

# Filter design for ECG signal processing

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**Abstract:** Digital filters, as one of the most essential tools for signal processing, are often used in ECG signal processing. The hardware implementation of conventional digital filters relies on a hybrid device composed of a digital multiplier, an adder, and a delay unit. However, it has characteristics of a large amount of calculation, complex structure, and high overhead. With the development of computer technology, software implementation of signal processing and filter designing has been increasingly used. As a typical physiological signal, an ECG signal is characterized by a weak signal, complex noise, broad frequency spectrum, and little data sampling. The ECG filter designed by software can effectively weaken the useless noise in ECG signals, helping medical personnel obtain relevant information on patients' symptoms and realize the diagnosis of diseases.

**Keywords:** signal processing, digital filter, software design, ECG filtering, disease diagnosis

## 1. Introduction

Physiologically, the ECG signal is a signal generated by the human heart through a series of mechanical activities such as myocardial contraction, valve closure, and blood pumping, and transmitted to the surface layer of the chest through the chest cavity, chest wall, and other tissues [1]. According to the results of theoretical studies at home and abroad, the frequency range of the ECG signal is mainly concentrated within 0.8-100 Hz. It includes several components, namely, first ECG, second ECG, third ECG, fourth ECG, heart murmur, and external disturbances [2]. As a typical physiological signal, ECG signal is characterized by weak signal, complex noise, broad-spectrum and little data sampling [3]. How to suppress the noise and extract the pure and effective ECG signal is the problem to be solved in modern biomedical signal processing, which is also the main difficulty of ECG filter design.

The ECG signal can be picked up by mounting electrodes on the human skin surface [4]. For the same subject, the test should be performed on the same site. Considering the phenomenon of polarization between electrodes and skin, the presence of other physiological signals in the complex living system of the human body, and the interference of the magnetic field of the earth where we are located [4], the ECG signal we detect is highly susceptible to interference. The main interfering signals can be grouped into three categories: industrial frequency interference, myoelectric interference, and baseline drift.

Considering the three types of interferences in the ECG signal, the ECG filter is often a comprehensive filter for three different interfering signals, which contains a total of three digital filters: the first filter, uses a low-pass filter to filter out the myoelectric interference because the frequency of the myoelectric signal is determined by the muscle type and its frequency is often higher than the ECG frequency; the second filter, which uses a band trap filter to suppress the industrial frequency interference generated by the power supply network often at 50 Hz The third filtering, using a zero-phase filter, corrects the baseline drift [3].

Since ECG signals are generated by mechanical vibrations of the heart and nearby organs, they reflect the physiological characteristics and work of the heart and cardiovascular system. Therefore, by analyzing ECG patterns, much information can be obtained to determine cardiovascular diseases. The raw ECG signal, which is disorganized and irregular, is not apparent. ECG filters can help to filter out the disturbing part of the raw signal and construct a signal spectrum that is conducive to identification to

help medical personnel obtain correct information for early detection, prevention, and treatment of cardiovascular diseases [6].

Digital filters are one of the most important tools for signal processing. The hardware implementation of a conventional digital filter relies on a hybrid device consisting of a digital multiplier, an adder, and a delay unit. Its function is to sample the signal and perform arithmetic processing on the input discrete digital signal to change the signal spectrum. However, for signal processing with steep narrow bands, a more significant amount of computation is required, and the hardware for its digital filter requires a more complex component structure, as well as more significant overhead.

With the development of computer technology in recent years, the application of software has played an increasingly important role in signal processing, including ECG signals. More and more filter designs are also realized by software. Compared to hardware filter design, using software to design filters has advantages such as easy design and low cost. [5].

In this paper, we briefly introduce the background knowledge about ECG signals and ECG filters; and construct a model of an ECG processing system. The filter involved in this paper is a comprehensive filter, including three parts: low-pass filter, band trap filter, and zero-phase shift filter, respectively, to cope with three primary disturbances (myoelectric interference, industrial frequency interference, and baseline drift). The filter was designed and its software functionality was initially verified by using Matlab simulation software.

