INSTITUTE OF ENGINEERING

ADVANCED COLLEGE OF ENGINEERING & MANAGEMENT
KUPONDOLE, LALITPUR
(AFFILIATED TO TRIBHUVAN UNIVERSITY)

LAB REPORT

LAB NO: 05
SUBJECT: DSAP

SUBMITTED BY:
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ROLL NO: ACE074BCT063
DATE: 2021/08/06

SUBMITTED TO:
DEPARTMENT OF COMPUTER AND ELECTRONICS

TITLE: DESIGNN OF IIR DIGITAL FILTERS

OBJECTIVE: To design and observe IRR filters

THEORY:

IIR filters are digital filters with infinite impulse response. Unlike FIR filters, they have the feedback (a recursive part of a filter) and are known as recursive digital filters therefore. Block diagrams of FIR and IIR filters

For this reason IIR filters have much better frequency response than FIR filters of the same order. Unlike FIR filters, their phase characteristic is not linear which can cause a problem to the systems which need phase linearity. For this reason, it is not preferable to use IIR filters in digital signal processing when the phase is of the essence.

Otherwise, when the linear phase characteristic is not important, the use of IIR filters is an excellent solution.

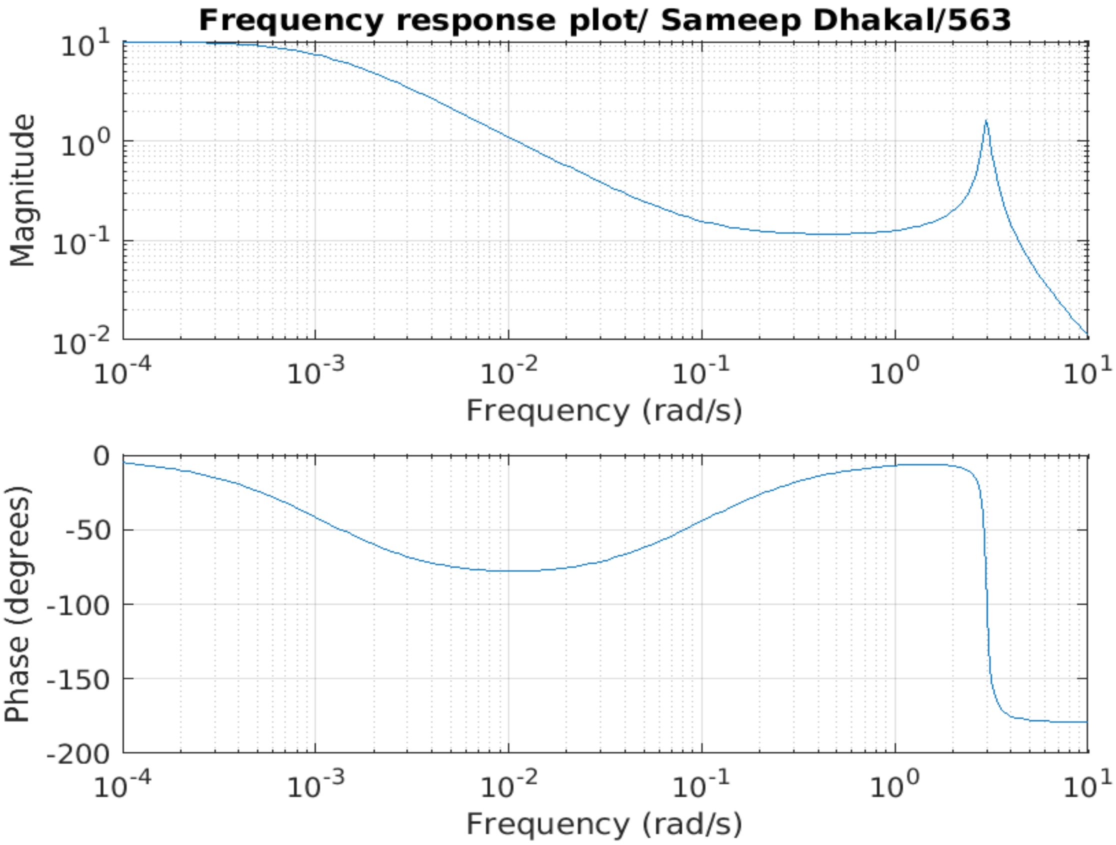
There is one problem known as a potential instability that is typical of IIR filters only. FIR filters do not have such a problem as they do not have the feedback. For this reason, it is always necessary to check after the design process whether the resulting IIR filter is stable or not.

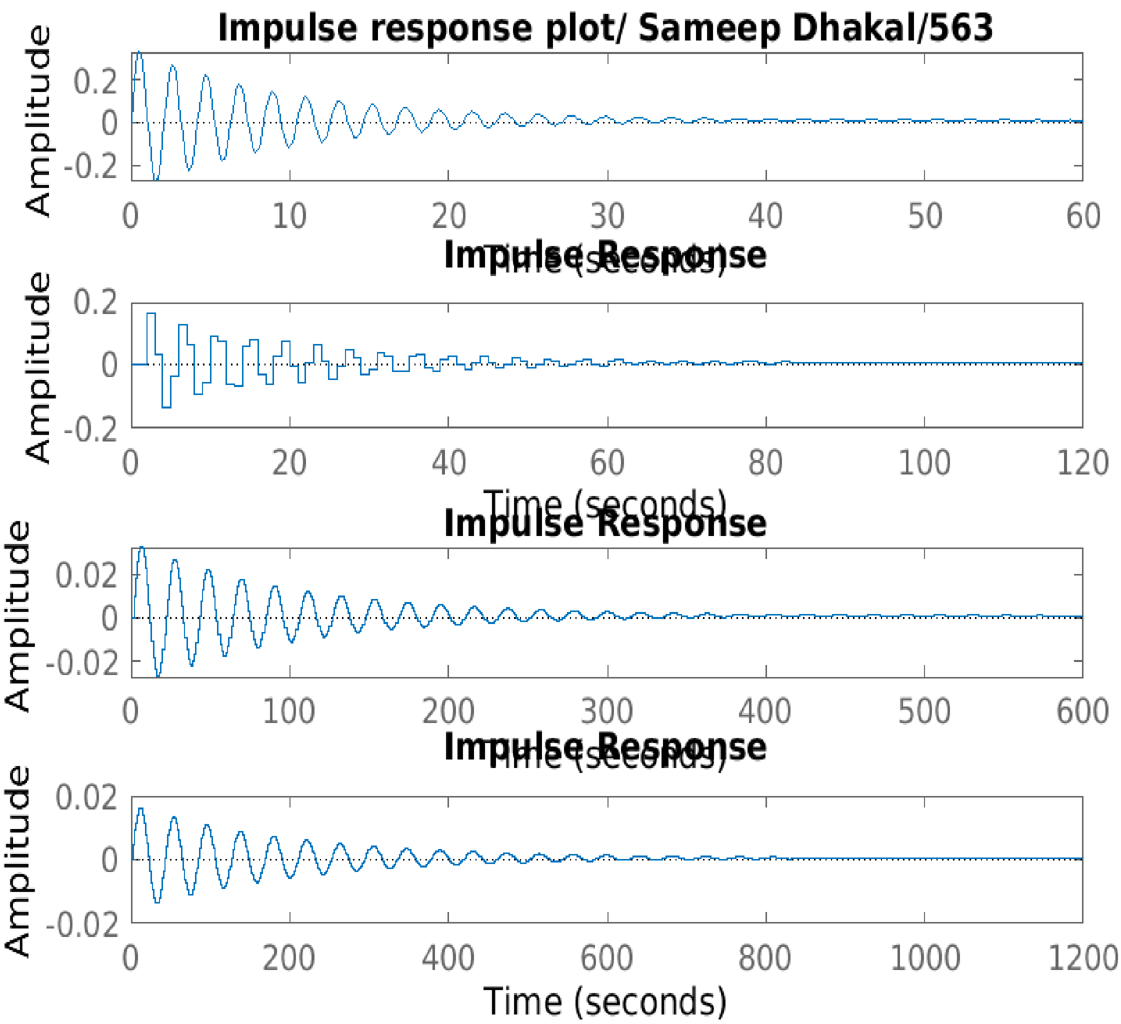
IIR filters can be designed using different methods. One of the most commonly used is via the reference analog prototype filter. This method is the best for designing all standard types of filters such as low-pass, high-pass, band-pass and band-stop filters.

