INDIAN INSTITUTE OF TECHNOLOGY KHARAGPUR

Department of Electronics & Electrical Communication Engineering
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EC60064: Biomedical System Engineering and Automation

Experiment No.1

Submitted by

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Contents

Introduction

The process or device used for filtering a signal from unwanted component is termed as a filter. To reduce the background noise and suppress the interfering signals by removing some frequencies is called as filtering. There are various types of filters which are classified based on various criteria such as linearity (linear or non-linear), time (time variant or time invariant), Analog or digital, active or passive, and so on. Some of the linear continuous time filters are Chebyshev filter, Bessel filter, Butterworth filter, and Elliptic filter.

The signal processing filter which is having a flat frequency response in the passband can be termed as Butterworth filter and is also called as a maximally flat magnitude filter. There are various types of Butterworth filters such as low pass Butterworth filter and digital Butterworth filter.

Butterworth filter had a reputation for solving very complex mathematical problems thought to be 'impossible'.. Butterworth stated that "An ideal electrical filter should not only completely reject the unwanted frequencies but should also have uniform sensitivity for the wanted frequencies". Such an ideal filter cannot be achieved, but Butterworth showed that successively closer approximations were obtained with increasing numbers of filter elements of the right values At the time, filters generated substantial ripple in the passband, and the choice of component values was highly interactive. Butterworth showed that a low-pass filter could be designed whose cutoff frequency was normalized to 1 radian per second and whose frequency response (gain) was

$$G(\omega) = \frac{1}{\sqrt{1+\omega^{2n}}},$$

Where ω is the angular frequency in radians per second and n is the number of poles in the filter—equal to the number of reactive elements in a passive filter. If $\omega = 1$, the amplitude response of this type of filter in the passband is $1/\sqrt{2} \approx 0.7071$, which is half power or −3 dB.

The frequency response of the Butterworth filter is maximally flat (i.e. has no ripples) in the passband and rolls off towards zero in the stopband.[2] When viewed on a logarithmic Bode plot, the response slopes off linearly towards negative infinity. A first-order filter's response rolls off at −6 dB per octave (−20 dB per decade) (all first-order lowpass filters have the same normalized frequency response). A second-order filter decreases at −12 dB per octave, a third-order at −18 dB and so on. Butterworth filters have a monotonically changing magnitude function with ω, unlike other filter types that have non-monotonic ripple in the passband and/or the stopband.

ABOUT EXPERIMENTS:

1. **PROMBLEM STATEMENT:**

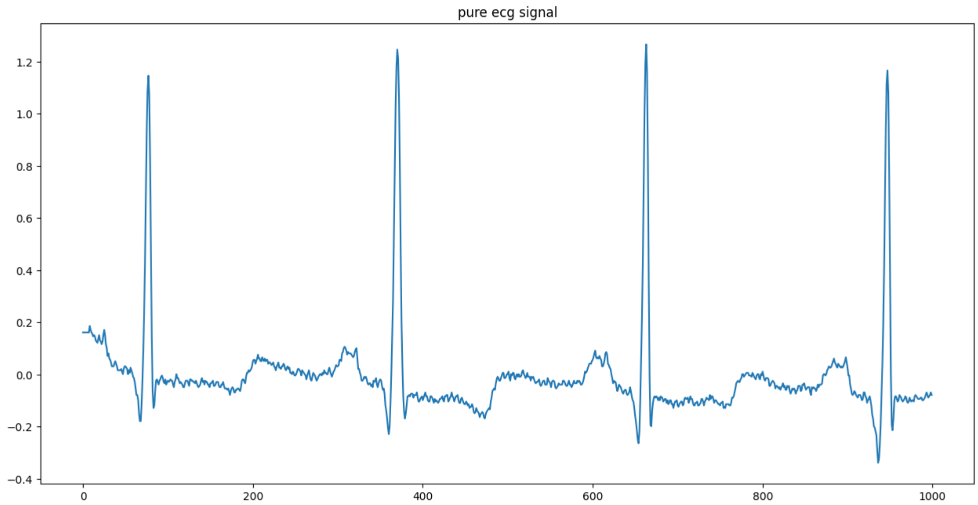
   The data file data_ecg_original.mat contains ECG signal, sampled at 300Hz. Add a powerline interference noise of 61 Hz in the original ECG signal. (Hint: For the powerline interference noise, generate a 61 Hz sinusoid signal with amplitude equal to 0.15 sampled at 300 Hz of the same duration as the original ECG signal.)