## 2. ECG processing system

### 2.1. ECG processing system model

Physiologically, the ECG signal is a vibration generated by the human heart through a series of mechanical activities such as contraction of the heart muscle, valve closure, and pumping of blood, and is transmitted to the surface of the chest through the chest cavity, chest wall, and other tissues[4]. ECG is an external expression of the physiological information of the heart and is a natural signal in the human body, which is universal, unique, and collectible. Since the human ECG signal is a fragile physiological low-frequency electrical signal, generally, its voltage amplitude does not exceed 5mV and its frequency is generally between 0.05 and 100Hz. The process of measuring the ECG signal is easily affected by a variety of other physiological signals, the environment surrounded by electromagnetic fields, and the signal itself is fragile. In order to solve this problem, we need to analyze the source of noise interference and take appropriate filtering facilities. One solution to this problem is the ECG filter. It is mainly a pre-processing of the ECG signal to filter out the noise signal. The model is roughly shown in Figure 1.

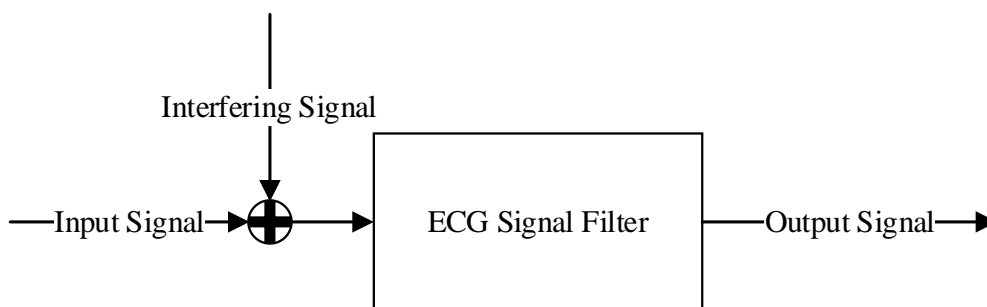


Figure 1: Processing process

### 2.2 Introduction of cardiac filter

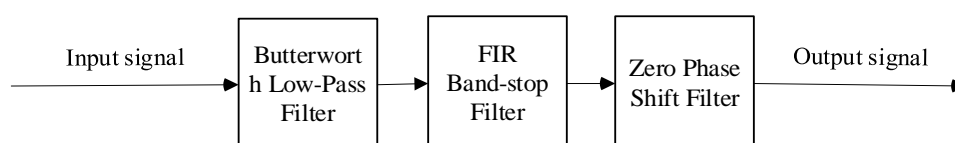


Figure 2: Cardiac filter model

The ECG filter designed in this paper is a hybrid filter, which consists of three primary filters, namely low-pass filter, band trap filter, and zero-phase shift filter, in order to cope with three main interferences (myoelectric interference, industrial frequency interference, and baseline drift).

The cardiac filter composition is shown in Figure 2.

### **2.1.1 Filtering of myoelectric interference - low-pass filter**

Generally speaking, the frequency of EMG signal is 20~5000HZ, and the frequency of its main components is related to the type of muscle, which is usually in 30~300HZ, while the frequency of ECG signal is mainly concentrated in 5~20HZ, so the low-pass filter is chosen to filter out the EMG interference.

The Butterworth filter is an IIR filter characterized by the flattest frequency response curve in the passband with no rise and fall, while in the stopband it gradually decreases to zero. The amplitude diagonal frequency of the Butterworth filter decreases monotonically, and the higher the order of the filter, the faster the amplitude decays in the stopband. The amplitude diagonal frequency diagrams of other filters with higher orders and lower orders have different shapes.

### **2.1.2 Suppression of Frequency Interference - Band -Stop Filters**

Industrial frequency stems due to the ubiquitous supply network, so 50Hz industrial frequency interference is the most common, but also the main source of interference in the ECG signal. 50HZ trap software design methods are diverse, common methods are wavelet transform filtering, adaptive filtering, template matching filtering, etc., but all need to be manually calculated to obtain the parameters of the filter, the operation is more complex. Filter design, the use of IIR filter, can make the order to reduce the number of operations, but the destruction of the phase characteristics; use FIR filter can get both good filtering effect, is the waveform distortion to the lowest, and, FIR filter can be made into a linear phase characteristics, which is exactly what is needed for ECG signal filtering.

MATLAB design FIR filter methods include window function method, frequency sampling method, and Chebyshev approximation method, etc. In this paper, the Kaiser window function method is used to design a 50HZ trap filter. It has the advantages of simultaneous adjustment of the main flap width and the side flap width, smaller out-of-band energy, and greater out-of-band attenuation.