LAB TASK:

1. **H(s)=(s+0.1)/((s+0.1) 2 +9)**
   a. **Impulse invariance**
      i. Convert the analog filter into a digital IIR filter by means of impulse invariance method.
      ii. the frequency response in s-domain and z-domain.
      iii. Plot the impulse response of the LTI model.
      iv. Plot the impulse response in z-domain.

```
clc;
clear all;
close all;
b= [1 .1];
a= [1 .2 9 .01];
figure;
freqs(b,a);
title('Frequency response plot/ Sameep Dhakal/563');
figure;
subplot(4,1,1);
impulse(b,a);
title('Impulse response plot/ Sameep Dhakal/563');
[bz,az]= impinvar(b,a,2);
subplot(4,1,2);
dimpulse(bz,az);
[bz,az]= impinvar(b,a,10);
subplot(4,1,3);
dimpulse(bz,az);
[bz,az]= impinvar(b,a,20);
subplot(4,1,4);
dimpulse(bz,az);
```

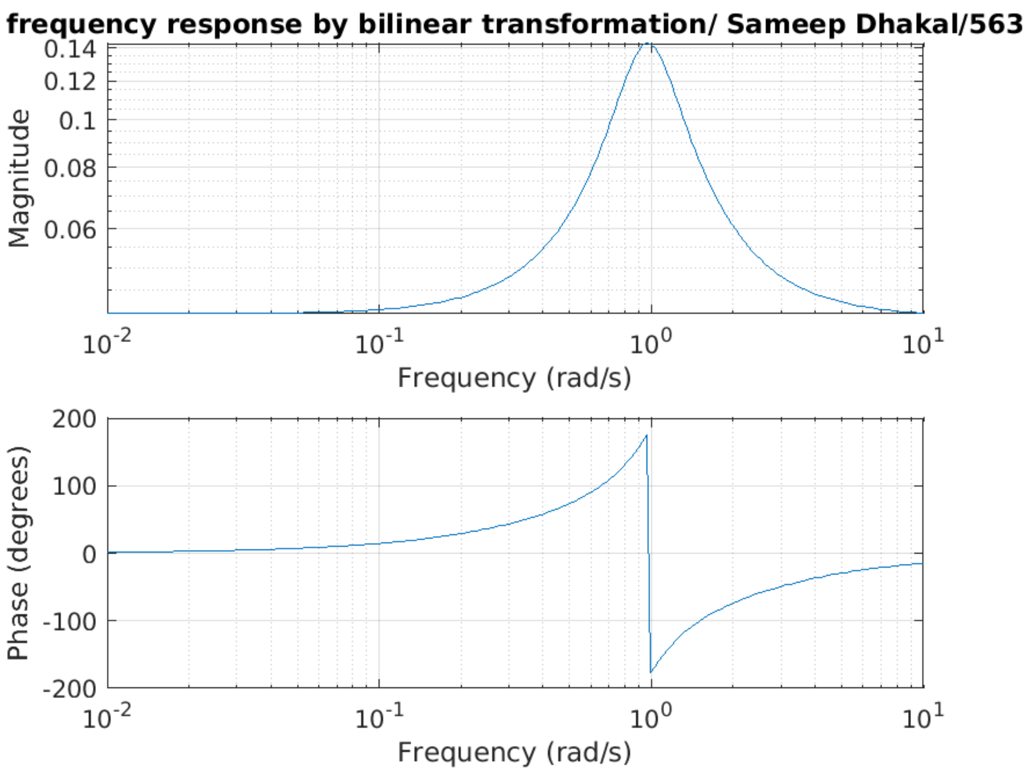

Frequency response plot/ Sameep Dhakal/563
Magnitude
Frequency (rad/s)
Phase (degrees)
Frequency (rad/s)

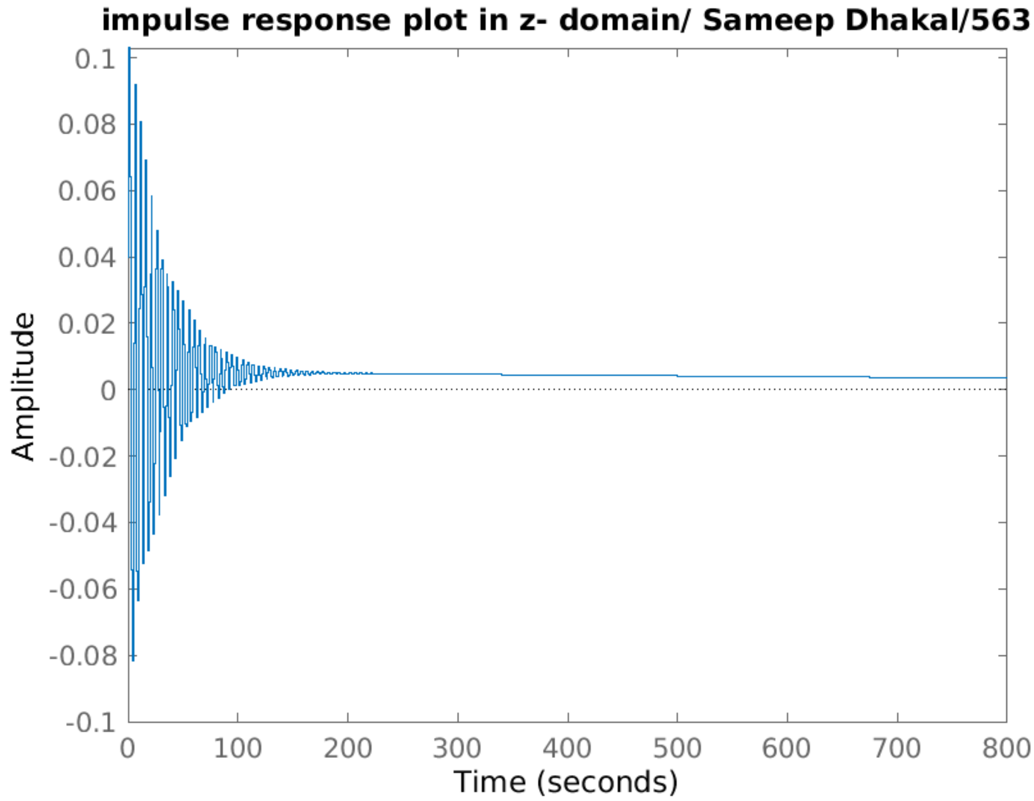

Impulse response plot/ Sameep Dhakal/563
Amplitude
Impulse Response
Amplitude
Time (seconds)
Impulse Response
Amplitude
Impulse Response
Amplitude
Time (seconds)