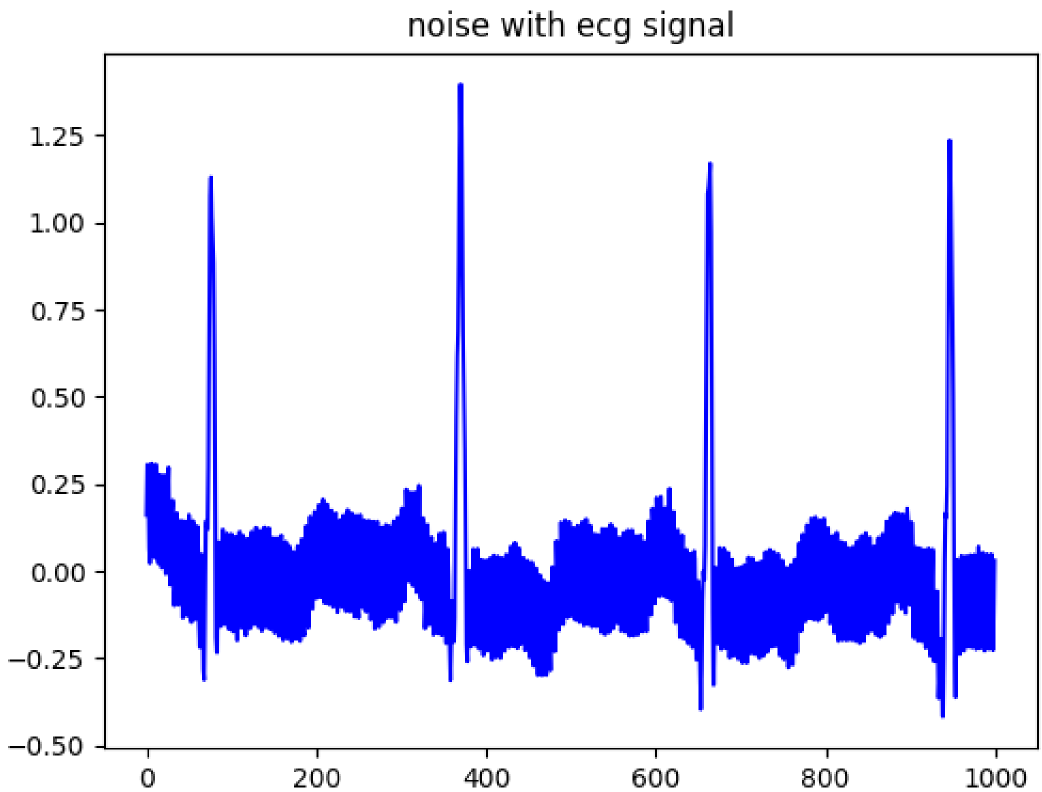
   a. Design a notch filter with two zeros to remove the artifact and implement it in Python.
   b. Add two poles at the same frequency as those of zeros, but with a radius that is less than unity. Study the effect of poles on the output of the filter as their radius is varied from 0.8 to 0.95.

2. **ALGORITHM:**

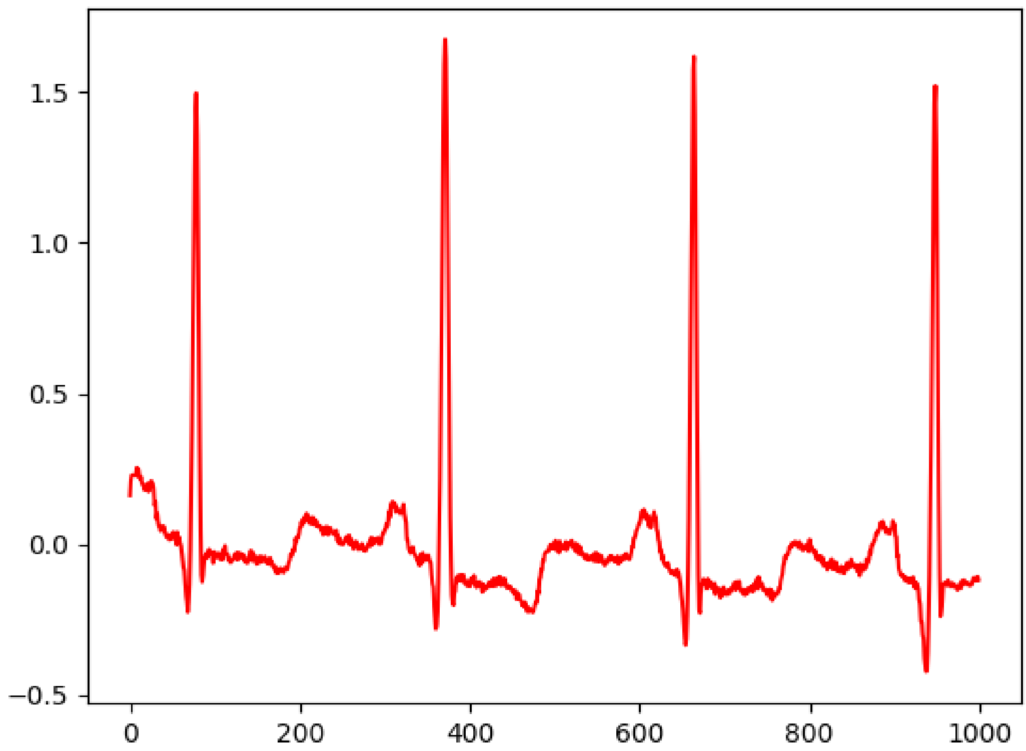
   i. First, the data is loaded from the mat format to the memory
   ii. The data was available in the vector named as ecg_original
   iii. Since the data was in row vector format it was transformed to column vector for further processing
   iv. A power noise with amplitude 0.15 and frequency 61 Hz was generated with sampling frequency 300 Hz
   v. It was added with original signal and stored in new variable named as n_s
   vi. Then a Notch Filter was designed with zeroes at 61 Hz
   vii. And then, the noisy signal was passed through the Notch filter, the output was taken in spatial domain and noisefree signal was generated and stored in variable y
   viii. Then with the original Notch Filter 2 new poles were added with the user given input with radius from 0.8 to 0.95
   ix. Then the noisy signal was passed through the new filter and response were compared