### **2.1.3 Correction of baseline drift - zero phase shift filter**

A zero-phase shift filter is a signal sequence whose phase does not change after filtering by this filter, i.e., the phase response of this filter system function is zero. It is impossible to achieve zero phase shift for a causal system, where the phase spectrum of the signal is not known in advance. Zero phase shift is only possible for non-causal systems. Specifically, the zero-phase shift filter uses the information contained in the signal points before and after the current signal point, essentially using "future information" to eliminate phase distortion.

## **3. MATLAB design cardiac filter**

In this integrated model, three different types of filters are mainly integrated for three different types of ECG noise.

### **3.1 Butterworth Low Pass Filter**

#### **3.1.1 Design Ideas**

The unit response of the IIR digital filter is infinitely long, and the analog filter generally has an infinitely long unit impulse response, so it is matched with the analog filter. That is, the most critical part of the design is to convert the system function  $H(s)$  of the analog filter to  $H(z)$ , and this paper will use the impulse response invariant method to complete this step, that is, to use the tangent transform to complete the frequency transformation ( $\Omega = \frac{2}{T} \tan(\frac{1}{2} \Omega_1 T)$ ,  $T$  is the sampling interval). The following figure shows the general design process of the IIR digital filter[7,10].

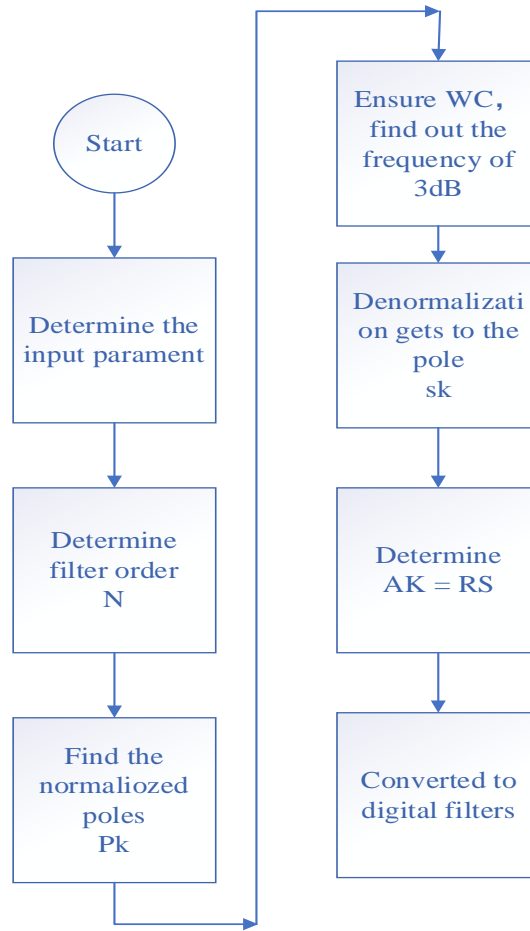


Figure 3: Butterworth digital filter design process

### 3.1.2 Design Process

The squared amplitude function of the Butterworth low-pass filter is  $|H_a(j\Omega)|^2 = \frac{1}{1+(\frac{\Omega}{\Omega_c})^{2N}}$ , where  $N$  is the order of the filter and  $\Omega_c$  is the 3 dB cutoff frequency.

The pole position and order  $N$  of the  $N$ th order Butterworth low-pass filter can be determined by the following equation.

$$\left\{ \begin{array}{l} 1 + \left(\frac{\Omega_p}{\Omega_c}\right)^{2N} = 10^{\alpha_p/10} \\ \left(\frac{\Omega_p}{\Omega_c}\right)^N = \sqrt{\frac{10^{\alpha_s/10} - 1}{10^{\alpha_p/10} - 1}} \\ \lambda_{sp} = \frac{\Omega_s}{\Omega_p} \\ k_{sp} = \sqrt{\frac{10^{\alpha_s/10} - 1}{10^{\alpha_p/10} - 1}} \\ N = \frac{\lg k_{sp}}{\lg \lambda_{sp}} \end{array} \right.$$

In these equations,  $\alpha_p$  denotes the maximum passband attenuation,  $\alpha_s$  denotes the minimum resistive attenuation,  $\Omega_p$  denotes the passband cutoff frequency, and  $\Omega_c$  denotes the resistive cutoff frequency. In this paper, it is 1.4, 1.6, 80, and 100 Hz respectively.