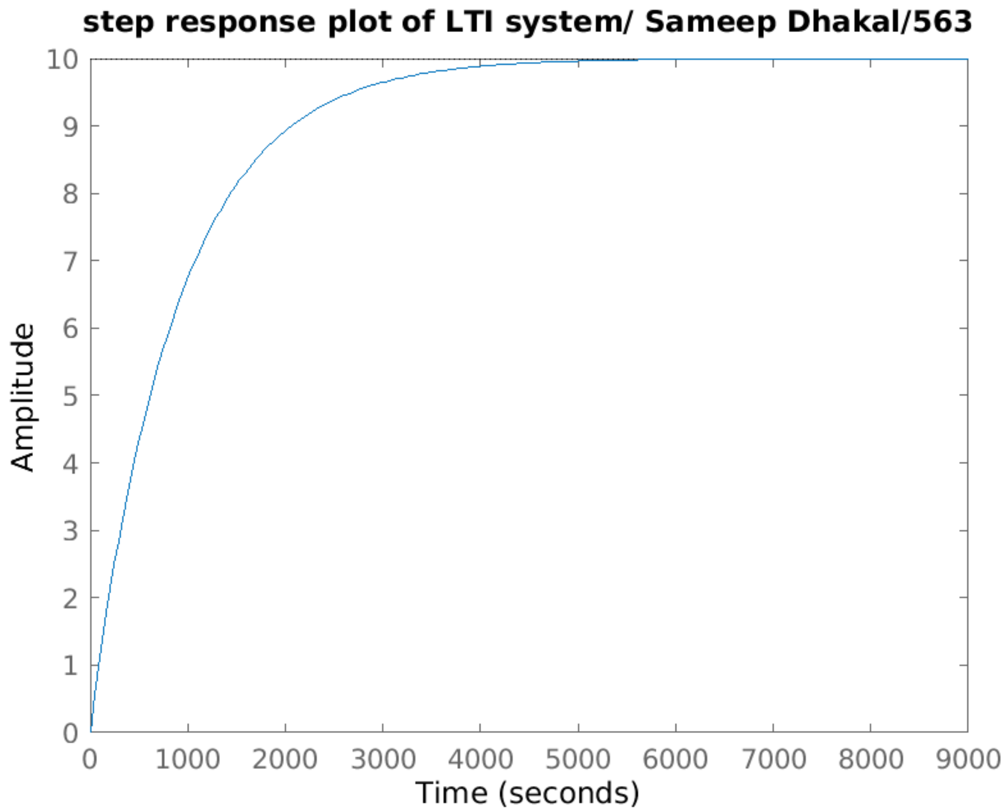
b. Bilinear Transformation

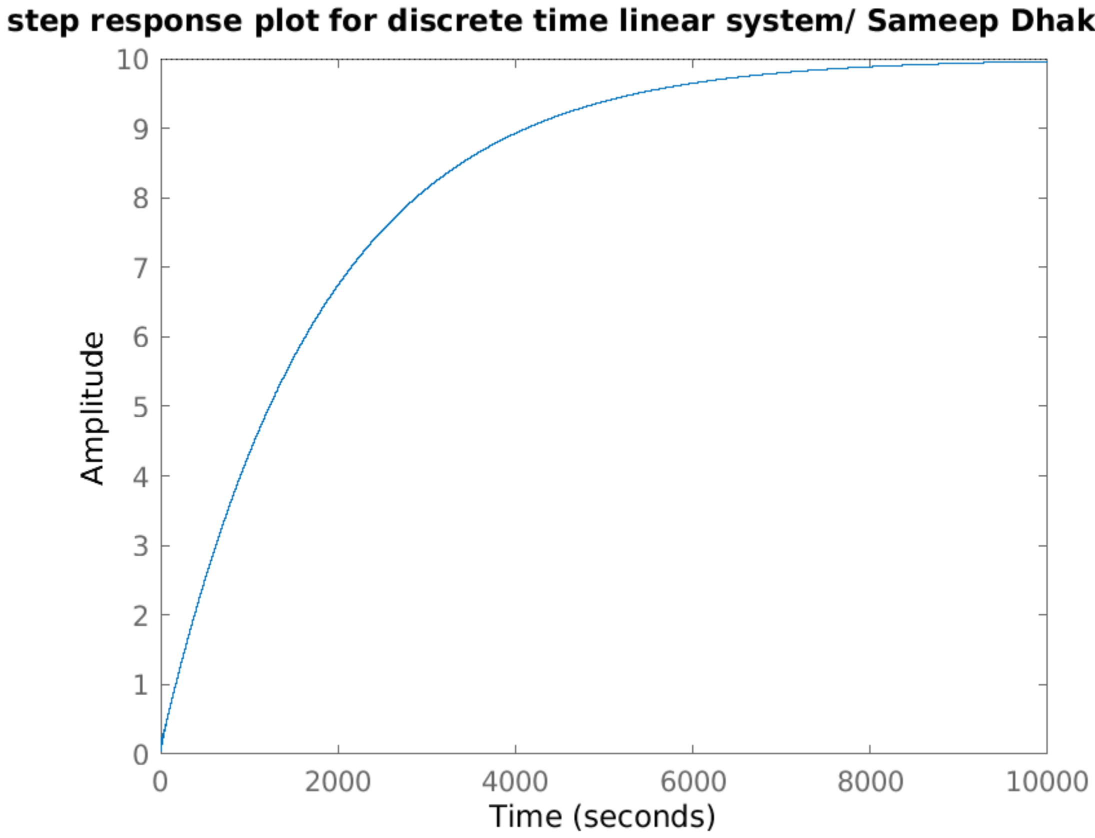
i. Convert the analog filter into a digital IIR filter by means of bilinear transformation method.
ii. Plot the frequency response of the transformed filter by bilinear transformation
iii. Plot the impulse response in z-domain.
iv. Plot the step response of the LTI model.
v. Step response of discrete time linear systems.

```
clc;
clear all;
close all;
b= [1 .1];
a= [1 .2 9 .01];
[bz,az]= bilinear(b,a,2);
freqs(bz,az);
title('frequency response by bilinear transformation/ Sameep Dhakal/563');
figure;
dimpulse(bz,az);
title('impulse response plot in z- domain/ Sameep Dhakal/563');
figure;
sys=tf(b,a);
step(sys);
title('step response plot of LTI system/ Sameep Dhakal/563');
figure;
dstep(bz,az);
title('step response plot for discrete time linear system/ Sameep Dhakal/563');
```




impulse response plot in z- domain/ Sameep Dhakal/563
Amplitude
Time (seconds)
0.1
0.08
0.06
0.04
0.02
0
-0.02
-0.04
-0.06
-0.08
-0.1
0
100
200
300
400
500
600
700
800


step response plot of LTI system/ Sameep Dhakal/563
Amplitude
Time (seconds)
10
9
8
7
6
5
4
3
2
1
0
0
1000
2000
3000
4000
5000
6000
7000
8000
9000

