Performance Evaluation Of Different Adaptive Filters For ECG Signal Processing

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***Abstract*— One of the main problem in biomedical data processing like electrocardiography is the separation of the wanted signal from noises caused by power line interference, external electromagnetic fields and random body movements and respiration. Different types of digital filters are used to remove signal components from unwanted frequency ranges. It is difficult to apply filters with fixed coefficients to reduce Biomedical Signal noises, because human behavior is not exact known depending on the time. Adaptive filter technique is required to overcome this problem. In this paper two types of adaptive filters are considered to reduce the ECG signal noises like PLI and Base Line Interference. Results of simulations in MATLAB are presented.**

Keywords-component; adaptive filter; adaptive algorithm; LMS RLS

I. INTRODUCTION

In the process of digital signal processing, often to deal with some unforeseen signal, noise or time-varying signals, if only by a two FIR and IIR filter of fixed coefficient can't achieve optimal filtering. Under such circumstances, we must design adaptive filters, to track the changes of signal and noise. Adaptive Filter is that it uses the filter parameters of a moment ago to automatically adjust the filter parameters of the present Moment, to adapt to the statistical properties that signal and noise unknown or random change, in order to achieve optimal Filter. Based on in-depth study of adaptive filter based on the least mean square (LMS) algorithm and recursive least squares [1].(RLS) are applied to the adaptive filter technology to the noise, and through the simulation results prove that its performance is usually much better than using conventional methods designed to filter fixed.

II. ADAPTIVE FILTER

The so-called adaptive filter, is the use of the result of the filter parameters a moment ago, automatically adjust the filter Parameters of the present moment, to adapt to the unknown Signal and noise or over time changing statistical properties in order to achieve optimal filtering. Adaptive filter has "self regulation" and "tracking" capacities. Filter out an increase noise usually means that the contaminated signal through the filter aimed to curb noise and signal relatively unchanged. This filter belongs to the scope of optimal filtering, the pioneering work started from Wiener, and Kalman who work to promote and strengthen. For the purpose of the filter can be fixed, and can also be adaptive. Fixed filter designers assume that the signal characteristics of the statistical computing environment fully known, it must be based on the prior knowledge of the signal and noise [2]. However, in most cases it is very difficult to meet the conditions; most of the practical issues must be resolved using adaptive filter. Adaptive filter is through the observation of the existing signal to understand statistical properties, which in the normal operation to adjust parameters automatically, to change their performance, so its design does not require of the prior knowledge of signal and noise characteristics.

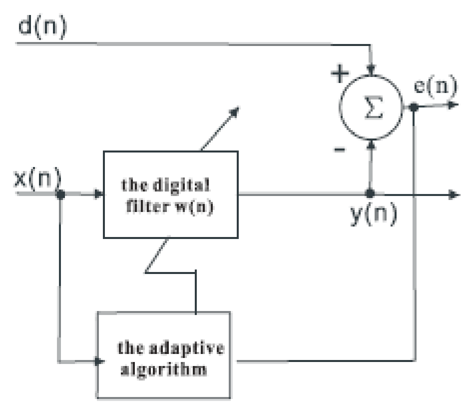

Fig. 1 Configuration for adaptive noise cancellation

The figure above is given the general adaptive filtering display digital filter carries on filtering on the input signal x(n), produce output signal y(n). Adaptive algorithm adjusts the filter coefficient included in the vector w (n), in order to let the error signal e(n) to be the smallest. Error signal is the difference of useful signal d(n) and the filter output y(n).

Therefore, adaptive filter automatically carry on a design based on the characteristic of the input signal x(n)and the useful signal d(n). Using this method, adaptive filter can be adapted to the environment set by these signals. When the environment changes, filter through a new set of factors, adjusts for new features. The most important property of adaptive filter is that it can work effective in unknown environment, and to track the input signal of time-varying characteristics. Filter out an increase noise usually means that the contaminated signal through the filter aimed to curb noise and Signal relatively unchanged. This filter belongs to the scope of Optimal filtering, the pioneering work started from Wiener, and Kalman who work to promote and strengthen. For the purpose of the filter can be fixed, and can also be adaptive. Fixed filter designers assume that the signal characteristics of the statistical computing environment fully known, it must be based on the prior knowledge of the signal and noise. However, in most cases it is very difficult to meet the conditions; most of the practical issues must be resolved using adaptive filter. Adaptive filter is through the observation of the existing signal to understand statistical properties, which in the normal operation to adjust parameters automatically, to change their performance, so its design does not require of the prior knowledge of signal and noise characteristics.

