mixed signal including ECG signal and 50 Hz power line interference. The ECG signal used in the simulation is extracted from the Record 100 of the MIT-BIH Arrhythmia database [11]. Its sampling frequency is 360 Hz. The reference input of adaptive filter is sine signal with the same frequency. The Log-LMS algorithm is set with $\mu = 2^{-5}$ and our modified algorithm is set with $\mu = 2^{-4}$, $\alpha = 2^{-8}$, and $\varepsilon = 2^{-3}$. Similarly, a smaller step size is used in the Log-LMS algorithm. The orders of digital filters for both algorithms are 10. A preprocess for removing the DC component from the ECG signal is needed before adaptive interference cancelling.

Fig. 4 shows the ECG signal corrupted by the 50 Hz power line interference and the simulation results of two algorithms. It is not hard to see that our algorithm can achieve better signal integrity. The green spot in Fig. 4(c) illustrates the switching point from converging stage to extracting stage.

V. DISCUSSIONS

In order to further analyze the performance of the modified algorithm, the key characteristics including signal integrity, convergence rate and implementation complexity will be compared between traditional Log-LMS algorithm and the modified algorithm.

A. Signal Integrity

In simulation, residual noise can be obtained by subtracting the source signal s from the system output e. Residual noise reflects the signal integrity of the system output. The residual noise is mainly from two sources. One is the residual part of ambient noise/interference. The other one is signal distortion which is generated by adaptive filter.

Fig. 5 and 6 give the residual noises of two algorithms in the applications of using stethoscope on emergency medical helicopter and ECG measurement under power line interference. The signals in the durations from 1.6s to 1.8s, and from 3.0s to 4.0s are used in the applications of stethoscope and ECG separately. From the figures, we can conclude that our modified algorithm is not only more capable of reducing ambient noises, but also more immune to the signal distortion. Tab. I lists the SNRs before and after

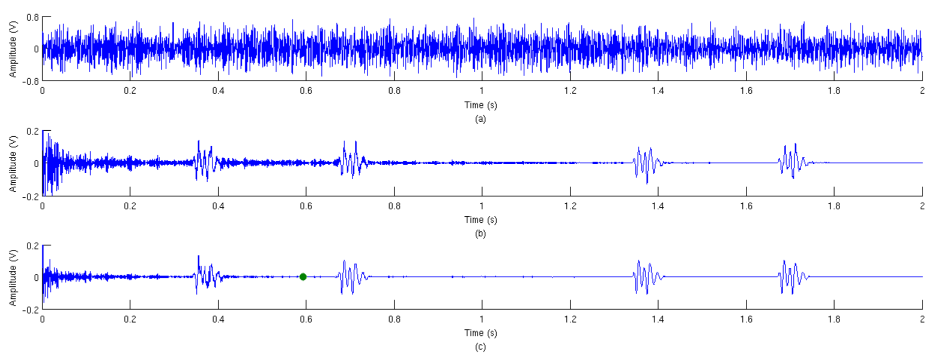

Figure 3. The simulation results of heartbeat after ANC, (a) original heartbeat signal corrupted by helicopter sound, (b) heartbeat signal after cancelling noise with traditional Log-LMS algorithm, (c) heartbeat signal with modified Log-LMS algorithm

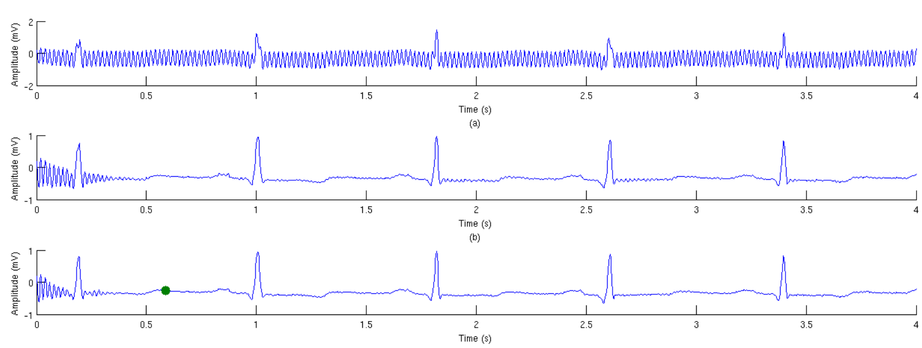

Figure 4. The simulation results of ECG after ANC, (a) original ECG signal corrupted by 50 Hz power line interference, (b) ECG signal after cancelling interference with traditional Log-LMS algorithm, (c) ECG signal with modified Log-LMS algorithm

