



## GROUP DELAY COMPENSATED DIGITAL FILTERS FOR DENOISING OF ECG SIGNAL

**Hitender Kumar Tyagi**

Department of Electronics, Institute of Integrated and Honours Studies, Kurukshetra University, Haryana, India.

**Reeta Devi\***

Department of Electronics and Communication Engineering, University Institute of Engineering and Technology, Kurukshetra University, Haryana, India. \*Corresponding Author

**Dinesh Kumar**

J C Bose, YMCA University of Science and Technology, Faridabad, Haryana, India.

**ABSTRACT** Cardiac diseases are the leading cause of deaths worldwide. Thus, the demand for routine checkup of the functional status of heart is increasing. Electrocardiography (ECG) is an easy and affordable non-invasive technique for this purpose. But for reliable diagnostic use of ECG signal in real-time or offline analysis, it is mandatory to acquire a noise free signal. To obtain clean ECG signal a common practice is to use digital finite impulse response (FIR) or infinite impulse response (IIR) filters. But an important requirement with use of these filters is to have a zero phase distortion along with compensated group delay response so as to catch the unaltered natural transients in the cardiac cycle. Thus in this paper, we have presented certain group delay compensation techniques to use with digital filters. Also the effect of the modified filter on characteristics of the ECG signal was also analyzed and presented in terms of the computed total power spectral density estimates (total PSD) of the filtered signal.

**KEYWORDS :** Compensation, ECG, FIR, Group delay and IIR.

#### Delayed response in digital filters

The ECG signal in human beings consists of many temporal relationships. These relationships play an important role in disease diagnostic applications. The denoising of ECG signal with digital IIR filters may be found risky due to non-linear phase characteristics of these filters. The non-linear phase may destroy or alter important information from the ECG signal. Thus, the use of linear-phase FIR filters serves the purpose in such applications, but at the cost of very high filter order which further increases the group delay. Therefore, the FIR filters produce the delayed response as explained below with the mathematical derivations<sup>1</sup> that may cause problems in real-time analysis and in many other post processing applications<sup>1,2</sup>.

The impulse response function  $H(z)$  of a causal FIR filter is given as:

$$H(z) = \sum_{n=0}^{N-1} h(n)z^{-n} \quad (1)$$

where  $h(n)$  are the filter coefficients of impulse response function. After solving equation 1,  $H(z)$  is represented as:

$$H(z) = h(0) + h(1)z^{-1} + h(2)z^{-2} + \dots + h(N-1)z^{-(N-1)} \quad (2)$$

From equation 2, it is evident that FIR filters have only zeros. Hence they are also known as all-zero filters or the feedforward filters presenting the following two main advantages:

- (1) The inherently stable FIR filters because of absence of feedback from output to the input and
- (2) these filters have a linear phase characteristic thereby eliminating the possibility of phase distortion in the output waveform.

The FIR filters possess linear phase characteristic as their impulse response function satisfies the relations in equations 3-4:

$$h(n) = h(N-1-n), n=0, 1, \dots, N-1 \quad (3)$$

$$h(n) = -h(N-1-n), n=0, 1, \dots, N-1 \quad (4)$$

Equations 3-4 describe the symmetric and anti-symmetric impulse response functions respectively. Incorporating the equation 3 into equation 2, the frequency response i.e. the discrete time Fourier transform (DTFT) of  $h(n)$  may be obtained as:

$$H(\omega) = e^{-j\omega\frac{(N-1)}{2}} H_r(\omega), \quad (5)$$

where  $H_r(\omega)$  is a real function of  $\omega$  and can be expressed as

$$H_r(\omega) =$$

$$\begin{cases} h\left(\frac{N-1}{2}\right) + 2\sum_{k=0}^{\frac{N-3}{2}} h(n) \cos\left(\omega\left(\frac{N-1}{2} - k\right)\right) & ; \text{if } N \text{ is odd} \\ 2\sum_{k=0}^{\frac{N}{2}-1} h(n) \cos\left(\omega\left(\frac{N-1}{2} - k\right)\right) & ; \text{if } N \text{ is even} \end{cases}$$

(6)

The phase characteristics of the filter for both  $N$  odd and  $N$  even are  $\theta(\omega) =$

$$\begin{cases} -\omega\left(\frac{N-1}{2}\right), & \text{if } H_r(\omega) > 0 \\ -\omega\left(\frac{N-1}{2}\right) + \pi, & \text{if } H_r(\omega) < 0 \end{cases} \quad (7)$$

The group delay and phase delay of the filter are defined as:

$$\text{Group delay} = -\frac{d\theta}{d\omega} \quad (8)$$

$$\text{Phase delay} = -\frac{\theta}{\omega} \quad (9)$$

Thus, as evident from equation 7, the  $\theta(\omega)$  is a linear phase with group delay

$$\frac{(N-1)}{2}$$

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Similar, results can be derived for the anti-symmetric filter also (by incorporating the equation 4 into equation 2).

This group delay is responsible for the delayed response of the FIR filter as shown in Fig. 1. The group delay is also plotted in Fig. 2 for the filter used in filtering of the ECG waveform shown in Fig. 1.