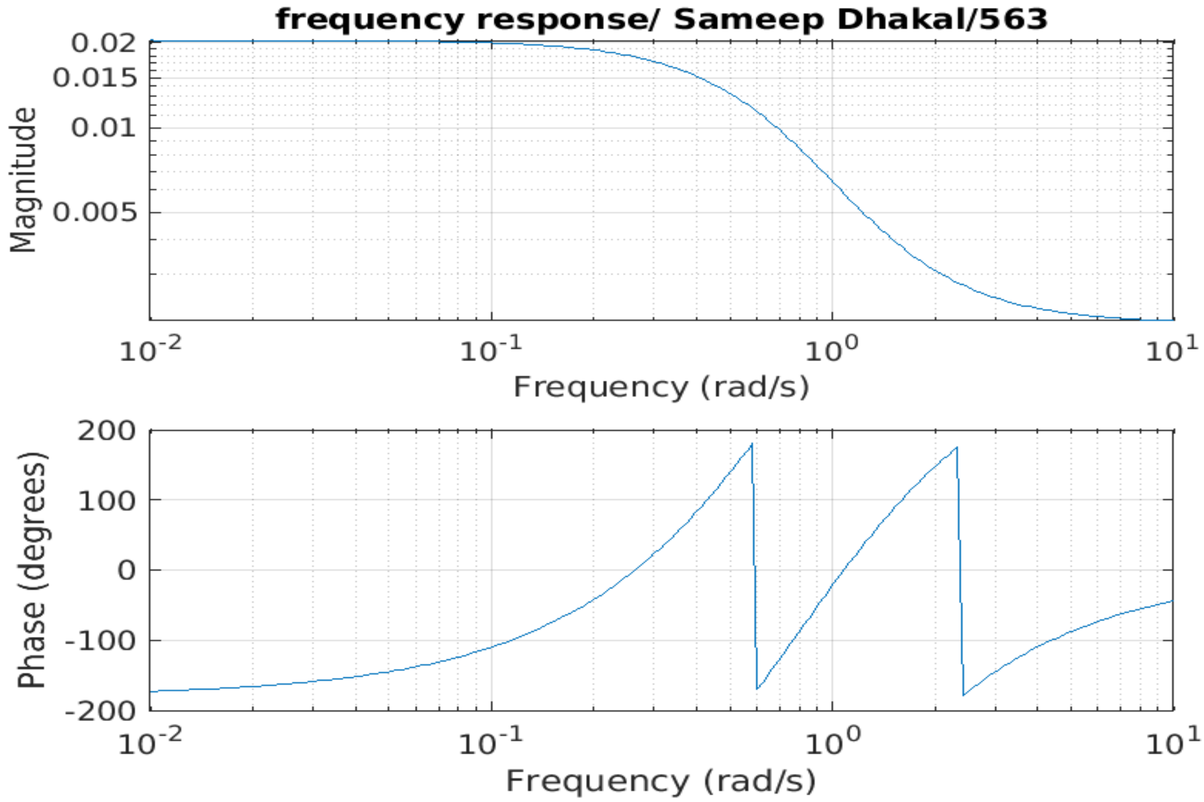

step response plot for discrete time linear system/ Sameep Dhakal/5
Amplitude
Time (seconds)
0
1
2
3
4
5
6
7
8
9
10
0
2000
4000
6000
8000
10000

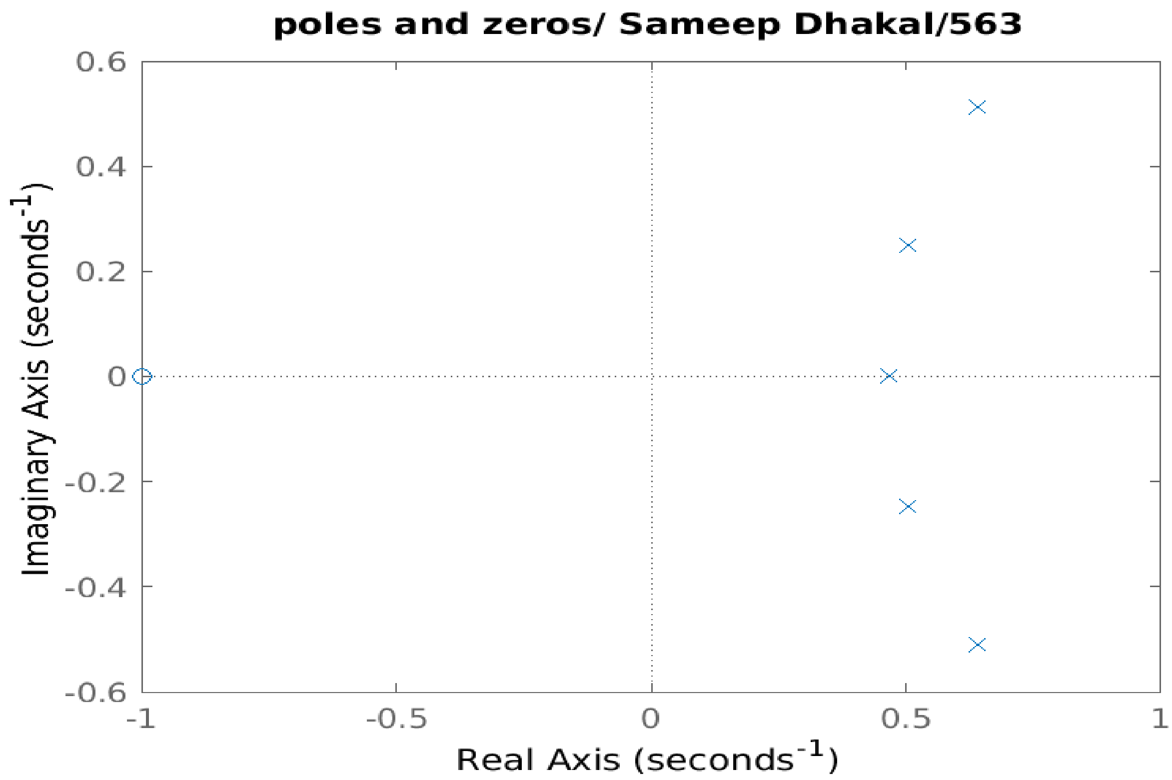
2. **Design of Butterworth Lowpass Filter**

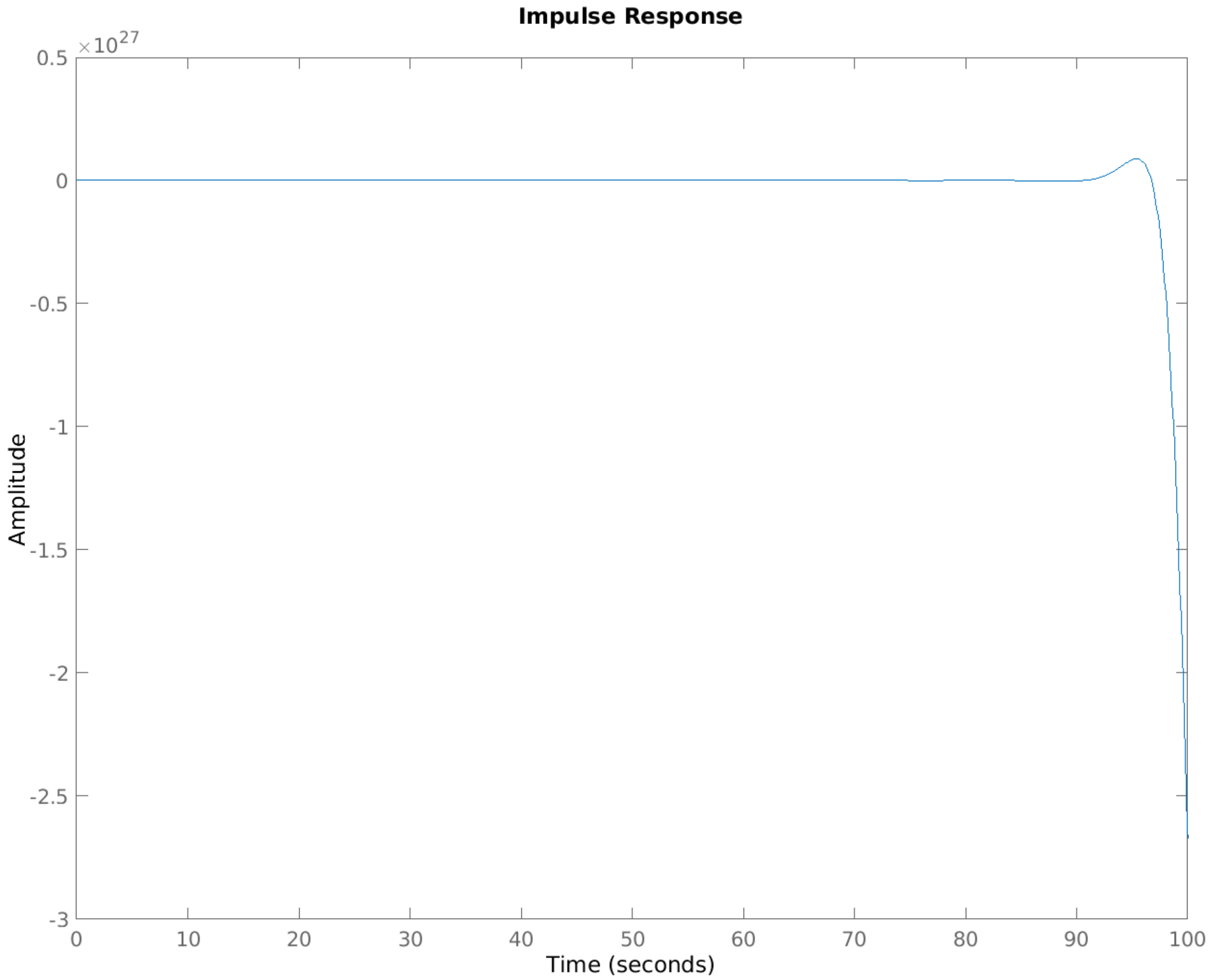
An IIR digital low pass filter is required to meet the following specifications:
Pass band attenuation≤ dB
Stop band attenuation≤30 dB
Passband edge=400 Hz
Stopband edge=800Hz
Sample rate=2 KHz