III. ADAPTIVE FILTER DESIGN AND SIMULATION.

A. *MATLAB Introduction*

MATLAB has a special function adaptfilt.lms to achieve Adaptive filtering; adaptfilt.rls function (1) is used for:

ha = adaptfilt.rls(L,lam,P0) (1)

[y,out] = filter(ha,x,d)
Out is the filter output signal, x is the filter input signal, d is the desired signal, y is the error signal.
Sampling Frequency fs=500
L=10 (FILTER ORDER OR COEFFICIENT)
N=2001 (NUMBER OF POINTS)
t =(1:N)/fs (TIME VECTOR FOR PLOTING)
w =(1:N)*4*pi/f (DATA FREQUENCY VECTOR)

DATA PLOTING
x=data(:,2)
plot(t,x,'m')

b = fir1(32,0.5)
d = filter(b,1,x) (DESIRED SIGNAL)

MATLAB offers several adaptive algorithm functions [4] to simplify the different word-length adaptive algorithm of MATLAB support for updating filter coefficient of LMS, normalization of the LMS, symbols LMS, RLS and Kalman filter algorithm. These algorithms provide the adaptive filter performance on the first step, but in this paper, I have used only two algorithms, which is given below.

B. *The Simulation Based on the Least Mean Square (LMS)* Algorithm

LMS algorithm the basic idea: adjustment of the filter parameters let the mean squares error between the filter output signal and the expectations output signals be smallest, such as system output is the best estimate of useful signal. Based on the steepest decline of the least mean square error (LMS) algorithm iterative formula is as follows:

$$e(n)=d(n)-X(n)^T *W(n) \quad (3)$$

$$W(n+1)=W(n)+2*u*e(n)*X(n) \quad (4)$$

$$X(n)=[x(n),x(n-1)...x(n-L-1)]^T \quad (5)$$

%n moment that the input signal vector;

$$W(n)^T =[w(n)_0 w(n)_{11111} ...w(n)_{L-1}] \quad (6)$$

Write a M document in the MATLAB environment, we design and simulate the application of LMS algorithm in noise cancellation, compare the function of filter under various different parameter circumstance, verify the feasibility of algorithm. Write a M document lms (noise, xn _noise, M, deft) in the MATLAB, this document is the application of LMS algorithm in noise cancellation [7]. Among them, the significance of the parameters are as follows:
noise: filter input signals, reference noise of column vector signal.
xn_noise: filter input signals, the column vector signal after the original signal and noise are superimposed.

M: filter order.

deft: filter step factor.

Ims document return two values, namely: [wn, en], respectively, said:

wn: the right to filter the vector sequence.

en: error signal.

The renewal expression of realization code in M the document:

```
for k = M: itr
% the k time iteration
noise _tap = noise ( k:-l:k-M+1 );
en(k) = xn__noise ( k ) – wn ( :,k-1 ) '*noise_tap;
wn(:,k) = wn ( :,k-1) + 2*delt*en ( k,l ) *noise_tap;
end ;
```

Simulation, the original signal chooses for
sin ((0.5 * pi) * t), which t = 0:0.1:15, noise signal adopts a

Standard white noise.

First select filter order M = 2 step factor deft = 0.1, in the MATLAB environment enter the following Order:

>> [wn,en] = Ims (noise1, xn_noise, M, deft)

Calling mapping function, get the simulation results the Application of LMS algorithm in noise cancellation:

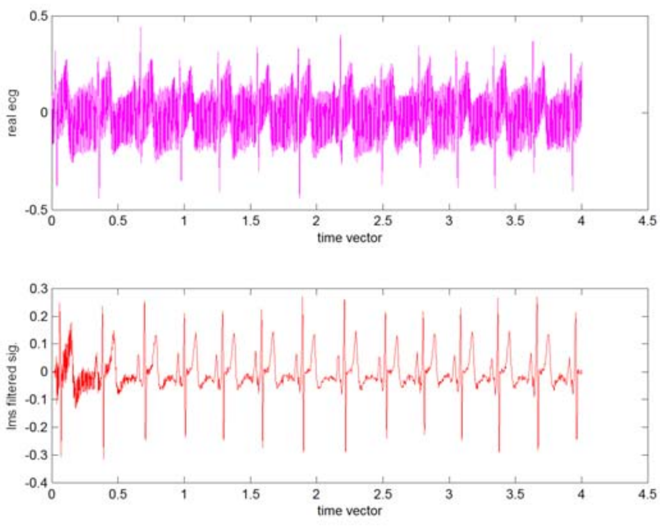

Fig.2 The LMS filter input (signal with noise) and filtered signal

C. The Simulation Based on the Recursive Least Square (RLS) Algorithm

RLS algorithm iteration expressions are following:

