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ECG Noise Removal using Adaptive Filtering

Mr. Deepak Bhoyar[1] , Prof .Primal Singh[2]
[1]Reasearch scholar 3rd sem M.Tech (D.COMM), ASCT, Bhopal [2]Assistant Professor, Deptt. Of Electronics & Commumication Engg. ASCT, Bhopal
deepak4d444@gmail.com[1], pre_army@ymail.com[2]

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.
Index Terms— 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. DESCRIPTION OF ECG

Just as the electrical activity of the pacemaker is communicated to the cardiac muscle, echoes of the depolarization and repolarization of the heart are sent through the specialized pacemaker cells that start the electrical sequence of depolarization and re-polarization of cardiac tissue is called inherent rhythmicity or automaticity. The electrical signal is generated in the sinus-atrial node(SA node) and spreads to the ventricular muscles via particular con ducting pathway; internodal atrial fibers, the atrioventricular node(AV node), the bundle of His, the right and left bundle brunch(RBB and LBB), the purkinje fibers then to ventricle (Fig. 1).

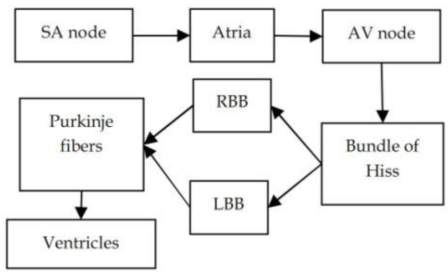

Fig. 1: Source of electrical activity in the heart

By introducing pairs of very sensitive receivers (Electrodes) on other parts of the body, the "echoes" of the heart's electrical activity can be detected. [2] - This electrical event of the heart is usually on the ECG as a pattern of a baseline (straight line on the ECG, a point of departure of the electrical activity of depolarizations and re-polarizations of the cardiac cycles) broken by P wave, QRS Complex and a T wave (Fig. 2)
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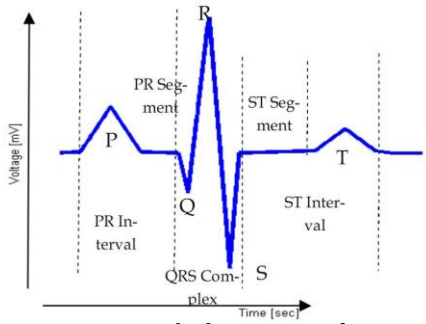

Fig. 2: Ideal ECG waveform

It should be noted that there are some interval (an interval is part of the ECG containing at least one wave and straight line. For example, the PR interval includes the P wave and the connecting line before the QRS complex) and segment (it is the period of time from the end of one wave to the beginning of the next wave. For example, the PR segment represents the time of AV nodal delay and transmission to the ventricle) between the waves. The generalized properties of ECG are briefly described in table I. [2] [3] [7]

TABLE I
DESCRIPTION OF ECG COMPONENTS

Seg-ment	Amp(mV)	Duration	Represents
P	0.25	0.08	Polarization of Artia
Q	25% of R		Spetal Depolarization
R	1.60		Ventricular Depolariza-tion
P-R Interval		0.12- 0.20	Time taken SA node to travel to Ventricle
QRS Com-plex		0.09	Ventricular depolariza-tion and Contraction
T	0.1 -0.5	0.16	Beginning of ventricular relaxation
S-T segment		0.05-0.15	Interval between S and T Wave

III. 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.

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

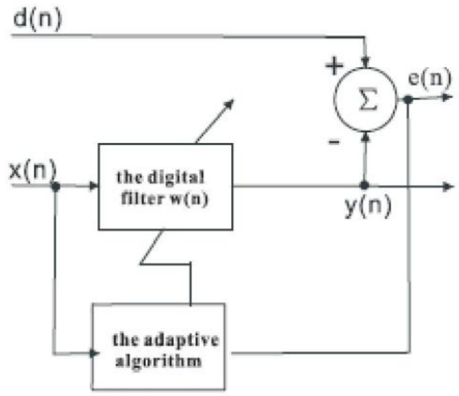

Fig. 3: Adaptive filter

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.

IV. ADAPTIVE FILTER DESIGN AND SIMULATION

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.

The RLS (recursive least squares) algorithm is another algorithm for determining the coefficients of an adaptive filter. In contrast to the LMS algorithm, the RLS algorithm uses information from all past input samples (and not only from the current tap-input samples) to estimate the (inverse of the) autocorrelation matrix of the input vector. To decrease the influence of input samples from the far past, a weighting factor for the influence of each sample is used. This weighting factor is introduced in the cost function. The dependence of the stability bound on the signal power is exploited in the normalized LMS algorithm by normalizing the step-size according to the signal power. The RLS algorithm is computationally more complex than the LMS algorithm. Note, however, that due the recursive updating the inversion of matrix $\Phi[n]$ is not necessary (which would be a considerably higher computational load). The RLS algorithm typically shows a faster convergence compared to the LMS algorithm.

V. COMPARISON OF THE TWO ALGORITHMS

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.
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.
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].

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.

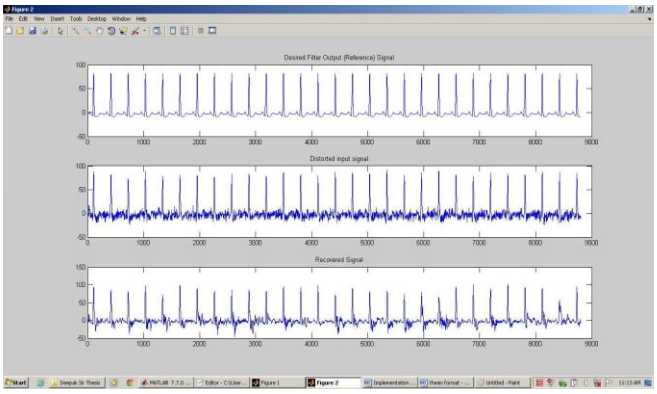

Fig. 4: ECG noise removal

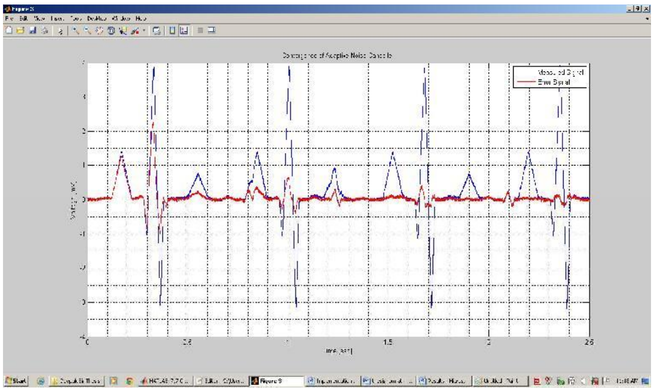

Fig 5: ECG filtering adaptive error signal

TABLE I
COMPARISON OF LMS AND RLS

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

VI. 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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