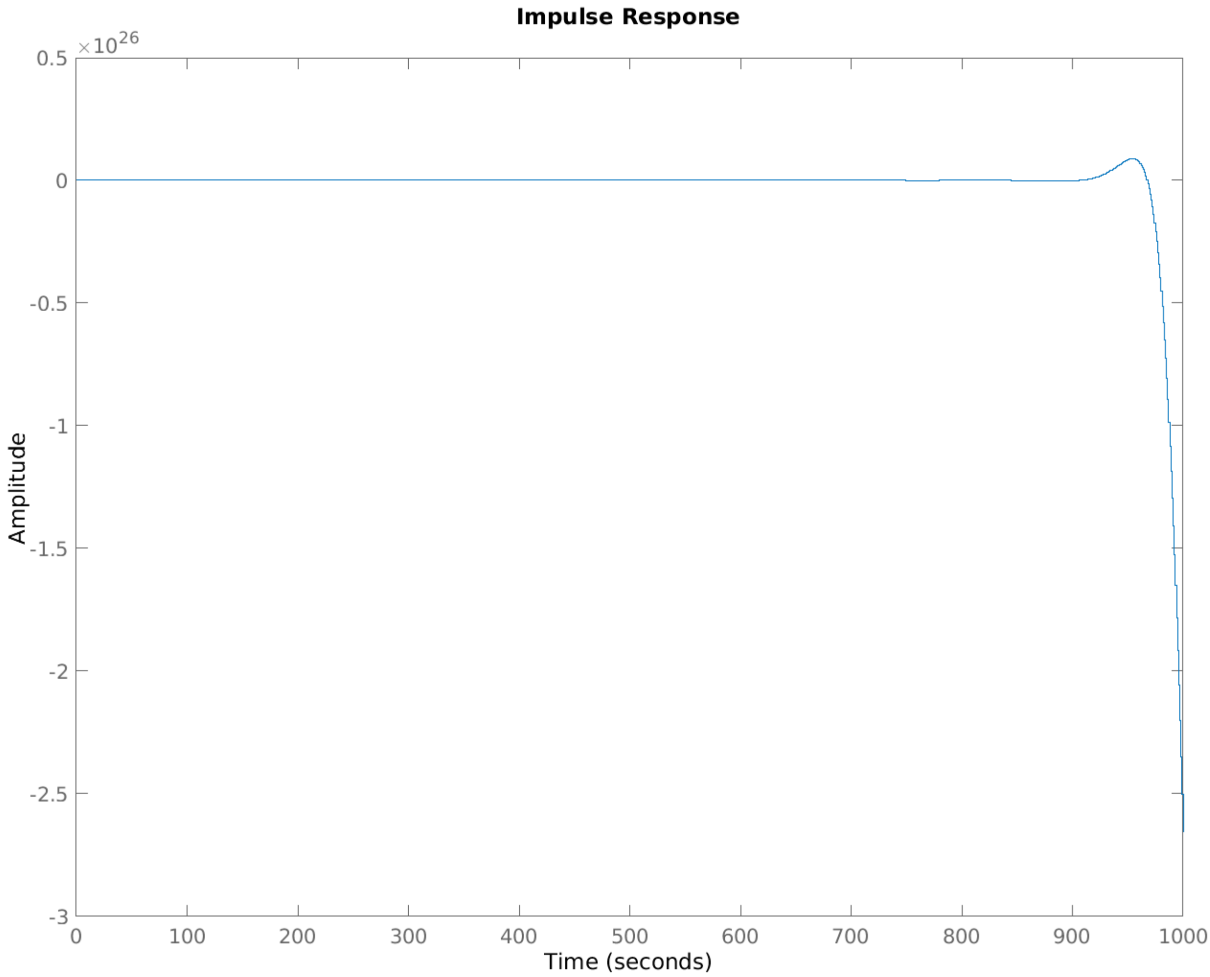
a. Find its order.
b. Plot its frequency response.
c. Plot its poles and zeros.
d. Plot its impulse response.
e. Convert this filter into digital IIR filter by means of impulse invariance method.
f. Plot the frequency response in z-domain.
g. Plot the impulse response in z-domain
h. Step response of the discrete time linear system

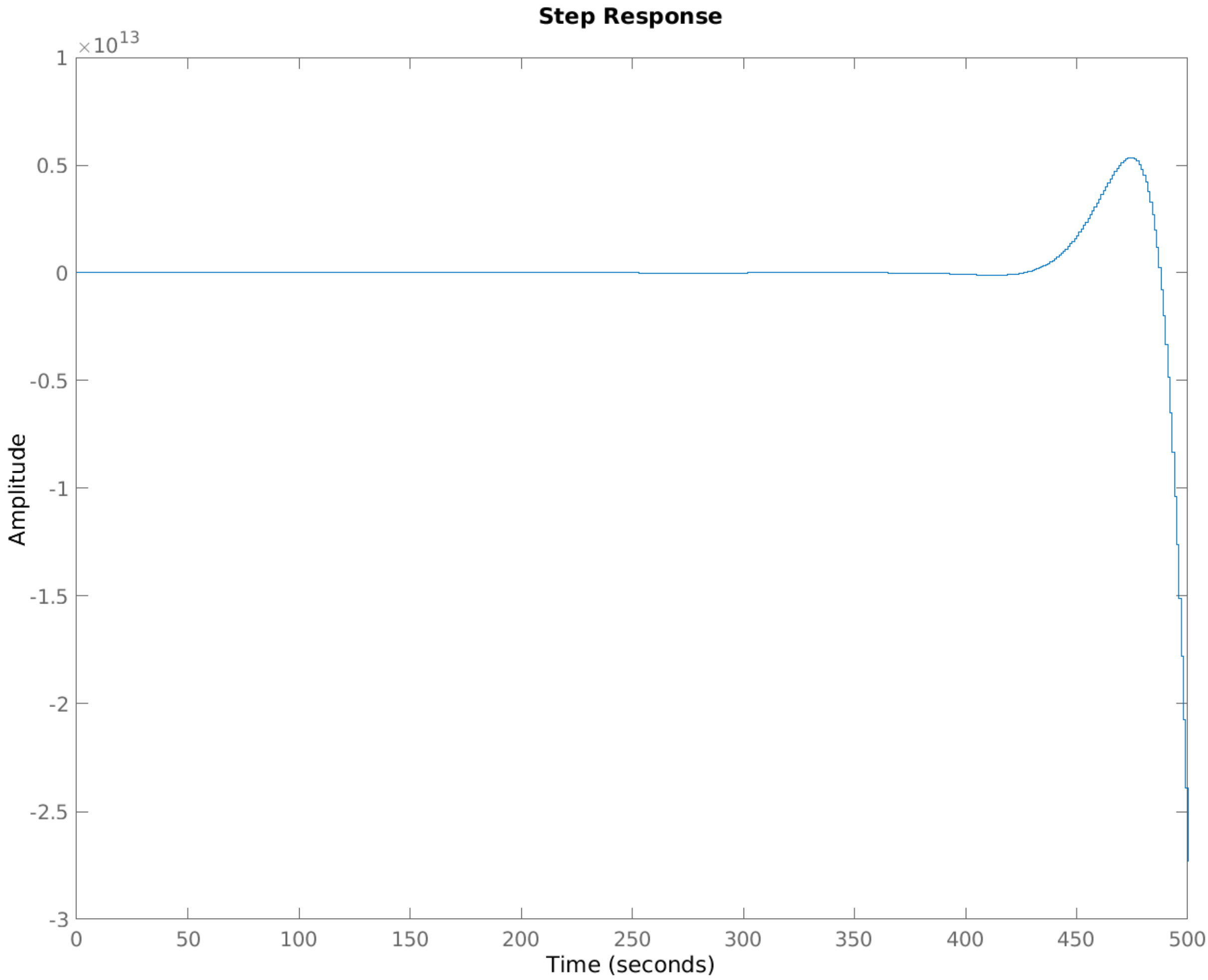
```
clear all;
close all;
wp= 400/2000;
ws= 800/2000;
rp= 4;
rs= 30;
[n,wn]= buttord(wp,ws,rp,rs);
[b,a]= butter(n,wn);
sys= tf(b,a);
figure;
freqs(b,a);
title('frequency response/ Sameep Dhakal/563');
figure;
pzmap(sys);
title('poles and zeros/ Sameep Dhakal/563');
figure;
impulse(b,a);
[bz,az]= impinvar(b,a,10);
figure;
dimpulse(bz,az);
figure;
dstep(bz,az);
```