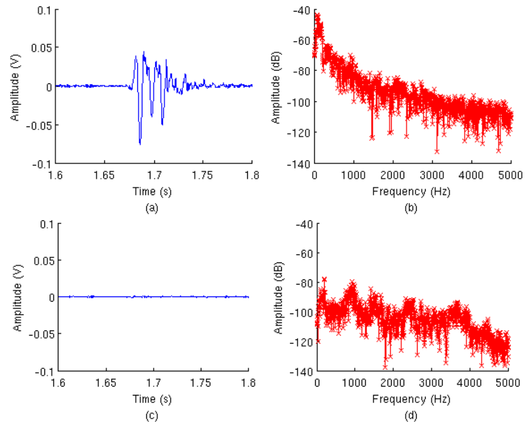

Figure 5. The residual noises in the stethoscope application: (a) and (c) are the time-domain residual noises after Log-LMS and our modified algorithms separately; (b) and (d) are their corresponding power spectra separately

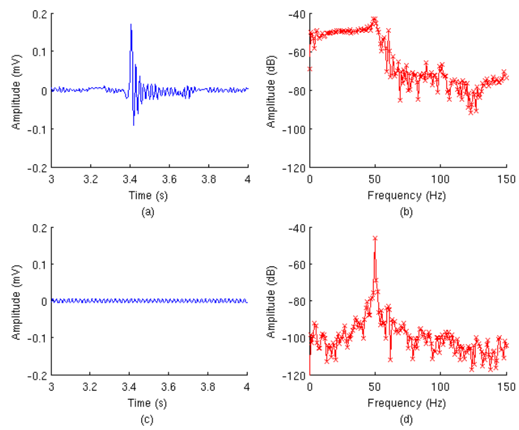

Figure 6. The residual noises in the ECG application: (a) and (c) are the time-domain residual noises after Log-LMS and our modified algorithms separately; (b) and (d) are their corresponding power spectra separately

TABLE I. COMPARISON OF SNR

Applications	**SNR (dB)**		
	Original	***After Log-LMS***	***After modified***
Stethoscope	-17.04	6.36	38.12
ECG	-1.19	33.28	40.15

adaptive filtering in the two applications. It is clear that modified Log-LMS algorithm can achieve larger gains than traditional Log-LMS algorithm.

B. Convergence Rate

Based on simulation result, our modified algorithm can achieve faster convergence rate than traditional Log-LMS algorithm on the premise of ensuring signal quality. This is because our modified algorithm can use larger step size in the whole convergence process.

C. Complexity

As two stages are involved, modified Log-LMS algorithm inevitably requires higher implementation complexity than traditional Log-LMS algorithm. Moreover, the modified algorithm is required to amplify the processed signal to integer type and shorten the signal after processing, these requirements also increases the complexity. However, the increased processing components are limited and easy to implement. The simple components include shifter, comparator and so on. More importantly, it is worthy if system output signal can maintain a higher signal-to-noise ratio (SNR).

VI. CONCLUSION

In this paper, a modified Log-LMS algorithm is used in adaptive noise or interference cancelling system for biomedical applications with periodic life signals. The whole convergence process is divided into two stages: converging stage and extracting stage. At each stage, different quantization methods are adopted. The algorithm can guarantee fast convergence in the converging stage and low signal distortion in the extracting stage. Compared with traditional Log-LMS algorithm, the modified algorithm is only required additional control of stage change and simple shift operations. In the two biomedical applications, the modified algorithm shows better performances, especially lower signal distortion, than traditional Log-LMS algorithm.

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