3. **OUTPUT**

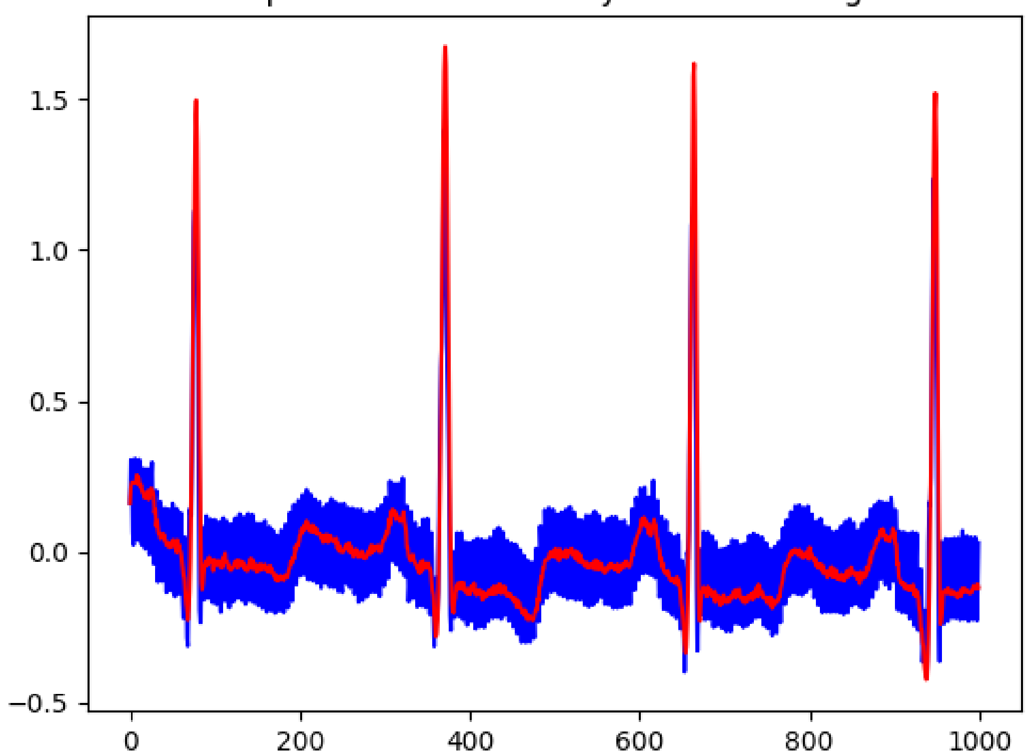

pure ecg signal
1.2
1.0
0.8
0.6
0.4
0.2
0.0
−0.2
−0.4
0
200
400
600
800
1000


noise with ecg signal
1.25
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
0
200
400
600
800
1000