frequency response/ Sameep Dhakal/563
Magnitude
0.02
0.015
0.01
0.005
Frequency (rad/s)
Phase (degrees)
200
100
0
-100
-200
10^{-2}
10^{-1}
10^{0}
10^{1}
Frequency (rad/s)


poles and zeros/ Sameep Dhakal/563
Imaginary Axis (seconds^{-1})
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-1
-0.5
0
0.5
1
Real Axis (seconds^{-1})


Impulse Response
×10^27
0.5
0
-0.5
-1
-1.5
-2
-2.5
-3
Amplitude
0
10
20
30
40
50
60
70
80
90
100
Time (seconds)


Impulse Response
×10^26
0.5
0
-0.5
-1
-1.5
-2
-2.5
-3
Amplitude
0
100
200
300
400
500
600
700
800
900
1000
Time (seconds)


Step Response
×10^13
1
0.5
0
-0.5
-1
-1.5
-2
-2.5
-3
Amplitude
0
50
100
150
200
250
300
350
400
450
500
Time (seconds)

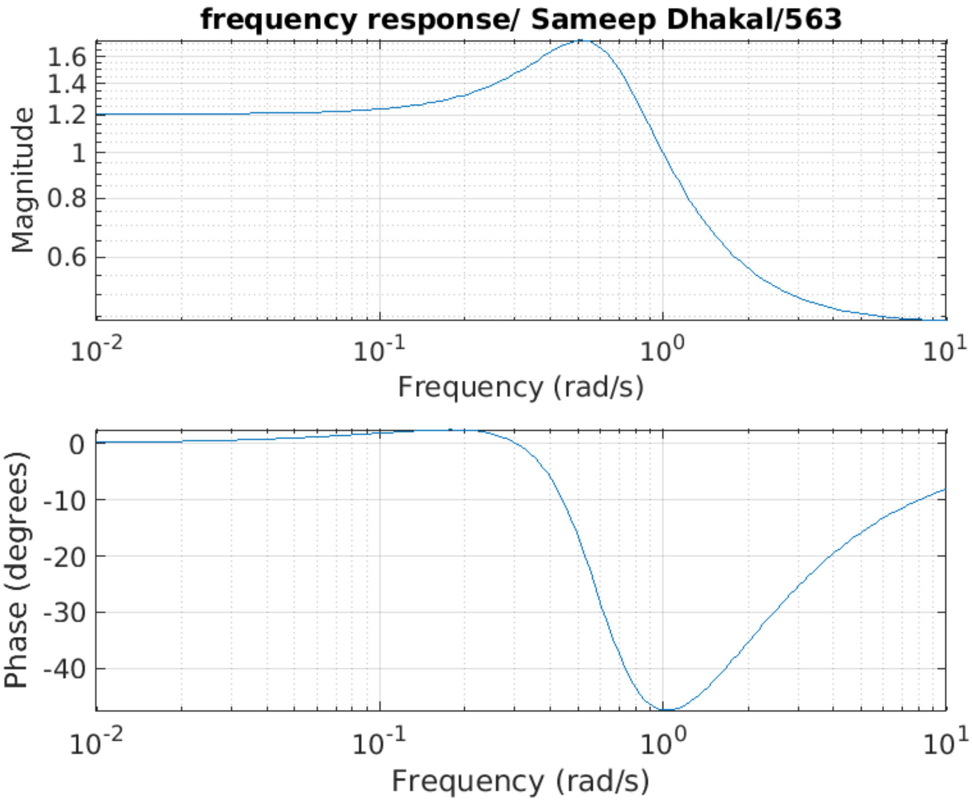
3. **Design of Chebyshev I and II Low Pass Filter**

   Passband attenuation≤1dB
   Stopband attenuation≤15dB
   Passband edge frequency=0.2π rad/sec
   Stopband edge frequency=0.3π rad/sec

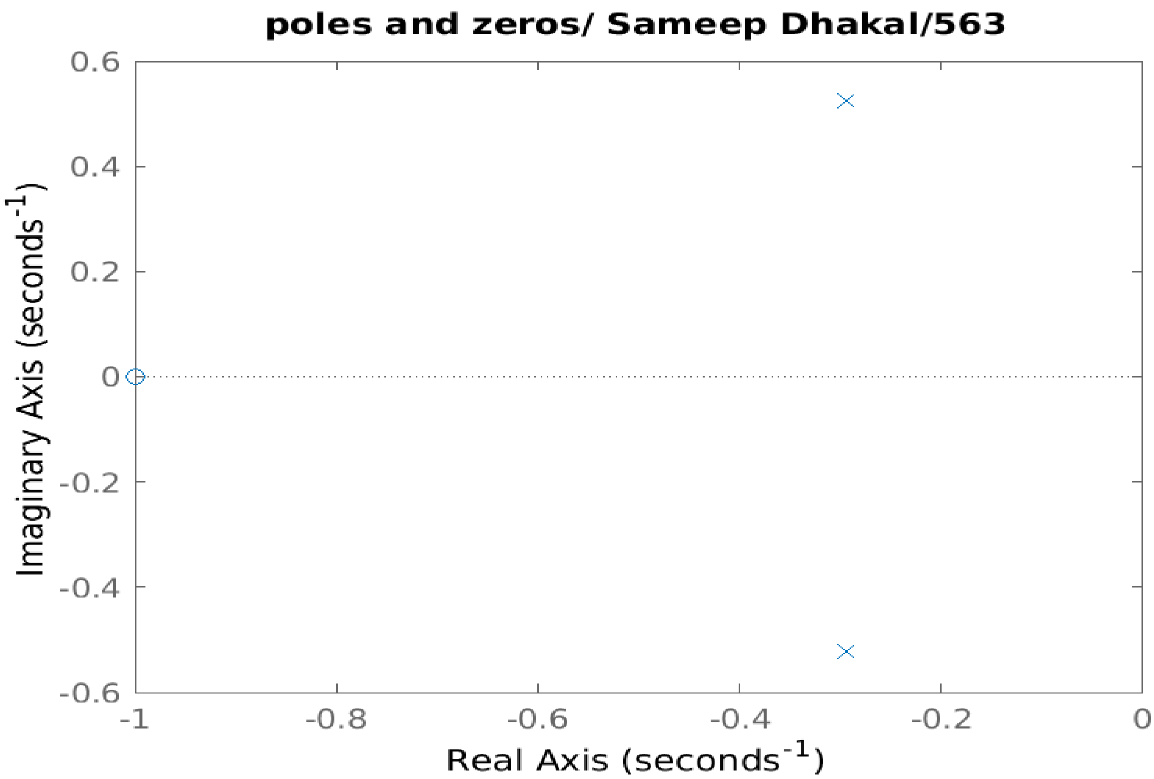
   a. Find its order.
   b. Plot its frequency response.
   c. Plot its poles and zeros.
   d. Plot its impulse response.
   e. Convert this filter into digital IIR filter by means of impulse invariance method.
   f. Plot the frequency response in z-domain.
   g. Plot the impulse response in z-domain
   h. Step response of the discrete time linear system