The above equation is designed as an analog filter, which can be converted into a digital filter by the

equation  $S = \frac{2}{T} \frac{1-e^{-s_1 T}}{1+e^{-s_1 T}}$  Which converts it to a digital filter, i.e., the transfer function H(s) is converted to H(z). The amplitude and frequency characteristics of the designed filter are shown in Figure4 below.

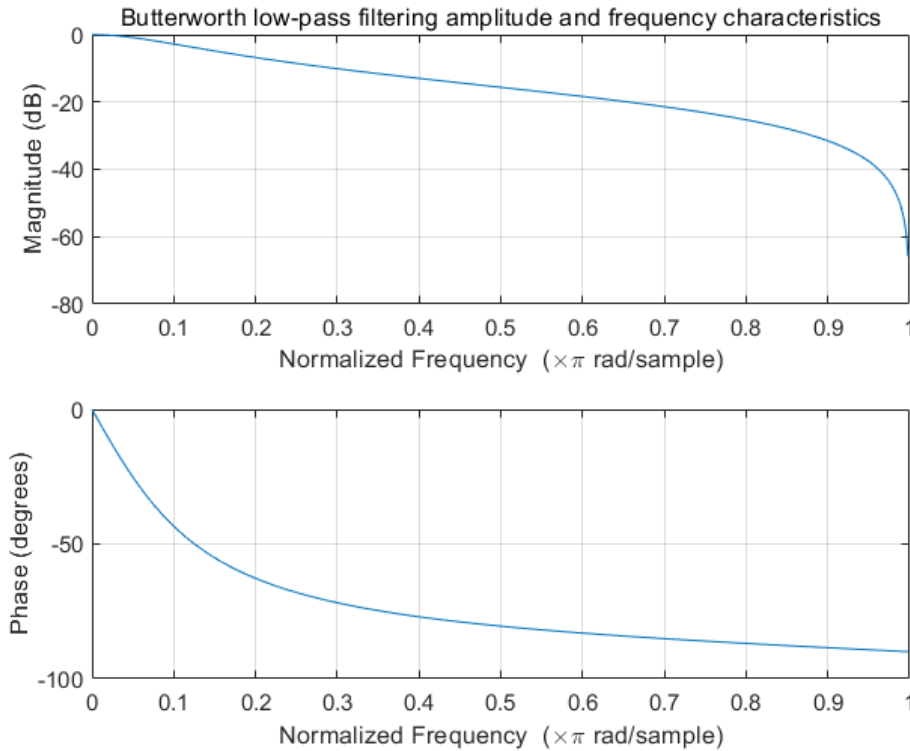


Figure 4: Butterworth low-pass filtering amplitude and frequency characteristics

For a low-pass filter, it simply means that the low frequency or DC component of the signal is allowed to pass, while the high-frequency component above the cutoff frequency and interference noise are suppressed. In this paper, its role is to initially filter out the interference noise in the ECG signal, i.e., the interference of the myoelectric signal. Because the EMG signal is generally 30-300Hz, and the ECG signal is mainly concentrated at 5-25Hz, it can be seen from the above figure that this low-pass filter can filter out most of the EMG interference.

### 3.2 Bandstop filter

#### 3.2.1 Design Ideas

The unit impulse response of the FIR filter is a finite length sequence, let the unit impulse response

The length of h(n) is N, and its system function H(Z) and difference function are :

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

$$y(n) = \sum_{m=0}^{N-1} h(m)x(n-m)$$

As can be seen from the difference function, the FIR filter is of order N-1 and length N. The output is only related to the input function and not to the past output, i.e., the system contains no feedback branches[9]. In practical applications, it is always desirable for a digital filter to have a linear phase, and one of the characteristics of the FIR digital filter is the ability to have a linear phase, i.e.

$$h(n) = \pm h(N-1-n)$$

And this is exactly what is needed for ECG signal filtering.

The main methods for designing FIR filters are window-function method, frequency sampling

method, etc. In this paper, we will use Kaiser window function method to design this filter. In this paper, the Kaiser window function method will be used to design this filter, because among the FIR digital filter design methods, the window function method is simple, practical and easy to implement, and the Kaiser window is an optimal window function.