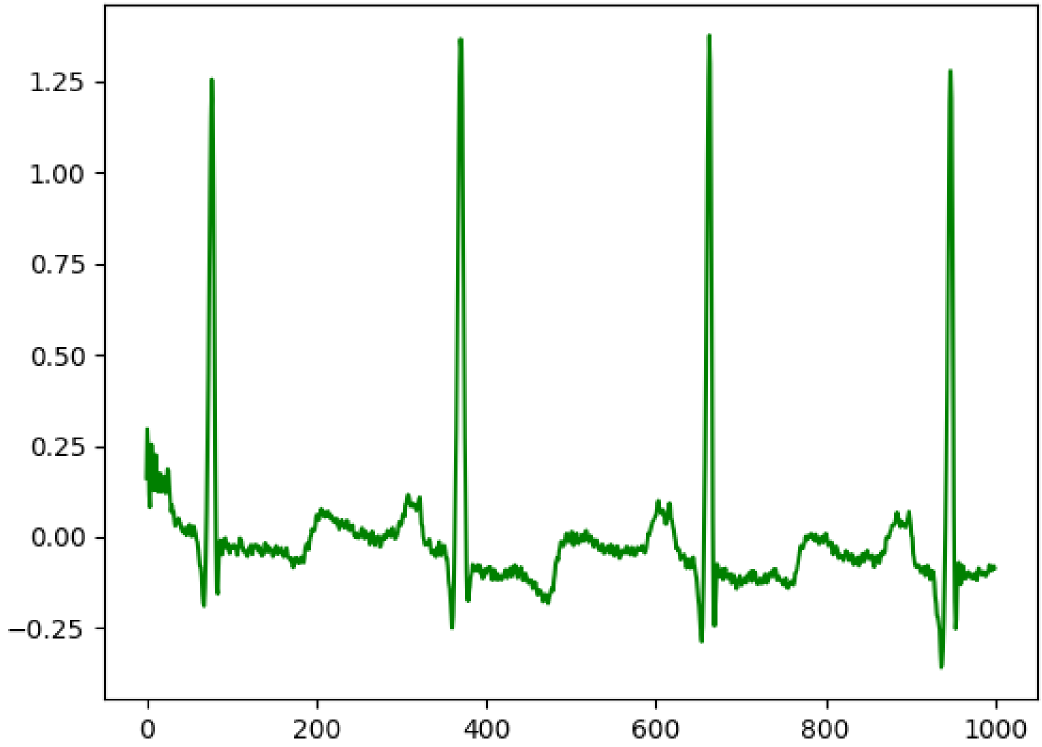
filtered output only with notch filter



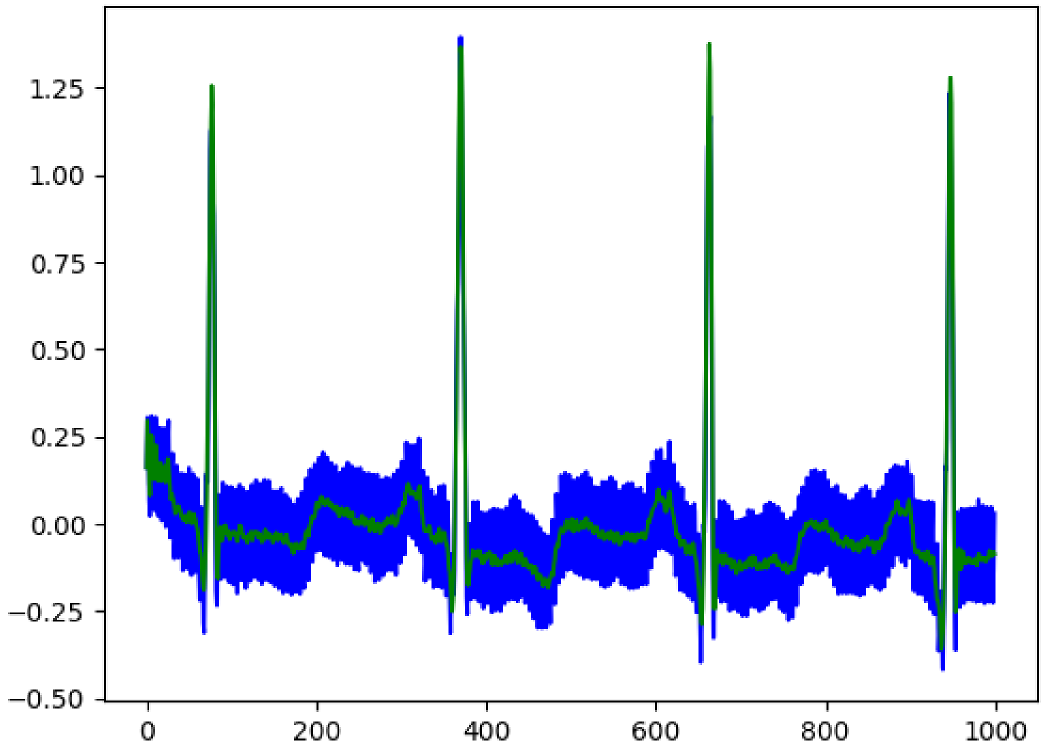
comparison between noisy and filtered signal

