HIGH FREQUENCY NOISE REDUCTION IN ECG SIGNAL BY ADAPTIVE FILTER USING FxLMS ALGORITHM

KHỬ NHIỄU CAO TẦN TRONG TÍN HIỆU ĐIỆN TIM BẰNG BỘ LỌC THÍCH NGHI SỬ DỤNG THUẬT TOÁN FxLMS

Pham Thi Quynh Trang[1,*], Duong Thi Hang[1]

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ABSTRACT

This paper proposes a solution using Filtered-x Least Mean Square (FxLMS) to remove high-frequency noise in electrocardiogram (ECG) signals. The simulation results show that the high frequency noise filtering using the proposed solution is better than the noise filtering using the traditional LMS technique in the same scenario.

Keywords: *Filter, LMS, FxLMS, ECG, noise reduction.*

TÓM TẮT

Bài báo đề xuất giải pháp sử dụng thuật toán bình phương sai số cực tiểu có bộ lọc (FxLMS) để loại bỏ nhiễu cao tần trong tín hiệu điện tim (ECG). Các kết quả mô phỏng cho thấy, sử dụng giải pháp đề xuất có khả năng lọc nhiễu cao tần tốt hơn khi so sánh với lọc nhiễu sử dụng kỹ thuật LMS truyền thống trong cùng một kịch bản.

Từ khóa: *Bộ lọc; LMS; FxLMS; ECG; khử nhiễu.*

[1]Faculty of Electronic Engineering, Hanoi University of Industry
[*]Email: pham.trang@haui.edu.vn
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1. INTRODUCTION

Today, the necessary in the information transmission systems are ensure that the signal is transmitted and received correctly. However, the signals from the transmitter to the receiver through different transmission environments which are affected by external objective factors such as temperature, humidity, and impurities in the environment… To ensure that information transmitted and received through the channel is effective and accurate, different solutions need to be implemented for signal processing. One of the solutions is to use digital filters, which are now often adaptive filters. An adaptive filter is a filter that has the ability to recalculate the weights to match the changes in the input or output signals in the system [1, 2].

Figure 1 shows the block diagram of an adaptive filter, the weights (W) of filter will be calculated based on the change of input and output signal.

An electrocardiogram (ECG) is a cyclic graph that records the electrical variations produced by the heart during contraction. An electrocardiogram (ECG) is a simple test that can be used to check your heart's rhythm and electrical activity. An ECG also is often used alongside other tests to help diagnose and monitor conditions affecting the heart.

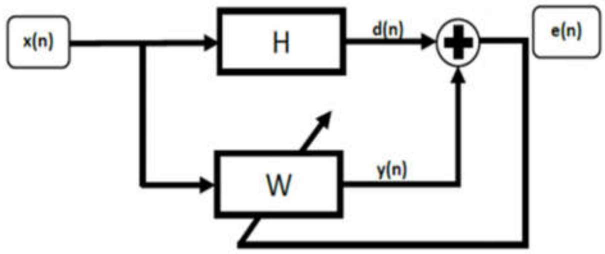

Figure 1. Block diagram of adaptive filter

There is a standard procedure for acquiring and analyzing ECG signals. A typical ECG wave of a healthy individual is as Figure 2.

The normal electrocardiogram of each heartbeat consists of 6 consecutive waves denoted by the 6 letters P, Q, R, S, T, and U. The electrical signal begins in the sinoatrial node which is located in the right atrium and travels to the right and left atria, causing them to contract and pump blood into the ventricles. This electrical signal is recorded as the P wave on the ECG. The PR Interval is the time, in seconds, from the beginning of the P wave to the beginning of the QRS complex. The electrical signal passes from the atria to the ventricles through the atrioventricular (AV) node. When the ventricles to fill with blood, the signal slows down and appears as a flat line on the ECG between the end of the P wave and the beginning of the Q wave. The PR segment represents the electrical conduction through the atria and the delay of the electrical impulse in the atrioventricular node.

The "QRS" complex is much larger than the "P" wave due to the relative difference in muscle mass of the atria and ventricles, which masks the relaxation of the atria. The relaxation of the ventricles can be seen in the form of the "T" wave [7, 8].