### 3.2.2 Design Process

The first step is to construct the ideal frequency response function that you wish to approximate  $H_d(e^{j\omega})$ . In this paper the frequency response function is shown in the following equation.

$$H_d(e^{j\omega}) = \begin{cases} e^{-j\omega\tau}, & |\omega| \leq \omega_l, |\omega| \geq \omega_h \\ 0, & \text{else} \end{cases}$$

Then it is to find its unit impulse response  $h_d(n)$ , which is obtained by performing the Fourier inverse transform of the above equation as

$$\begin{aligned} h_d(n) &= \frac{1}{2\pi} \int_{-\pi}^{\pi} H_d(e^{j\omega}) e^{j\omega n} d\omega \\ &= \frac{1}{2\pi} \int_{-\pi}^{\omega_h} e^{j(n-\tau)\omega} d\omega + \frac{1}{2\pi} \int_{-\omega_l}^{\omega_l} e^{j(n-\tau)\omega} d\omega + \frac{1}{2\pi} \int_{\omega_h}^{\pi} e^{j(n-\tau)\omega} d\omega \\ &= \frac{\sin[(n-\tau)\omega_l] + \sin[(n-\tau)\pi] - \sin[(n-\tau)\omega_h]}{\pi(n-\tau)} \end{aligned}$$

Finally, after multiplying it by the window function  $w(n)$  is truncated, the impulse response of the digital filter  $h(n)$  is obtained. the amplitude function of the Kaiser window is

$$w(\omega) = w_k(0) + 2 \sum_{n=1}^{(N-1)/2} w_k(n) \cos(\omega n)$$

Its window function is.

$$w_k(n) = \frac{I_0(\beta)}{I_0(\alpha)}, \quad 0 \leq n \leq N-1$$

Among them  $\beta = \alpha \sqrt{1 - \left(\frac{2n}{N-1} - 1\right)^2}$ ,  $I_0(\beta)$  is the zero-order first class modified Bessel function.

The impulse response  $h(n)$  is.

$$\begin{aligned} h(n) &= h_d(n)w(n) \\ &= \frac{\sin[(n-\tau)\omega_l] + \sin[(n-\tau)\pi] - \sin[(n-\tau)\omega_h]}{\pi(n-\tau)} \cdot \frac{I_0(\beta)}{I_0(\alpha)} \end{aligned}$$

The formula requires  $\tau = (N-1)/2$ ,  $\omega_l = 49\text{Hz}$ ,  $\omega_h = 51\text{Hz}$ .

The characteristics of the designed bandstop filter are shown in Figure 5 below.

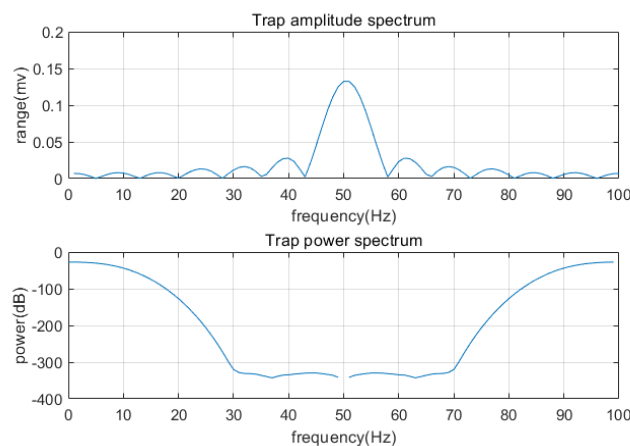


Figure 5: Characteristic diagram of band-Stop filter

### 3.3 Zero Phase Filters

In order to overcome the problem of baseline drift, the zero-phase filtering method can be used. The basic idea of this method is: first determine the initial conditions of the filter, then truncate and extend the first segment of the original sequence, then invert the extended sequence after feeding it into the filter, then pass the result through the filter, and finally invert the result again and remove the first and last extensions to obtain the zero-phase filtered The output sequence is obtained after zero-phase filtering[4,8].

## 4. MATLAB simulation results

### 4.1 Experimental environment

For the ECG data, five sets of data are selected for testing and validation in this paper. The computer configuration is as follows.

OS: Microsoft Windows 10 Home China

Processor name: QuadCore Intel Core i5-9300H, 4000 MHz (40 x 100)

Motherboard Chipset: Intel Cannon Point HM370, Intel Coffee Lake-H

System memory: 16304 MB (DDR4 SDRAM)

DirectX: DirectX 12.0

Software Version: MATLAB2019A

### 4.2 Experimental results and analysis

After data processing and filtering of the ECG signal according to the above method, the ECG signal waveform and baseline drift was improved. The original waveform of the ECG signal after reading is shown in Figure 6.