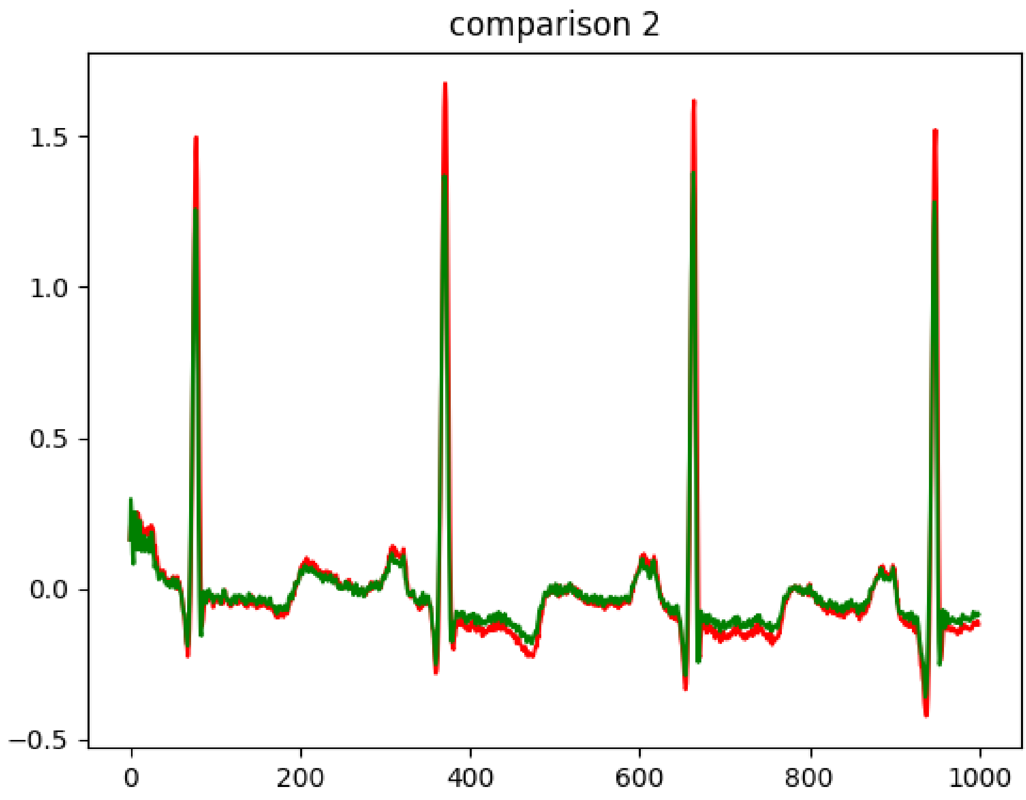


filtered output with zeroes and poles



comparison 1





4. **Observation:**
    i. The filtered output from the Notch filter was almost same as noiseless ECG Signal
    ii. When we include the poles, with poles closer to 0.95 value there are more ripples and the system was going towards unstability. Whereas, when the radius was near to 0.8 there were less oscillations

5. **PROMBLEM STATEMENT:**

Generate the AWGN channel noise with 0.5dB and Add with in data_ece_original.mat and filter this noisy ECG signal using four different Butterworth low pass filters (individually) realized through Python with the following characteristics:

i. Order 6, cutoff frequency 30Hz;
ii. ii. Order 6, cutoff frequency 40Hz;
iii. iii. Order 8, cutoff frequency 50Hz;
iv. iv. Order 8, cutoff frequency 70Hz;

a. Compute the SNR improvement for each filter in (i) – (iv).
b. Filter the original ECG signal in data_ecg_original.mat using Butterworth filters (i) – (iv) and compute the Signal-to-Distortion Ratio (SDR) in each case.

6. **ALGORITHM:**

i. First, the data is loaded from the mat format to the memory
ii. The data was available in the vector named as ecg_original
iii. Since the data was in row vector format it was transformed to column vector for further processing
iv. Input signal power is calculated and then a noise signal with 0.5 db SNR is created using

$$A = \sqrt{\frac{P_{noise}}{var(w(t))}}$$

$$P_{noise} = P_{signal} 10^{\frac{-SNR}{10}}$$

v. Then original snr of the signal is calculated
vi. Then the noisy signal was passed through the low pass butterworth filter with various cutoff frequency and order one by one
vii. The improvement in SNR is calculated using

$$Improvement = SNR_{new} - SNR_{original}$$

Where,

$$SNR_{new} = 10\frac{var(d(t))}{var(v(t))}$$

Where d(t) and v(t) is desired output and the filtered output.
viii. To calculate variance statistics.variance()

7. **OUTPUT**

SNR of input signal (before filtering) : 0.5148643125765241
Improvemnet with fc = 30 and n= 6 : 55.92618532213332
SDR: = 56.44104963470985
Improvemnet with fc = 40 and n= 6 : 50.56369950082093
SDR: = 51.07856381339746
Improvemnet with fc = 50 and n= 8 : 41.33930287411246
SDR: = 41.85416718668899
Improvemnet with fc = 70 and n= 8 : 29.711130884679612
SDR: = 30.225995197256136

8. **Observation:**

i. The SNR value decreases with increase in cutoff frequency as with increase with cutoff frequency, more and more high frequency noise is getting attached with original signal
ii. With increase in cutoff frequency the distortion is decreacing as higher frequency band signal are getting allowed to mix with original signal and hence high harmonic distortion is reducing

9. **PROMBLEM STATEMENT:**

The ECG signal in the file ecg_noisy_high.mat has a wandering base-line (low-frequency artifact).