a. Chebyshev I

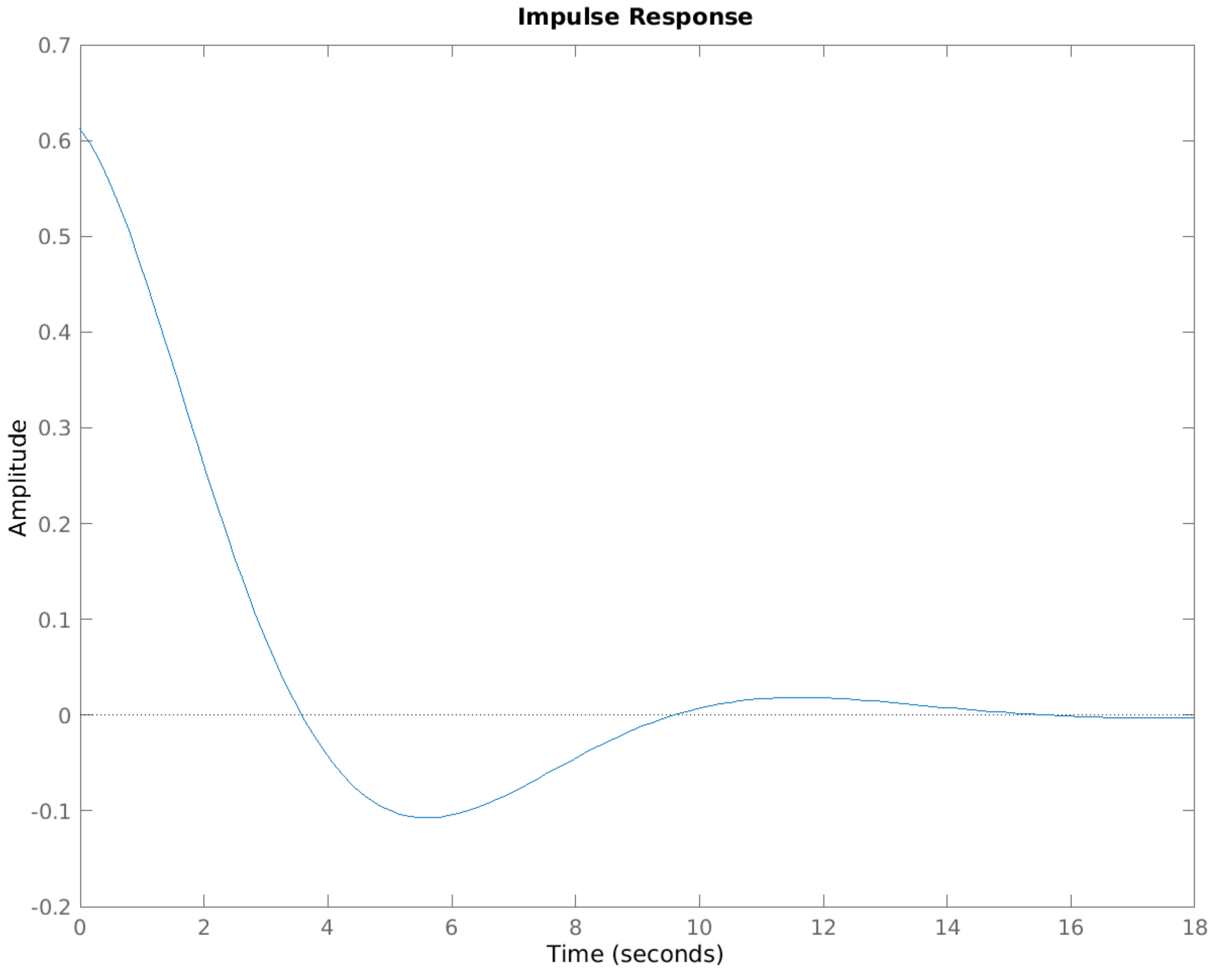
```
clc;
clear all;
close all;
wp= 0.2*pi;
ws=0.3*pi;
rp= 1;
rs= 15;
[n,wn]= cheb1ord(wp,ws,rp,rs);
[b,a]= cheby1(n,rp,wn);
sys= tf(b,a);
figure;
freqs(b,a);
title('frequency response/ Sameep Dhakal/563');
figure;
pzmap(sys);
title('poles and zeros/ Sameep Dhakal/563');
figure;
impulse(b,a);
[bz,az]= impinvar(b,a,10);
figure;
dimpulse(bz,az);
figure;
dstep(bz,az);
```