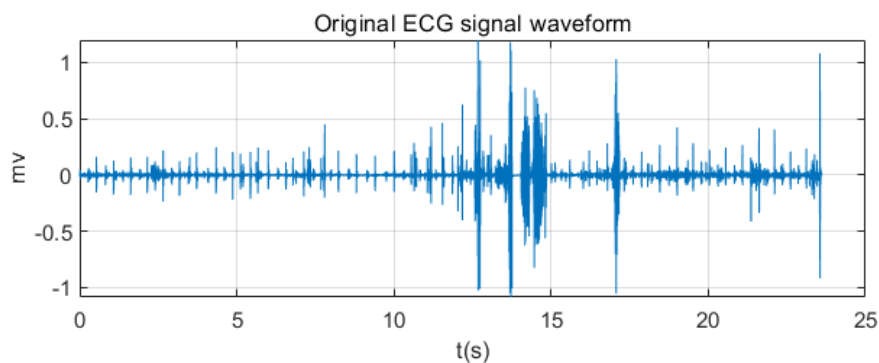


Figure 6: Original waveform of ECG signal

The details are shown in Figure 7, where the obvious rough parts of the waveform can be seen.

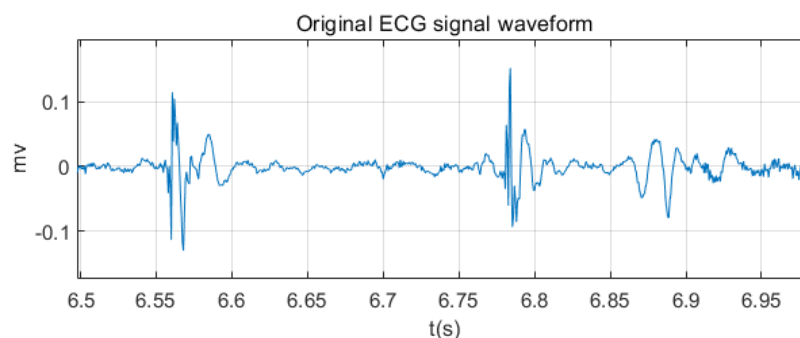


Figure 7: Original waveform of ECG signal (partially enlarged)

The waveform of the ECG signal after low-pass filtering is shown in Figure 8, and the waveform is smoother, with a certain degree of myoelectric interference removed.

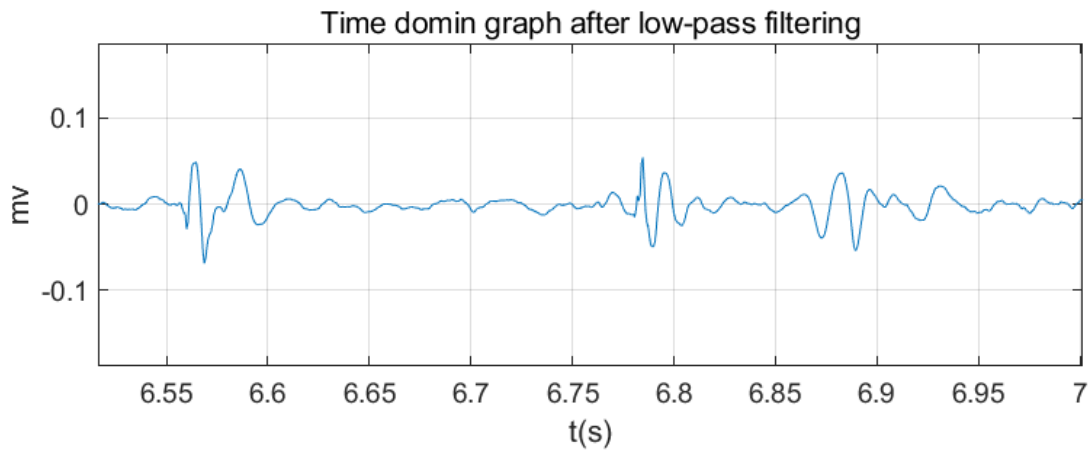


Figure 8: Time-domain waveform after low-pass filtering

The good filtering effect can be seen more clearly in the spectral plot before and after low-pass filtering of the ECG signal.