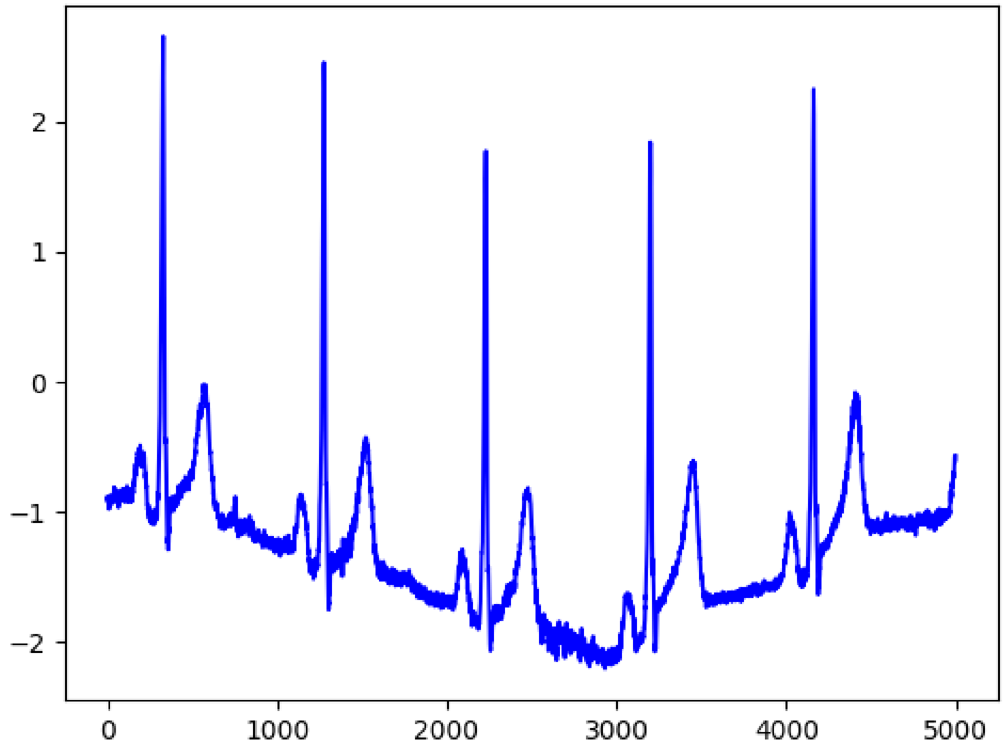
a. Filter the signal with the three point central difference operator given below to remove base-line drift in the ECG signal and plot the results. $(n) = 1\ 4\ [x(n) - x(n - 2)]$

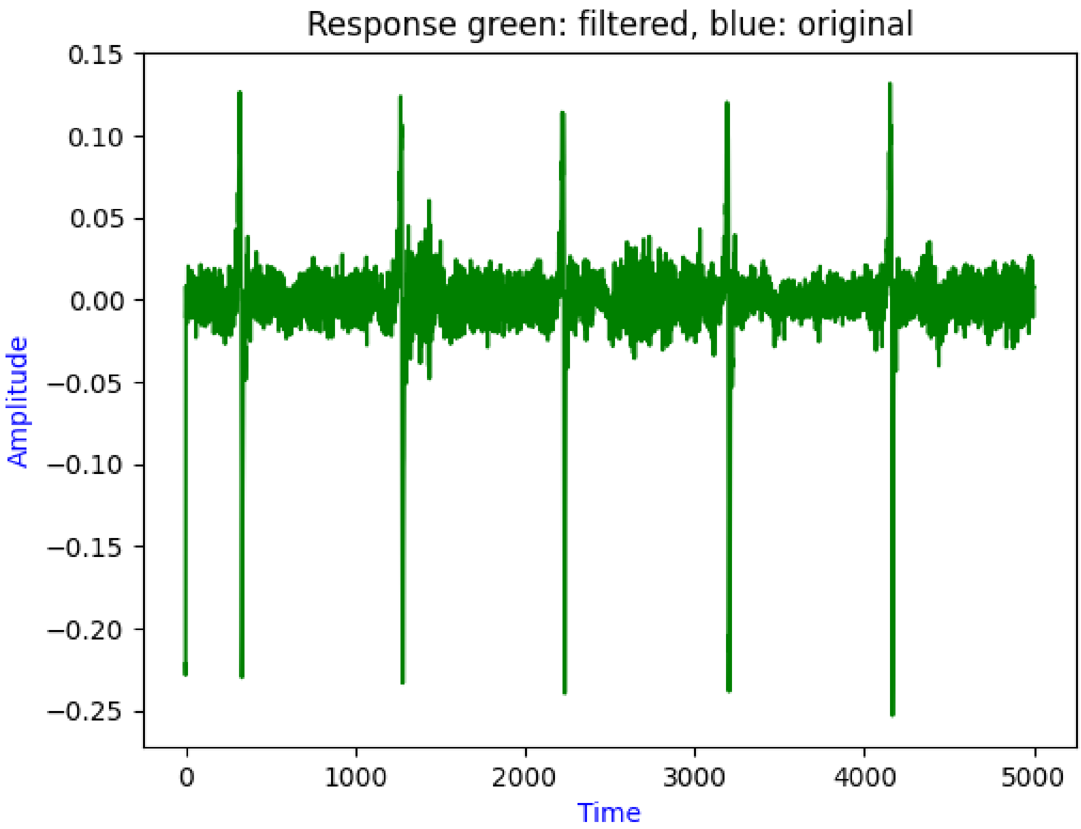
b. Filter the signal in the file ecg_noisy_high.mat using Butterworth highpass filters (i) – (iii) and plot the results.
i. Order 3 , cutoff frequency 3Hz
ii. Order 5 , cutoff frequency 5 Hz
iii. Order 8, cutoff frequency 8 Hz

