NOISE REDUCTION TECHNIQUES IN ECG USING DIFFERENT METHODS
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ABSTRACT: Due to fast life style Heart related problems are increasing day by day and it’s very important that disease related to heart can be diagnosed easily by simple medical techniques. These diseases can be diagnosed by ECG (Electrocardiogram) signals. ECG measures electrical potentials from the body surface with contact electrodes, thus it is treated as one of the important signals. The ECG recording is often deteriorated by several factors such as power line interference and baseline wander noise. Various noises have to be removed for better clinical evaluation. In this paper methodologies Like, Adaptive filtering using LMS algorithm, Digital IIR filter & FIR filter using hamming window is used to remove such noise in the ECG signal. Signal to noise ratio (SNR) are calculate from ECG signal and compared with the performance of proposed methods used for removal of ECG noise.

Keywords—ECG, power line interference, Adaptive filter, IIR filters, Hamming Window

I INTRODUCTION

The electrocardiogram (ECG) [1] is a non-invasive test that shows the electrical activity of the cardiac over time and it is very useful in the study and investigation of cardiac disease, for example a cardiac arrhythmia. In recording an ECG signal is frequently corrupted with different types of noises in the form electrical and mechanical noises. Electrical noise such as 50Hz power line interference baseline drift, electrode movement, white noise and motion artifact etc. Hence, removal of artifacts in ECG signal is as a pre-processing action in number of disease analysis for diagnosis and clinical applications. When we try to de noise the ECG signal, gets some sort of success when using traditional methods such as linear filers, signal averaging, and their combination. Recently, adaptive and non-adaptive filters and wavelet transformation have been developed as one of the most common and effective tools in processing and analysis of biomedical signals such as ECG. It can be recorded by electrodes placed on the surface of body. Fig.1 shows ECG signal of normal heart beat. The power line interference noise (50 Hz) introduce in system due to the electromagnetic interference from the electric-power system. The main cause for the power line interference is poor grounding of ECG machine. Such noise can cause problems interpreting low-amplitude waveforms. Baseline wander noise, usually in the range of 0.15Hz - 0.3Hz occurs due to perspiration, respiration, body movements of the human. The least mean squares (LMS) criterion is a search algorithm that can be used to provide the strategy for adjusting the filter coefficients. A number of adaptive structures have been used for different applications in adaptive filtering [2].

The objective of this paper is removal of noise from ECG signal. The paper suggests the methods which are not much complicated. ECG signal is shown in Fig.1
Signal processing is a big challenge since the actual signal value will be 0.5mV in an offset environment of 300mV. Other factors like AC power-supply interference, RF interference from surgery equipment, and implanted devices like pace makers and physiological monitoring systems can also impact accuracy [3]. The main sources of noise in ECG are
- Baseline wanders (low frequency noise)
- Power line interference (50Hz or 60Hz noise from power lines depends upon country)
- Muscle noise (This noise is very difficult to remove as it is in the same region as the actual signal. It is usually corrected in software.)
- Other interference (i.e., radio frequency noise from other equipment)

**NOISES IN ECG:**

Different values of artificial noise patterns were assessed for each of the noise categories. Potential differences of 1 to 3 mV generated at the body surface by the current sources in the heart are picked up by the electrodes and are amplified in order to improve the signal to noise ratio (SNR).

The digitization process should use a sampling rate of at least 1 kHz to ensure that the ECG trace is of a high enough resolution as required for biometric purposes [4]. ECG measurements may be corrupted by many sorts of noise. The ones of primary interest are,
1. Power line interference
2. Base Line Wandering
3. Motion artifacts

**II. PREVIOUS TECHNIQUES**

Digital FIR filters are successfully employed in processing electrocardiographic [5] signals for measurement. ECG which is a biomedical signal is naturally corrupt by various interferences such as 50Hz Power Line Interferences (PLI) and some other biomedical signals like baseline wander ECG signal frequency is approximately between 0.05Hz and 100Hz. Baseline Wander frequency is below 1 Hz.[1] These interferences have to be removed from ECG signal in order to obtain correct clinical information of the heart. Since the frequency of ECG depends on the muscle movement rate and pressure it can be reduced to the barest minimum during ECG measurement by the patient staying still and quiet so that the muscles are fully relaxed.

When designing digital FIR filters using window functions it is necessary to specify[1][5]
- A window function to be used; and
• The filter order according to the required specifications (selectivity and stop band attenuation). These two requirements are interrelated. Each function is a kind of compromise between the two following requirements:
• The higher the selectivity, i.e. the narrower the transition region; and
• The higher suppression of undesirable spectrum, i.e. the higher the stop band attenuation.

There are two categories of digital filter: the recursive filter and the non-recursive filter. The desired frequency response specification \( H_d(\omega) \), corresponding unit sample response \( h_d(n) \) is\[1\][5]
determine using the following relation

\[
h_d(n) = \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(\omega) e^{j\omega n} d\omega \] (1)

Where

\[
H_d(\omega) = \sum_{n=-\infty}^{\infty} h_d(n) e^{-j\omega n} \] (2)

A comparison is also done for all the different windows. In this section we present different window methods which are used to design of FIR filters.[1][9]

A. Kaiser Window:

In this window the side lobe level can be controlled by with respect to the main lobe peak by varying a parameter \( \alpha \). Kaiser window parameter \( \beta \) affects by side lobe attenuation \( \alpha \) db [9]. The width of main lobe can be varied by adjusting the length of the filter.

\[ \beta = \begin{cases} 
0.1102(\alpha - 8.7), & \alpha > 50 \\
0.582(\alpha - 21)^{0.4} + 0.7886(\alpha - 21), & 21 \leq \alpha \leq 50 \\
0, & \alpha < 21 
\end{cases} \] (3)

Where \( \alpha = -20\log_{10}\delta \) is the stop band attenuation in db. Increasing \( \beta \) then decreases the amplitude of side lobe. Filter order for FIR filter is

\[ N = \frac{\alpha - 8}{2.285\Delta \omega} + 1 \] (4)

Here \( N \) is the filter order and \( \Delta \omega \) is the width of the smallest transition region.