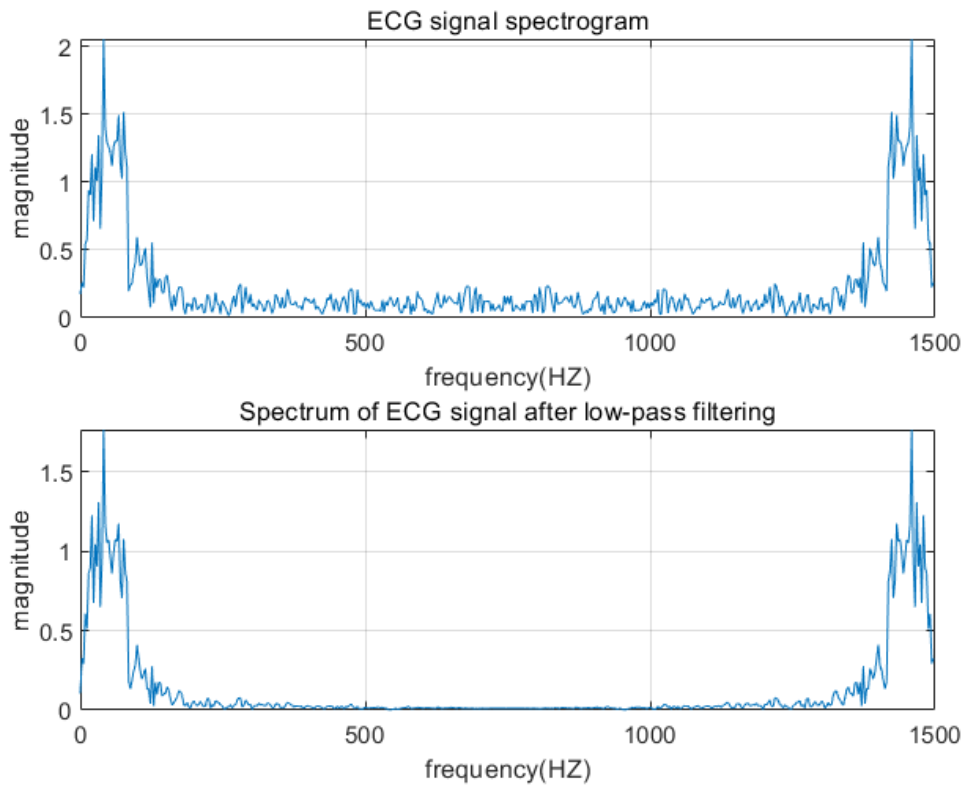


Figure 9: Comparison of pre and post-filtering of ECG signal

The waveform after band-stop filtering to remove the 50 Hz industrial frequency interference generated by the supply network is shown in Figure 10, which shows a more regular signal compared to the low-pass filtered waveform.



