

Digital Filtration of ECG Signals for Removal of Baseline Drift

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Abstract. Baseline wander elimination is often one of the first steps required in the processing of the electrocardiogram (ECG). Baseline wander makes manual and automatic analysis of ECG records difficult, especially in the detection of ST-segment deviations. This segment is very important and has the information related to heart attack. Since the spectrum of baseline wander and low frequency component of ECG signal usually overlaps, removing of baseline wander may cause distortion of important clinical information. Baseline noise occurs due to respiratory signal and body movements. Respiratory signal wanders between 0.15Hz and 0.5Hz frequencies [1]. One of the most common methods to remove baseline wander is high pass filtering. This research review is based on High pass filtering of ECG signal to remove this baseline wander while preserving the low frequency ECG clinical information. The rapid development of high-speed digital integrated circuit technology in the last three decades has made digital signal processing the technique of choice for most of the applications [2]. Today, the hardware has been replaced by software, as digital signal processing is very much advantageous over analog signal processing. Digital systems can be reprogrammed; easily duplicated and digital signals can be easily stored which makes remote processing possible. The performance of digital signal processing and communication system is generally limited by precision of digital input signal which is achieved at the interface between analog and digital information. The development of Sigma delta modulation based analog to digital converter is a cost effective alternative for high resolution converter (> 12 bits) which can be integrated on digital signal processor IC's has also eliminated this problem.

Keywords: FIR, IIR, Notch Filter, ECG, Window method

1. Introduction to Digital Filters

A digital filter is characterized by its transfer function. The transfer function has the form:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1z^{-1} + b_2z^{-2} + \dots + b_Nz^{-N}}{1 + a_1z^{-1} + a_2z^{-2} + \dots + a_Mz^{-M}}$$

where order of the filter is the greater of N or M. There are two types of filters on basis of impulse response:

FIR (Finite Impulse Response): if all the coefficients a_1, a_2, \dots, a_M make equal to zero in equation (1), the filter becomes non-recursive. The difference equation for FIR filter is as follows:

$$y[n] = \sum_{k=0}^M b_k x[n-k]$$

The output of FIR filter depends only upon present and previous inputs. There is no involvement of feedback i.e. its output does not depend on previous outputs.

FIR filters are simple to design; they are guaranteed to be bounded input-bounded output (BIBO) stable. FIR filter can be guaranteed to have linear phase. These filters also have a low sensitivity to filter coefficient quantization error. This is an important property to have when implementing a filter on a DSP processor or on an integrated circuit.

IIR (Infinite Impulse Response): The difference equation for this filter is as follows:

$$y(n) = - \sum_{k=1}^N a_k y(n-k) + \sum_{k=1}^M b_k x(n-k)$$

In this case the filter output depends upon previous inputs, present inputs and also on previous outputs. IIR filters are useful for high-speed designs because they typically require a lower number of multiply compared to FIR filters. IIR filters can be designed to have a frequency response that is a discrete version of the frequency response of an analog filter. These filters also are very sensitive to filter coefficient quantization errors that occur due to using a finite number of bits to represent the filter coefficients[3].

2. Design of Filter

For design the filter, there are following necessary steps shown in Fig 1[4]. Ideal filter have constant gain in the pass band and zero gain in the stop band. These ideal filters have frequency responses with sharp cut off edges (discontinuities) and cannot be implemented directly because ideal filters are non-causal ($h(n)$ is present for negative values of n) as shown in Fig 2. To realize a digital filter, the response of filter should be causal; the sharp cutoff edges need to be replaced with transition bands as shown in Fig 3[6].