$$\pi(n)=P(n-1)x(n) \quad (7)$$

$$k(n)=\pi(n)/(\lambda+x^{T}(n)\pi(n)) \quad (8)$$

$$e(n)=d(n)-w^{T}(n-1)x(n) \quad (9)$$

$$w(n)=w(n-1)+k(n)e(n) \quad (10)$$

$$P(n)=\lambda-1P(n-1)-\lambda-1k(n)xT(n)P(n-1) \quad (11)$$

According to the above expressions, establish M document rls (x, d, N, lambda, delta) in MATLAB software, which is RLS algorithm of adaptive filter for the realization document [10].
The parameters are as follows:

x: filter input signal, reference noise input.

d: the original signal and noise of the signal superimposed.

N: filter order.

lambda: the corresponding expressions λ .
delta: the corresponding expressions δ .

The function return two value, respectively w and e, its significance is as follows:

w: filter weights.

e: filter the output signal.

All parameters were taken as follows:

N = 16, lambda = 1, delta = 1,
Weights for the initial value of 0, $P(0)= 1/\delta\ I$.

Calling mapping function, get the simulation results the Application of RLS algorithm in noise cancellation:

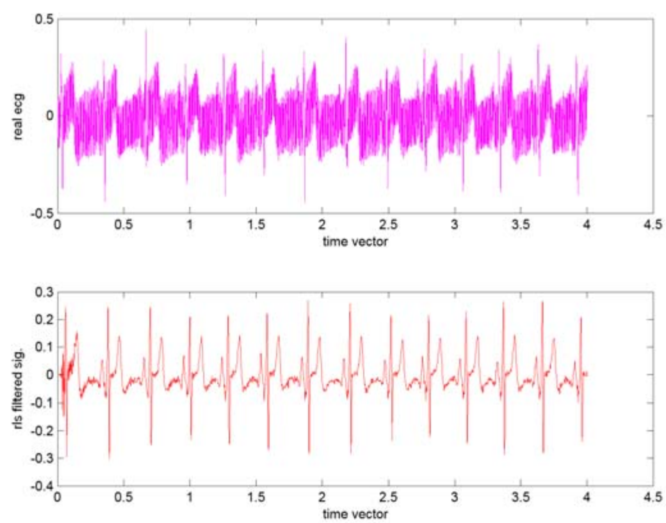

Fig.3 The RLS filter input (signal with noise) and filtered signal

Part of its algorithm for the realization code is that:

```
for n = 1: Nmax;

xt=x(n+N-1:-l:n).' ;

e(n)=d(n)-w(:,n)'*xt ;

v = P*xt;
q=xt'*v ;

Factor = 1/( lambda + q );

c2(n) = lambda*factor;
```

```
k = factor*v;

w (:,n+1) = w (:,n) + (e(n))*k;

P=invlamb*(P-k*v');

End;
```

IV. COMPARISON OF THE TWO ALGORITHMS SIMULATION RESULTS

The simulation results show that LMS algorithm give good results in comparison to RLS algorithm in the area of Biomedical Signal Processing to cancel the noise. To complete the task of noise reduction LMS filtering results is relatively good, the requirements length of filter is relatively short, it has a simple structure and small operation and is easy to realize hardware.

But the shortcomings of LMS algorithm convergence rate are slow. The noise signal and signal power when compared to larger, LMS filter output is not satisfactory, but we can step through the adjustment factor and the length of the filter method to improve [11].

ECG NO.	**LMS**		**RLS**	
	MSE	SNR	MSE	SNR
01	0.00091754	65.552	0.00087971	26.022
02	0.00078286	65.942	0.00077315	32.52
03	0.0060816	42.417	0.0060969	30.582
04	0.0044897	2.3955	0.0043666	1.1335

RLS algorithm filter the convergence rate is faster than the LMS algorithm, the convergence is unrelated with the spectrum of input signal, its each iteration is much larger operation than LMS.

TABLE I.

V. CONCLUSION

The work presented in this paper is to design and analyze the performance characteristics of the two adaptive filters LMS & RLS. The MSE reduces and SNR increases in LMS adaptive Filter in comparison to RLS adaptive filter.

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