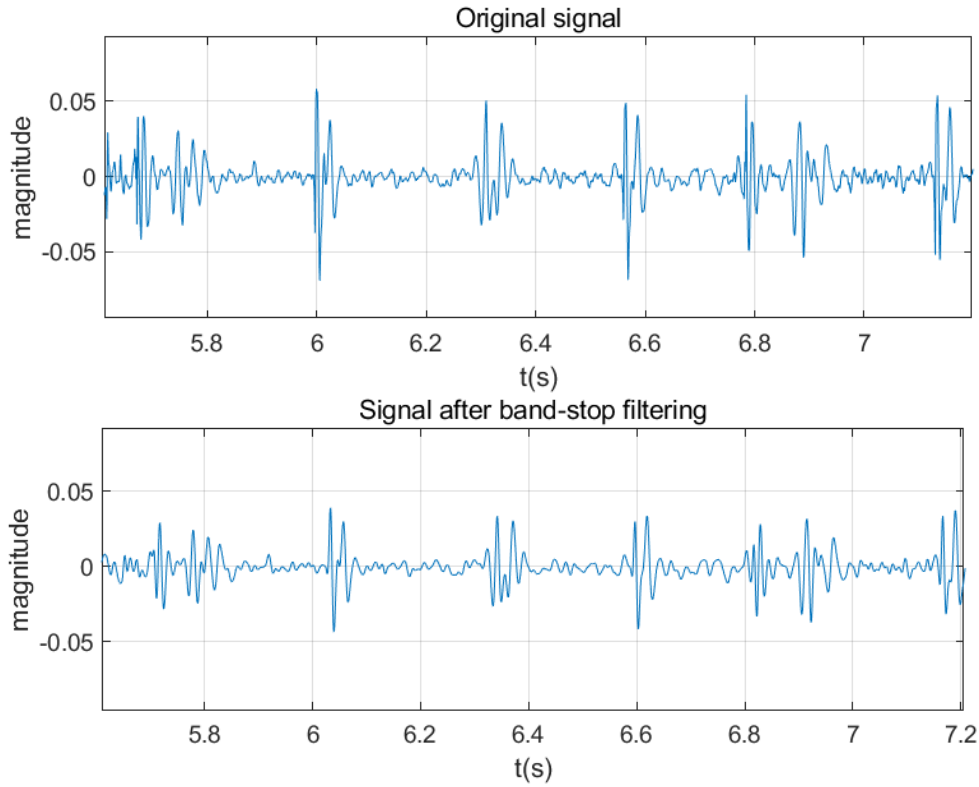


Figure 10: Signal waveform after band-stop filtering (partial enlargement)

For the problem of baseline drift in some signals, the waveforms before and after using zero-phase shift filtering to correct the baseline drift are shown in Figure 11.

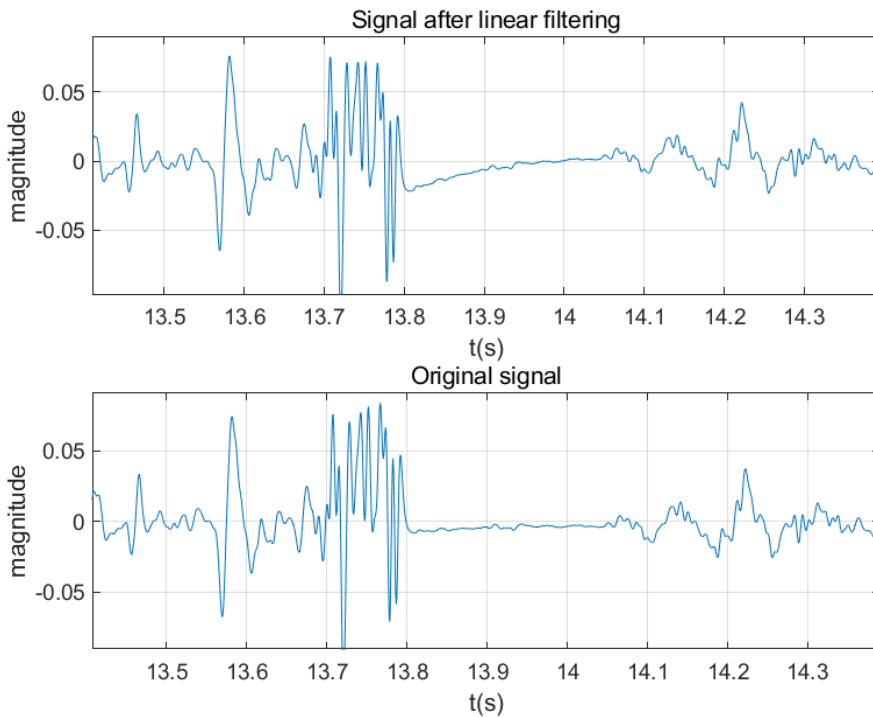


Figure 11: Pre- and post-filtering comparison of ECG signals

The final filtering results show that the myoelectric interference and the industrial frequency interference are filtered out with good effect, and the baseline drift is corrected. The filtering effect of the ECG filter designed in this paper is good and has some practical value.

## 5. Conclusion

The integrated digital filters designed in this paper include Low pass filter, Band-Stop filter and Zero-Phase filter. The Low pass filter selects Butterworth filter to remove the low frequency EMI. A 50Hz Band-Stop Filter is designed by using Kaiser function. Finally, the Zero-phase filter is used to overcome the baseline drift problem. The integrated digital filter was simulated by MATLAB, 10 groups of data were tested and 100 groups of data were verified. The filter effectively reduces electromechanical interference, power frequency interference and baseline drift that ECG signals are prone to. The signal obtained by the filter is improved obviously. The filtering results show that the filter has practical value.

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