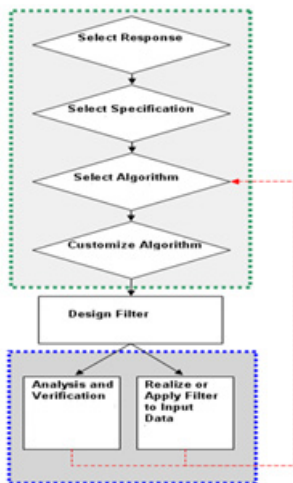


Fig 1: Ideal frequency responses of commonly used frequency-selective filters

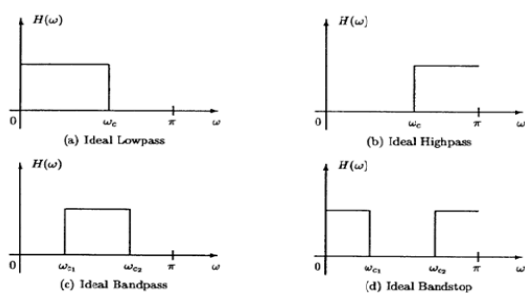


Fig: 2 common ideal Digital filter types

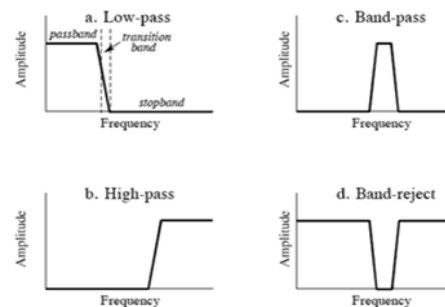


Fig3: Digital filters with transition bands

Approximations to realize a filter

1. It is not necessary to have constant magnitude in entire range of passband
2. A small amount of ripple in passband is allowed.
3. It is not necessary that the filter response in stopband should be perfectly zero

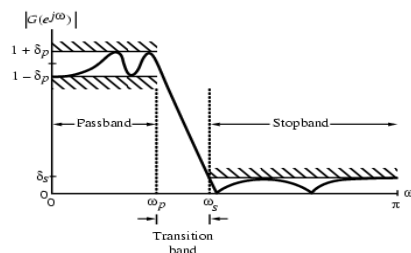


Fig 4: Approximations

Digital Filter Specifications:The design involves several considerations, of which the most crucial is the choice of filter cut-off frequency. The cut-off frequency should be chosen so that the clinical information in the ECG signals remains undistorted while as much as possible of the baseline wander is removed.

3. Design and Implementation of IIR Filters

There are four classical IIR filters namely Butterworth, Chebychev Type I and II, and Elliptic filters[4].

Butterworth filter: The decrease is very slow in the pass band and quick in the stop band. In a design problem where no ripple is acceptable in pass band and stop band, Butterworth filter is a good choice.

Chebyshev Type 1: This filter results in appearance of ripples in the passband. The stopband response is maximally flat. The transition from passband to stopband occurs faster than the Butterworth filter. If the ripples in the passband are acceptable, a Chebychev filter usually require a lower-order transfer function than a Butterworth filter for the same specifications.

Chebyshev Type II: This filter results in appearance of ripples in the stopband. The passband response is maximally flat. The transition from passband to stopband occurs slower than the type I filter and for even filter order, it never reaches zero. Its advantage over type I is the absence of ripples in the passband.

Elliptic filter: The elliptic filter results in the steepest transition from passband to stopband by allowing ripples in both passband and stopband.

Design using MATLAB: These filters are designed in matlab software using Filter Design Toolbox with followings specifications:

cut-off frequency= 0.5Hz, Sampling frequency= 360 Hz, Order of filter= 2, Filter response = High pass

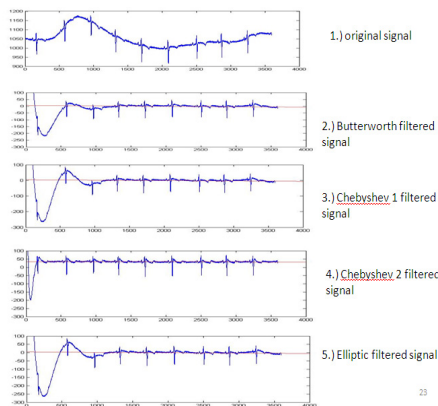


Fig 5: Original ECG signal and IIR filtered signals

Filter	PSD before filtration (dB/Hz)	PSD after filtration (dB/Hz)	Average power (dB)	Ringing effect
Butterworth	49.16	35.35	40.94	yes
Chebyshev 1	49.16	33.94	40.70	yes
Chebyshev 2	49.16	13.85	36.21	yes
Elliptic	49.16	13.85	36.21	yes

Table 1: Comparison of IIR filters

4. Design and Implementation of FIR Filters

FIR filters are further classified into two types[7]: window method and Frequency sampling domain method

Design Steps of Windows Method

- choose a proper ideal frequency-selective filter (which always has noncausal, infinite-duration impulse response).
- Then truncate (or window) its impulse response to obtain a linear-phase and causal FIR filter.
- To obtain a causal and linear-phase FIR filter $h(n)$ of length M ,

$$h(n) = \begin{cases} h_d(n) & 0 \leq n \leq M-1 \\ 0 & \text{elsewhere} \end{cases}$$

$$h(n) = h_d(n)w(n)$$

$$w(n) = \begin{cases} \text{some symmetric function} \\ 0, \text{ otherwise} \end{cases}$$