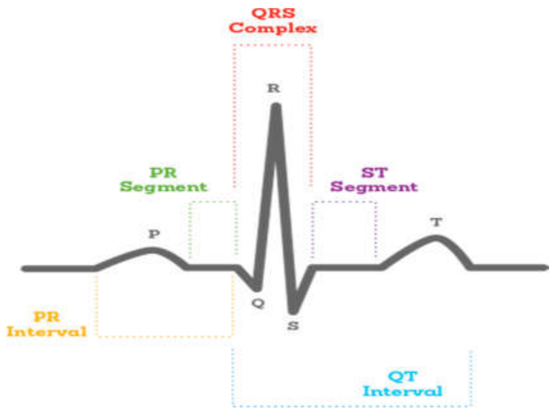

Figure 2. The ECG signal

The characteristics of the ECG signal are small amplitude, high frequency because this signal is measured directly on the human body, so it is often affected by different types of noise in the environment. The characteristics of the ECG signal are small amplitude, high frequency and measured directly on the human body, so it is often affected by different types of noise in the environment. Therefore, the problem of noise reduction in the ECG signal is always studied and improved to improve efficiency.

2. ADAPTIVE ALGORITHM

2.1. Least squares algorithm (LMS)

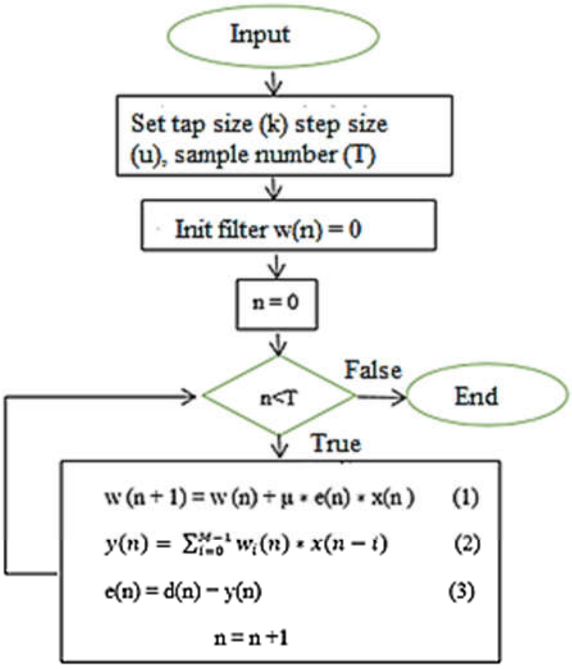

Figure 3. LMS algorithm flowchart

The least-squares algorithm is widely used in adaptive digital signal processing [3]. The LMS reference design consists of two main functional blocks - an FIR filter and the LMS algorithm. The FIR result is normalized to minimize saturation. The LMS algorithm iteratively updates the coefficient and feeds it to the FIR filter. The FIR filter uses the coefficient (W) with the input reference signal x(n) to generate the output y(n). The output y(n) is then subtracted from the desired signal d(n) to generate an error e(n), which is used by the LMS algorithm to compute the next set of coefficients [3].

The flowchart of the LMS adaptive filtering algorithm is shown in Figure 3.

Where: μ is step size; w(n) are coefficients; x(n) is input; e(n) is error at n - time [4, 6].

2.2. FxLMS Algorithm

The block diagram of the FxLMS filter is shown in Figure 4.

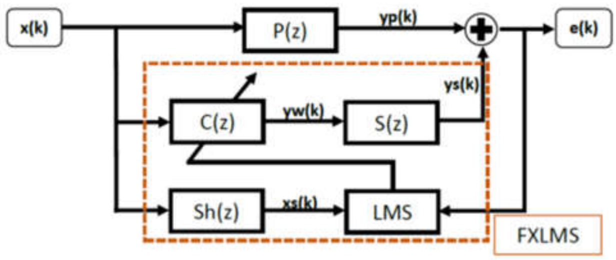

Figure 4. The block diagram of the FxLMS filter

Where:

x(k): noise signal.

xs(k) noise signal base on emulator parameter Sh(z) and S(z)

P(z): Primary path transfer function

yp(k) Primary noise signal at the error

e(k): Modified error signal

S(z): Secondary path transfer function

C(z) Controller for the FxLMS algorithm.

yw(k): output signal form secondary filter C(z)

ys(k): output signal form FxLMS filter

The FxLMS algorithm is a straightforward variant of the LMS algorithm, thus the reason of it being selected based on the minimization of mean square error condition. In this adaptive filter, there are two different control systems, which is feedback and feedforward which is distinguished by the reference signal x(n). The feedback uses reference signal based on the feedback predictor which can attenuate the components of the unwanted noise by predicting, thus although the feedback is less robust but it is more compact and cost effective compared to feedforward [4, 5].

Flowchart of FxLMS algorithm is described as Figure 5.

Where: ∇ is the gradient of w; μ is step size; w(n) are coefficients; x(n) is input signal; s(n) is the secondary path impulse response; e(n) is the error at n - time.