The group delay of FIR filtering methods can be avoided to a significant amount by the use of IIR filtering methods<sup>3</sup>. But the non-linear phase distortion arises in this filtering which may alter the important cardiac transients. The IIR filtered signal and the group delay of this filter are also shown in Figs. 3-4.

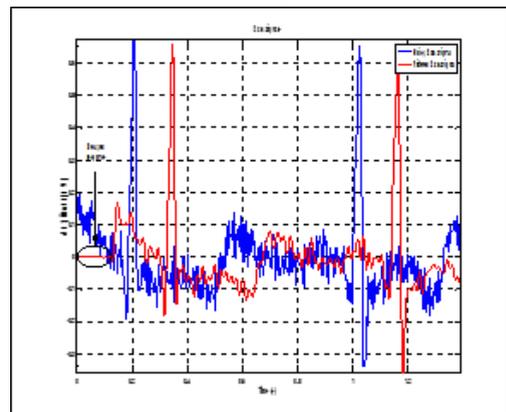


Figure 1: Filtering with digital FIR filter

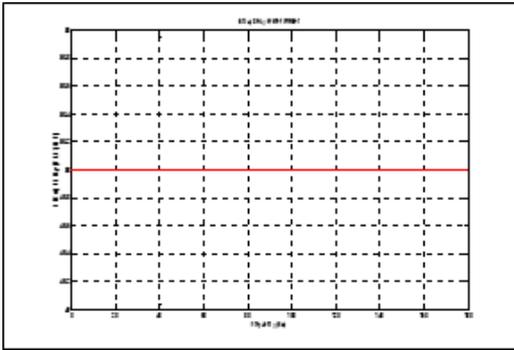


Figure 2: Group delay of the FIR filter

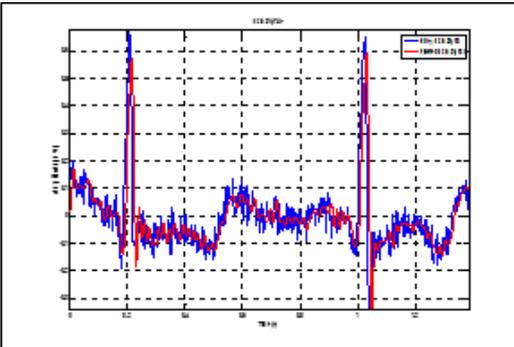


Figure 3: Filtering with digital IIR filter

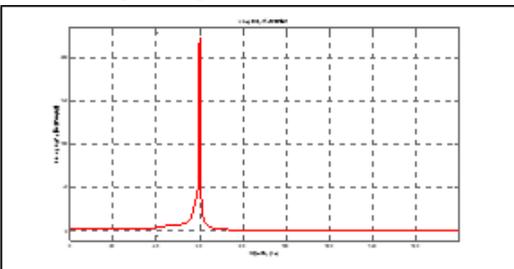


Figure 4: Group delay of the IIR filter

**Group delay compensated FIR digital filters**

The group delay of the digital filters can be compensated by employing different methods for FIR and IIR filters<sup>3-5</sup>. In case of FIR filter, the important methods are zero padding, zero delay modified overlap and save method (ZDMOSM), reduced delay modified overlap and save method (RDMOSM) and enhanced-RDMOSM (ERDMOSM1 and ERDMOSM2)<sup>3</sup>. The algorithms for these methods are explained below:

**Algorithm 1: Zero padding**

1. Compute  $D = \text{mean}(\text{group delay})$  of the impulse response function of the filter
2. Insert  $D$  zeros at the end of the input signal
3. Perform filtering of the padded input signal by the convolution method
4. Shift the filtered signal ( $D+1$ : end) to compensate for the delay.

**Algorithm 2: ZDMOSM (used in offline processing applications)**

1. Add  $(N-1)/2$  samples before the input signal to be filtered
2. With zero phase impulse response function perform block circular convolution
3. Remove approximately half samples  $((N-1)/2)$  from the left extremity of the filtered signal
4. Similarly remove another approximately half samples  $((N-1)/2)$  from the right extremity of the signal

**Algorithm 3: RDMOSM (used in online real time processing applications)**

1. Add  $(N-1)$  samples before the input signal to be filtered
2. With zero phase impulse response function perform block circular convolution
3. Remove approximately three-fourth samples  $(3(N-1)/4)$  from the left extremity of the filtered signal

4. Similarly remove approximately one-fourth samples  $((N-1)/4)$  from the right extremity of the filtered signal

**Algorithm 4: ERDMOSM1**

1. Add  $(N-1)$  samples before the input signal to be filtered
2. With zero phase impulse response function perform block circular convolution
3. Remove approximately four-fifth samples  $(4(N-1)/5)$  from the left extremity of the filtered signal
4. Similarly remove approximately one-fifth samples  $((N-1)/5)$  from the right extremity of the filtered signal

**Algorithm 5: ERDMOSM2**

1. Add  $(N-1)$  samples before the input signal to be filtered
2. With zero phase impulse response function Perform block circular convolution
3. Remove approximately nine-tenth samples  $(9(N-1)/10)$  from the left extremity of the filtered signal
4. Similarly remove approximately one-tenth samples  $((N-1)/10)$  from the right extremity of the filtered signal.