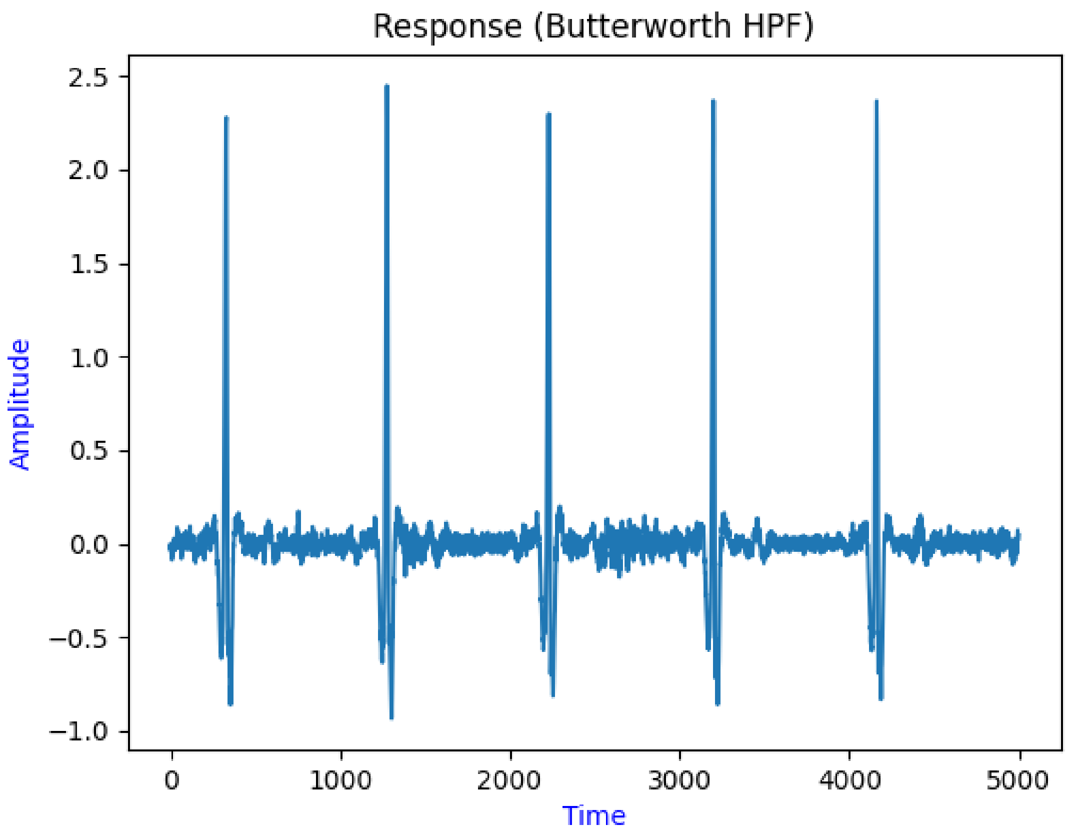
10. **ALGORITHM:**

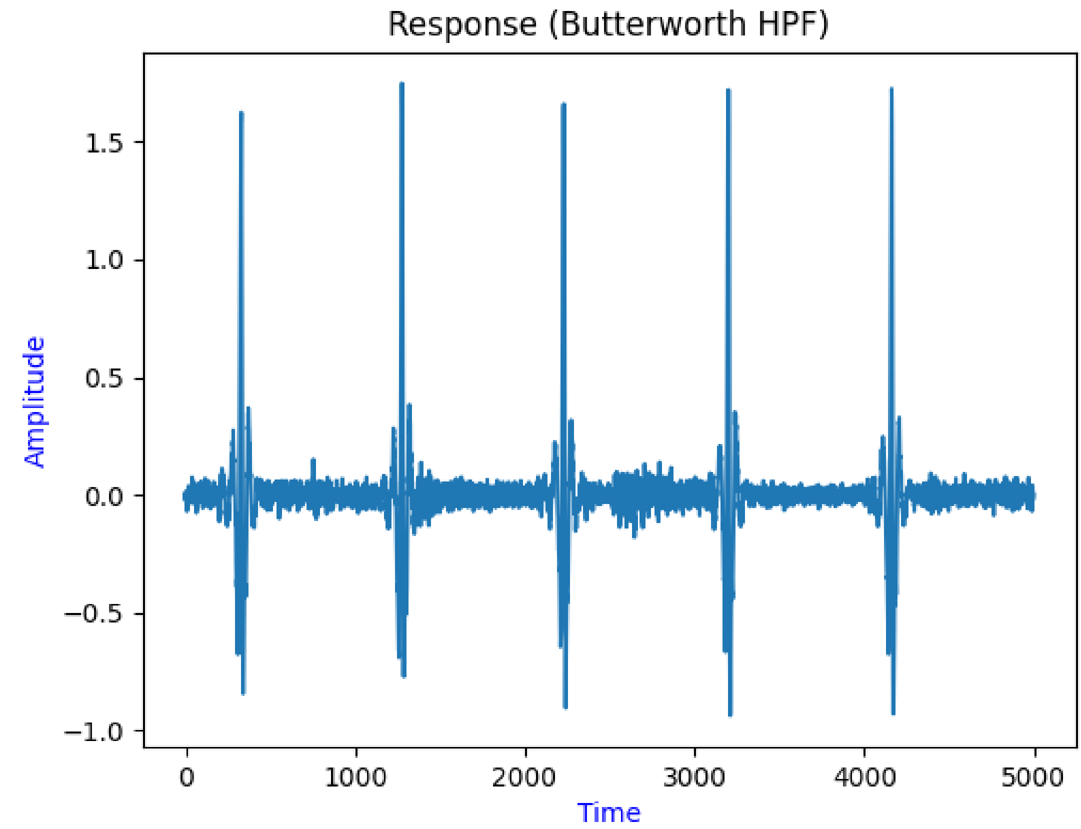
i. First, the data is loaded from the mat format to the memory
ii. The data was available in the vector named as ecg
iii. Then the signal was passed through three point central difference operator, in spatial domain and the output is stored in variable y and the result is plotted
iv. The signal then was passed through High pass filter Butterworth filter with different cutoff frequency and order
v. The sliced result is plotted