The stability of the FxLMS algorithm is highly dependent on the convergence power s(n); proportional to the μ parameter and this parameter is indirectly proportional to steady-state performance.

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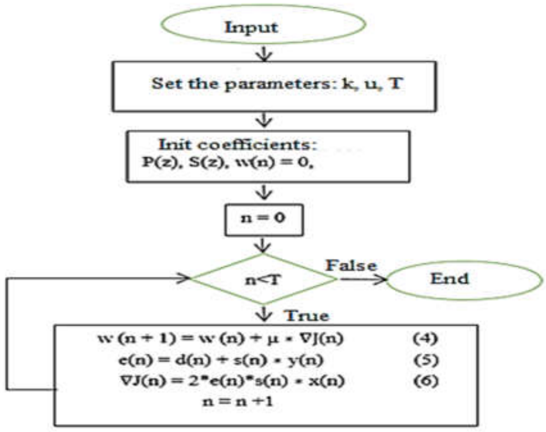

Figure 5. The FxLMS Algorithm

3. RESULT OF SIMULATION

3.1. Simulation scenario

This simulation scenario is performed on Matlab 2018a software, which was installed on Lenovo IdeaPad Intel core i3 computer with 2GHz frequency, 6GB RAM.

The program simulates the process of filtering the high-frequency noise of the ECG signal. The input signal is an electrocardiogram signal with noise that is passed through two adaptive filters LMS and FxLMS with the following parameters in Table 1.

Table 1. The initialization parameter for simulation

	LMS filter	FxLMS filter
Tap size	k = 16	k = 16
Step size	μ = 0.1	μ = 0.1
Pz	None	[0.01 0.25 0.5 1 0.5 0.25 0.01];
Sz	None	Pz*0.25

3.2. Simulation result

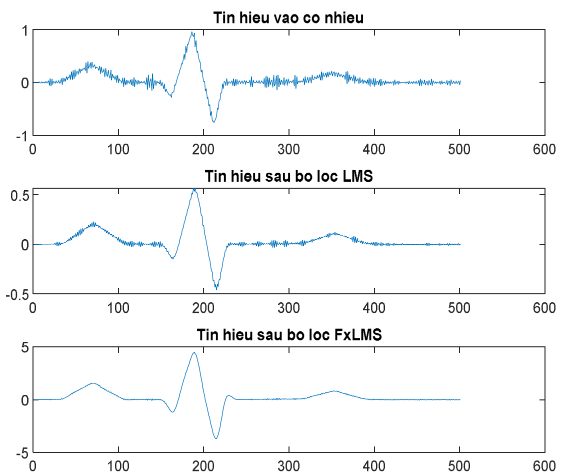

Figure 6. The input and output signal from LMS and FxLMS filters

The results of the input signal and the output signal after the filter are shown in Figure 6, this result shows that the output signal after the FxLMS filter separates noise cleaner than the output signal after the LMS filter.

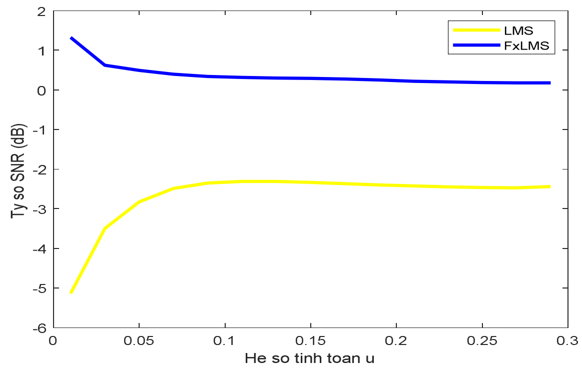

Figure 7. The SNR ratios of output signal from LMS and FxLMS filters

Figure 7 shows the relationship between the filter step parameter μ and the signal-to-noise ratio (SNR) of the output signal. The simulation results show that when μ is between 0.01 and 0.1, the SNR ratio of the output signal after the FxLMS filter is higher than that of the output signal after the traditional LMS filter.

4. CONCLUSION

The paper presented two adaptive filtering algorithms which are traditional LMS and improved FxLMS. Further more, the paper also proposed to use the adaptive FxLMS filter in reduce high frequency noise in the electrocardiogram signal. The simulation results show that the filter using the FxLMS algorithm has a more complex structure than the LMS filter, but it is more effective in filtering high-frequency noise.

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THÔNG TIN TÁC GIẢ

Phạm Thị Quỳnh Trang, Dương Thị Hằng

Khoa Điện tử, Trường Đại học Công nghiệp Hà Nội