**Group delay compensated IIR digital filters**

The reductions of group delay in IIR filter is performed by zero-phase forward and reverse digital filtering method. In this method, the length of the input signal must be at-least 3 times more than the filter orders. The input noisy signal is convolved in the forward direction with the impulse response function of the filter. This filtered signal is reversed and convolved back again with the impulse response function of the filter. The final filtered signal is the time reversed signal of the second convolution operation and has precisely zero phase distortion. But the magnitude of the forward-backward filtered signal is modified by the square of the magnitude of the filter's impulse response function. The initial and closing transients are diminished by matching the initial conditions. But a major drawback with IIR filters is their non-linear phase response that can alter various temporal relationships in the cardiac cycle. This filter can't be used in online real time processing applications (specially the backward part). Also the high sampling rates can't be achieved with this filter as the poles move closer to the unit circle at high sampling rates which results in instability. The time-varying cut-off frequency are also hard to realize with IIR filters.

**RESULTS**

The reduction of group delay for effective filtering of ECG signal was carried out using the algorithms stated above. To compare the results of reduced delay filtering methods with the traditional filtering, the total power spectral density estimates over the frequency range of (0, 0.4) Hz was also computed. The filtered signals with various group delay compensation techniques were compared with the filtered signal of the traditional method as shown in Figs. 5-7.

The values of total PSD estimates obtained with ECG signal (record number 100) downloaded from the international standard database of Physiobank<sup>6</sup> are reported in Table 1. The effect of group delay reduced by different methods can easily be seen from the computed values of total PSDs which show the significant reduction with corresponding reduced delay from traditional filtering to RDMOSM1 filtering.

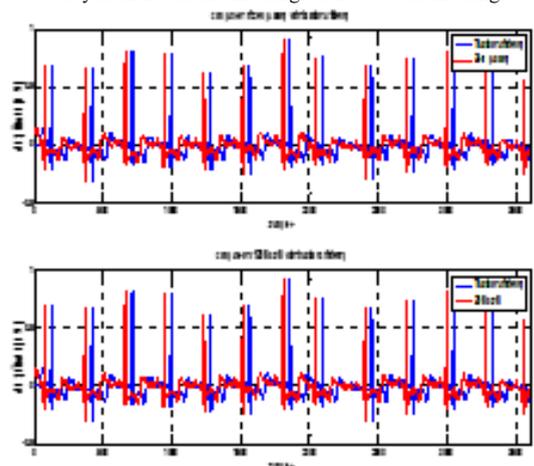


Figure 5: Comparison of traditional filtering and zero-padding filtering methods

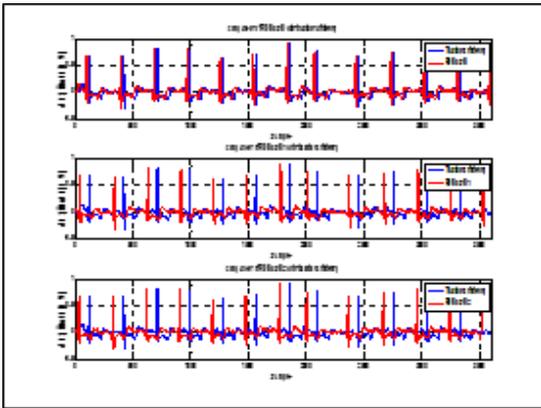


Figure 6: Comparison of traditional filtering with RDMOSM, RDMOSM1 and RDMOSM2

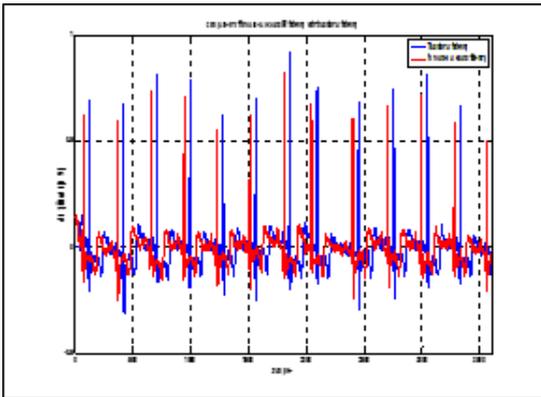


Figure 7: Comparison of traditional filtering and forward-backward IIR filtering

Table 1: Total PSD estimates of ECG signal

Method	Total PSD
Noisy signal	5.5929e-04
Traditional filtering	5.3914e-04
Zero-padding	5.3303e-04
ZDMOSM	5.3303e-04
RDMOSM	5.3231e-04
RDMOSM1	5.2128e-04
RDMOSM2	5.3447e-04
Forward-backward	8.5350e-04

However, the increased values of total PSD with RDMOSM2 may be contributed by the amount of ripples that occurred due to the large reduction of group delay by this method. Similarly, the high value of total PSD at the output of forward-backward zero phase IIR filtering can be stated by the modified magnitude by the square of the filter's magnitude response.

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