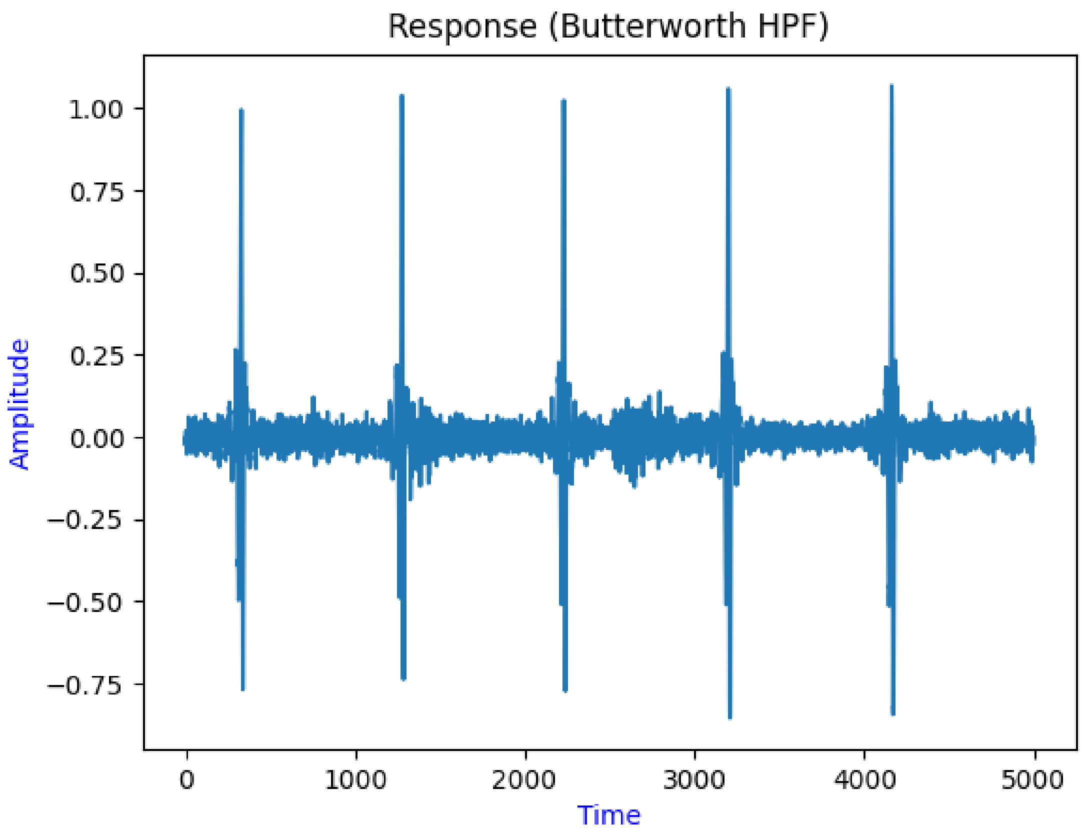
11. **OUTPUT**




Response green: filtered, blue: original
Amplitude
Time
0.15
0.10
0.05
0.00
−0.05
−0.10
−0.15
−0.20
−0.25
0
1000
2000
3000
4000
5000


Response (Butterworth HPF)
Amplitude
Time
2.5
2.0
1.5
1.0
0.5
0.0
−0.5
−1.0
0
1000
2000
3000
4000
5000


Response (Butterworth HPF)
Amplitude
Time
1.5
1.0
0.5
0.0
−0.5
−1.0
0
1000
2000
3000
4000
5000


Response (Butterworth HPF)
Amplitude
Time
1.00
0.75
0.50
0.25
0.00
−0.25
−0.50
−0.75
0
1000
2000
3000
4000
5000

12. **Observation:**

i. The base line drift is reduced by both type of filter
ii. With increase in cutoff frequency and order of the filter more and more noisy signal appeared near transition point, this is due to removal of low frequency component from the signal.