Depending on how we define $w(n)$ above, we obtain different window design. These are Rectangular window, Kaiser window, Hanning window, Hamming window and Blackmann window

Window Type	Peak Sidelobe Amplitude (relative)	Approximate Width of Mainlobe
Rectangular	-13	$4\pi/(M+1)$
Bartlett	-25	$8\pi/M$
Hanning	-31	$8\pi/M$
Hamming	-41	$8\pi/M$
Blackman	-57	$12\pi/M$

Table 2: Summary of different windows

Features of a good window for FIR filter are that attenuation should be more, side lobe level should be small, smoother magnitude response, Smoother ends, the tradeoff between main lobe width and side lobe level can be adjusted

5. FIR Equiripple Filter

An Equiripple design technique provides an alternative to windowing by allowing the designer to achieve the desired frequency response with the fewest number of coefficients

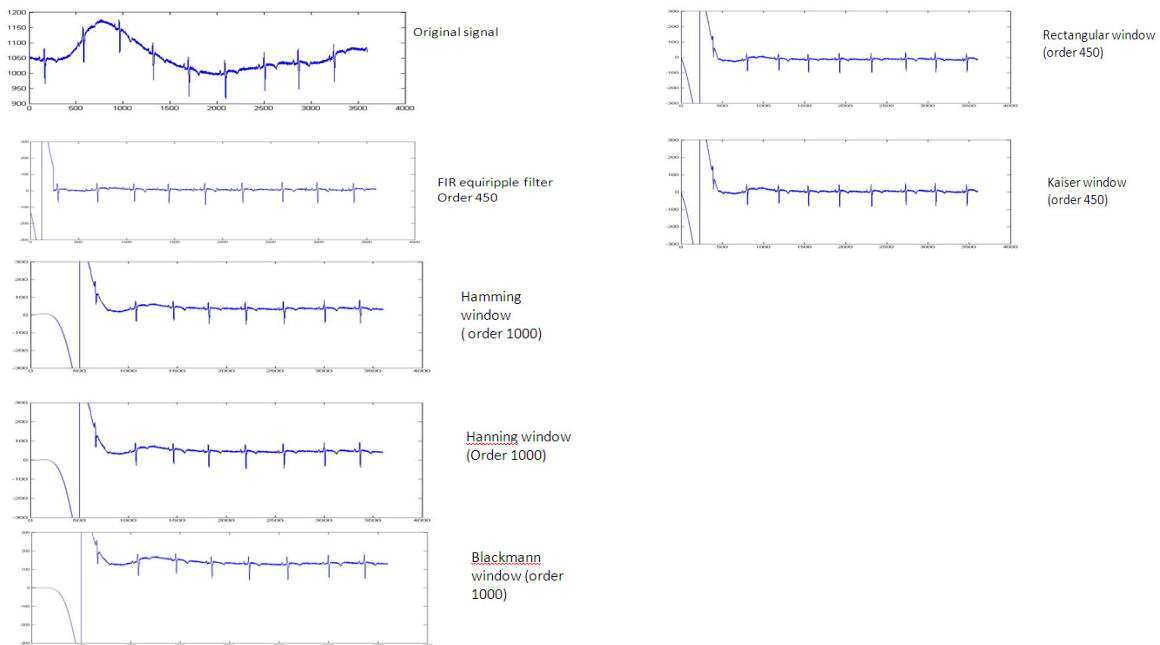


Fig 6: Original signal and FIR filtered signal

Filter	PSD before filtration(dB/Hz)	PSD after filtration (dB/Hz)	Average power (dB)	Ringing effect
FIR equiripple	49.16	24.92	40.78	yes
Kaiser	49.16	29.96	39.91	yes
Rectangular	49.16	29.42	39.95	yes
Hamming	49.16	30.17	40.55	yes
Hanning	49.16	30.25	40.83	yes
Blackmann	49.16	28.01	44.15	yes

Table 3: Comparison of FIR filters

Filter	Order of filter	No. of adders	No. of multipliers	No. of delays
FIR	450	450	451	450
IIR	2	4	4	2

Table 4: Complexity of Filters

6. Conclusions

It can be concluded that:

- FIR and IIR filter both have removed the baseline noise of ECG signal at the expense of some ringing effect at the starting of waveform.

- But their comparison shows that due less order of IIR filter there is no phase delay but there is a phase delay in case of FIR filters.
- The complexity of FIR filter is far greater than IIR filters.
- It increases the memory requirement for FIR filters.

So, IIR filters can be a better choice for removing baseline wander. This ringing effect can be removed by using bidirectional filtering i.e. Zero phase filtering.

7. References

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