B. Hanning Window:

The coefficient of a Han window is calculated from the following equation.

\[ \omega(n) = \begin{cases} 
0.5 - 0.5\cos \frac{2\pi n}{N}, & 0 \leq n \leq M - 1 \\
0, & Otherwise 
\end{cases} \] (5)

The width of main lobe is approximately \( 8\pi/M \) and peak of first side lobe is at \( -32\text{dB} \) [1][10].

III.PRAPOSED SYSTEM

A. FIR filter design using hamming window:

We are used windowing technique to design FIR. This paper investigates the performance of Hamming window with detailed design and performance waveforms and responses using matlab generated data so as to confirm the suitability or otherwise of Hamming window in removing power line noise. A typical ECG signal [11] generated by human heart is shown in fig.
The Hamming window is one of the most popular and most commonly used windows. A filter designed with the Hamming window has minimum stop band attenuation of 53dB, which is sufficient for most implementations of digital filters. The causal Hamming window function are calculated by

$$\omega(n) = \begin{cases} 0.54 - 0.46\cos\frac{2\pi n}{M-1}, & 0 \leq n \leq M - 1 \\ 0, & \text{Otherwise} \end{cases}$$

The width of main lobe is approximately $8\pi /M$ and the peaks of first lobe is at $-43$dB.

B. IIR filter design with Butterworth approximation:

The theory of Butterworth function is explained here but, the order of the filter should be high and implementing a filter of that order is not easy to perform. In addition to this difficulty, solving these high order equations is not straightforward.

The Butterworth family of lowpass filter is defined by the following formula
\[ |H_B(j\Omega)|^2 = \frac{1}{1 + \left(\frac{\Omega}{\Omega_c}\right)^2} N = 1, 2, .... \]

Pole locations
The solution of the equation \(1 + \left(\frac{s}{j\Omega c}\right)^2 N = 0\) yields the \(2N\) poles:
\[ s_k = \sigma_k + j\Omega_k, \quad \text{Where} \quad \begin{cases} \sigma_k = \Omega_c \cos \theta_k \\ \Omega_k = \Omega_c \sin \theta_k \end{cases} \]

With
\[ \theta_k = \frac{\pi}{2} + \frac{2k - 1}{2N} \pi, \quad k = 1, 2, ..., 2N \]

- poles never occur on the imaginary axis
- poles occur on the real axis only for odd values of \(N\)
- stability is obtained by selecting the \(N\) poles lying on the left-half plane (gray)

C. Design of an Adaptive Filter Algorithm:
The adaptive filters are self-designing filters, which determine the updating of filter coefficients requires the extra information in the form of signal. This signal is called desired or reference signal. [12]. The signal processing system for adaptive filters processes different signal in different algorithm grounded in statistical basis. The adaptive filters are used for those applications where it needs to operate for high speed, it is essential to minimize the hardware complexity. [13] Due to simple mathematics of the LMS algorithm, it focuses on its transversal filter structure: the input data vector which is stored in the delay elements of the filter is computed as a linear combination with its output.

The Adaptive algorithm tries to minimize an appropriate objective or error function that involves the input, reference and filter output signal. This algorithm can be consist of three parts, the definition of minimizing algorithm, the definition of objective function and the definition of error
signal. The LMS algorithm is much attractive for different application due to its simplicity and accessible analysis under idealized conditions.

Fig: 7 Outline of adaptive transversal filter

The outline of adaptive signal processing system is describes in figure 7. The input to the adaptive filter is described as scalar real-valued discrete-time signal $x(n)$ where $n$ is the time index. At time $n$, the samples $x(n)$, $x(n-1)$, ... through $x(n-N+1)$ are simultaneously present in the delay elements of the filter.

**IV. RESULTS**

The results are obtained with the designed filters using different raw ECG MIT-BIH data for different filters & hamming window in the digital FIR filter. The filter with hamming window & different samples gave different results. The graphs for the signal and their SNR are shown below for different Fs and different methods.

**Fig: 8 Waveform for output Result using Adaptive Filter**

**Fig: 9 Waveform for output Result using IIR Filter**
Result Table:

Table 1: SNR of ECG Denoising signal using Different Methods

| Sampling Frequency | SNR          |
|--------------------|--------------|
|                    | Adaptive Filter | IIR Filter | Hamming Window |
| 500                | 21.55        | 24.55      | 13.90          |
| 1000               | 24.15        | 26.66      | 13.89          |
| 2000               | 26.47        | 28.90      | 13.90          |

V. CONCLUSION

The goal of the adaptive filter is to match the filter coefficients to the noise so that the adaptive filter can subtract the noise out from signal. So adaptive filtering technique was selected to achieve the goal of this paper. FIR filter is easy to implement but it has poor result while increasing order of filter. IIR filter has better result than windowing techniques when we take same sampling frequency.

By result analysis we conclude that IIR using Butterworth approximation having better results compared to adaptive and hamming window function.

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