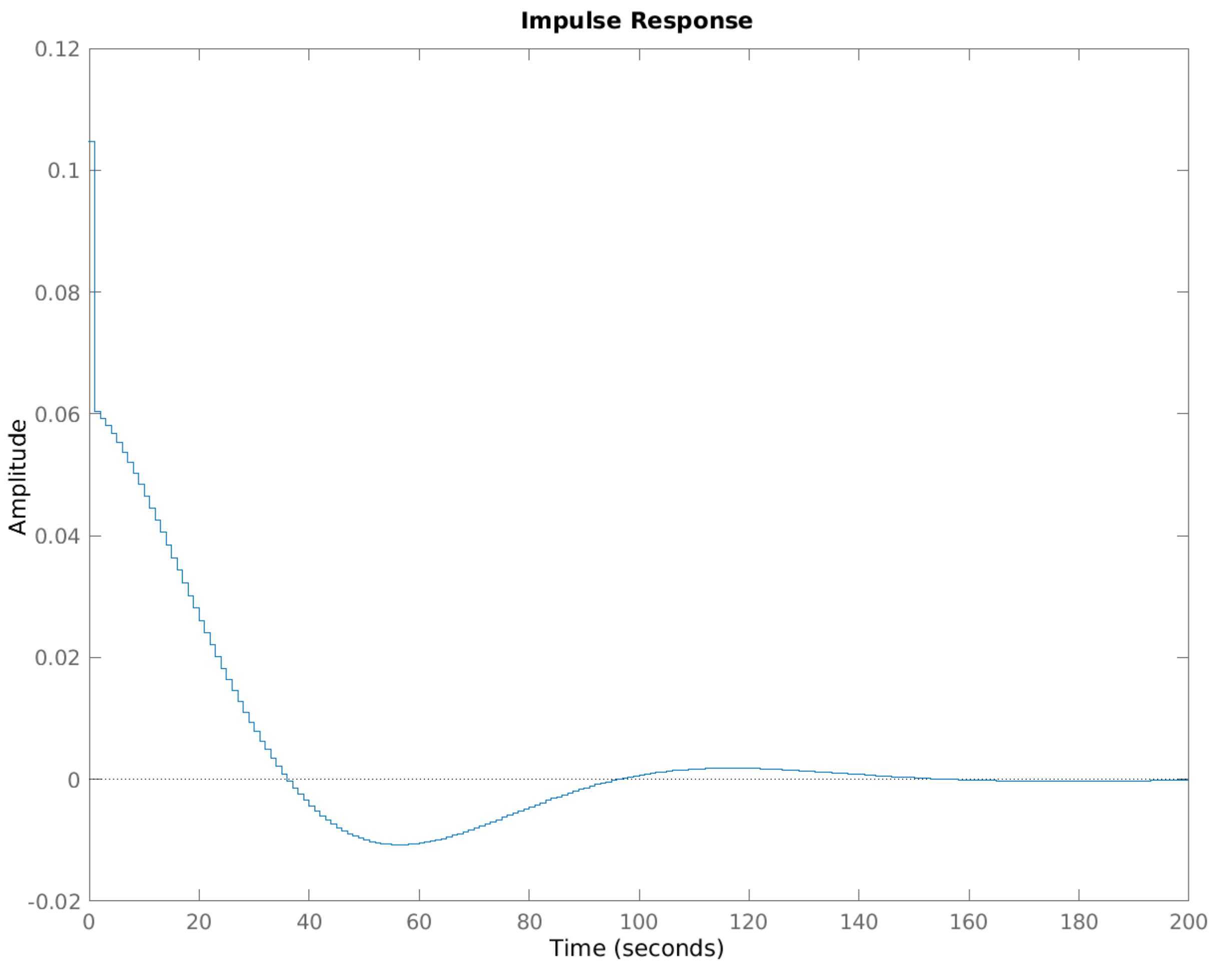


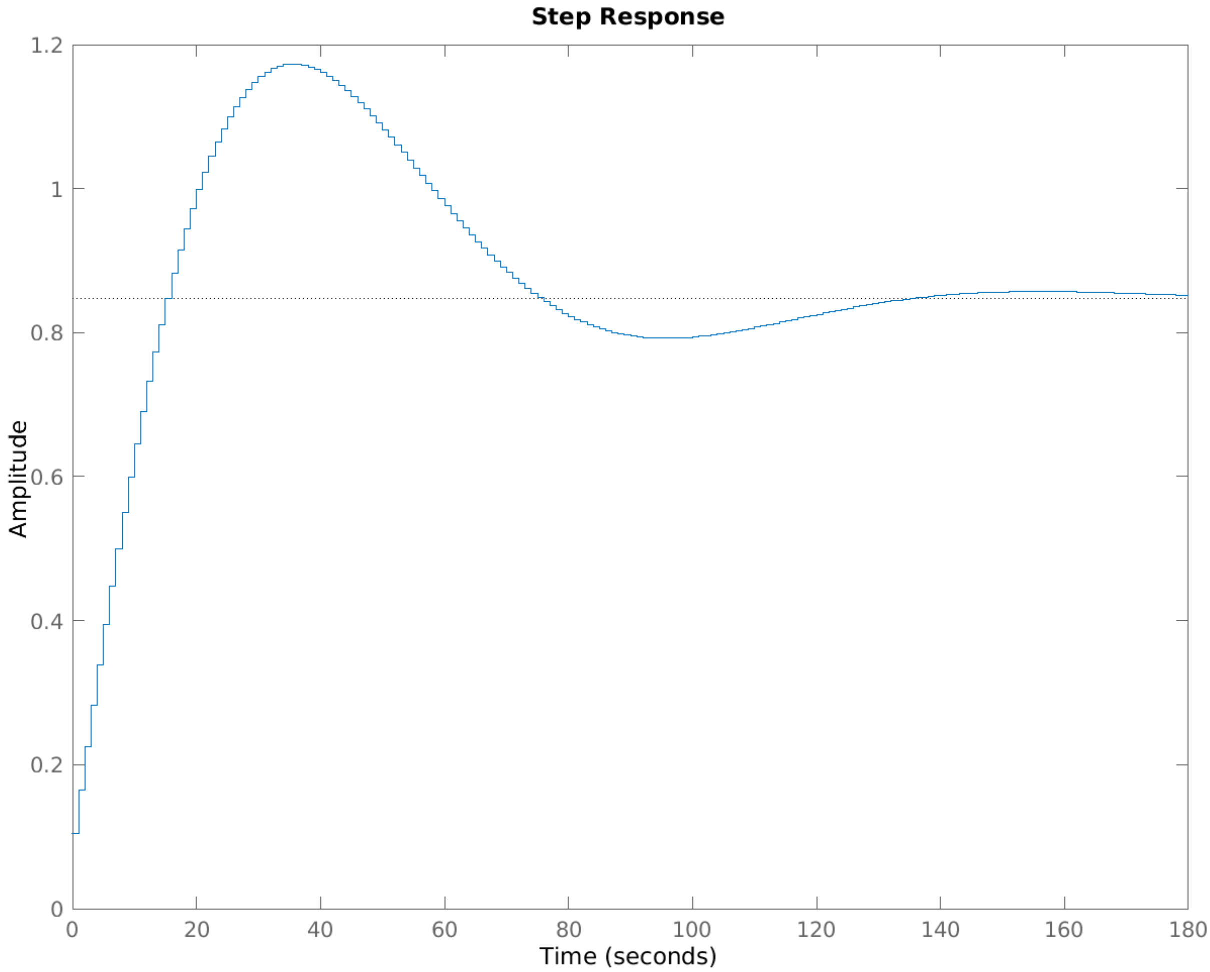
frequency response/ Sameep Dhakal/563
Magnitude
1.6
1.4
1.2
1
0.8
0.6
Frequency (rad/s)
Phase (degrees)
0
-10
-20
-30
-40
10^{-2}
10^{-1}
10^{0}
10^{1}



poles and zeros/ Sameep Dhakal/563
Imaginary Axis (seconds^{-1})
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-1
-0.8
-0.6
-0.4
-0.2
0
Real Axis (seconds^{-1})

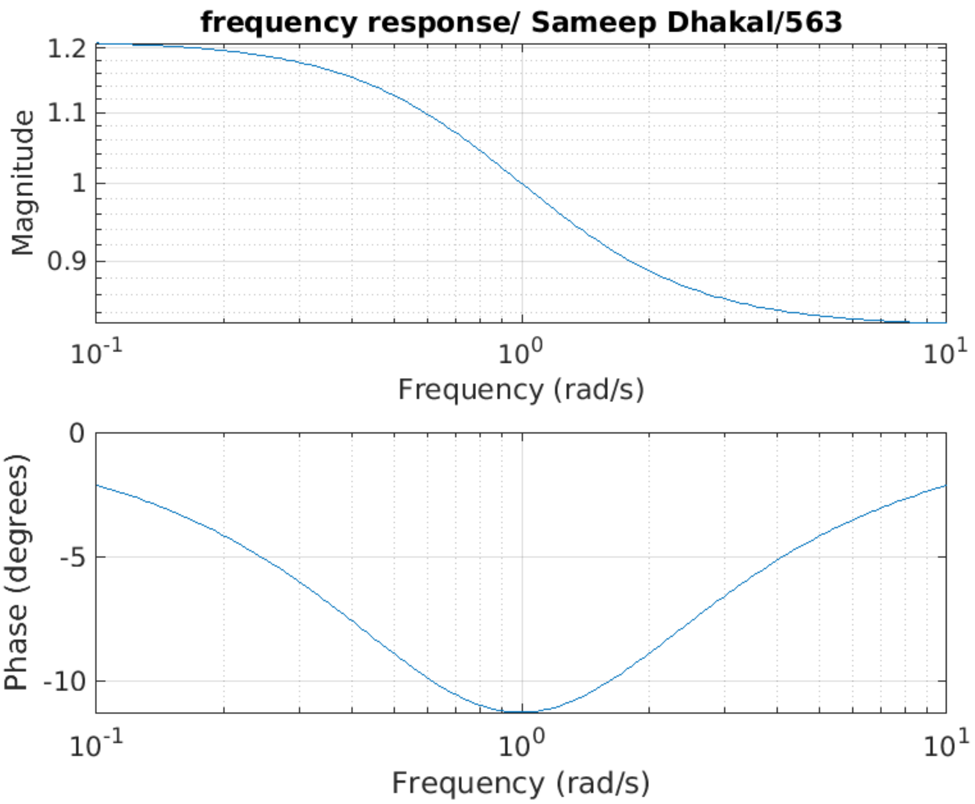

Impulse Response
Amplitude
Time (seconds)
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
-0.1
-0.2
0
2
4
6
8
10
12
14
16
18

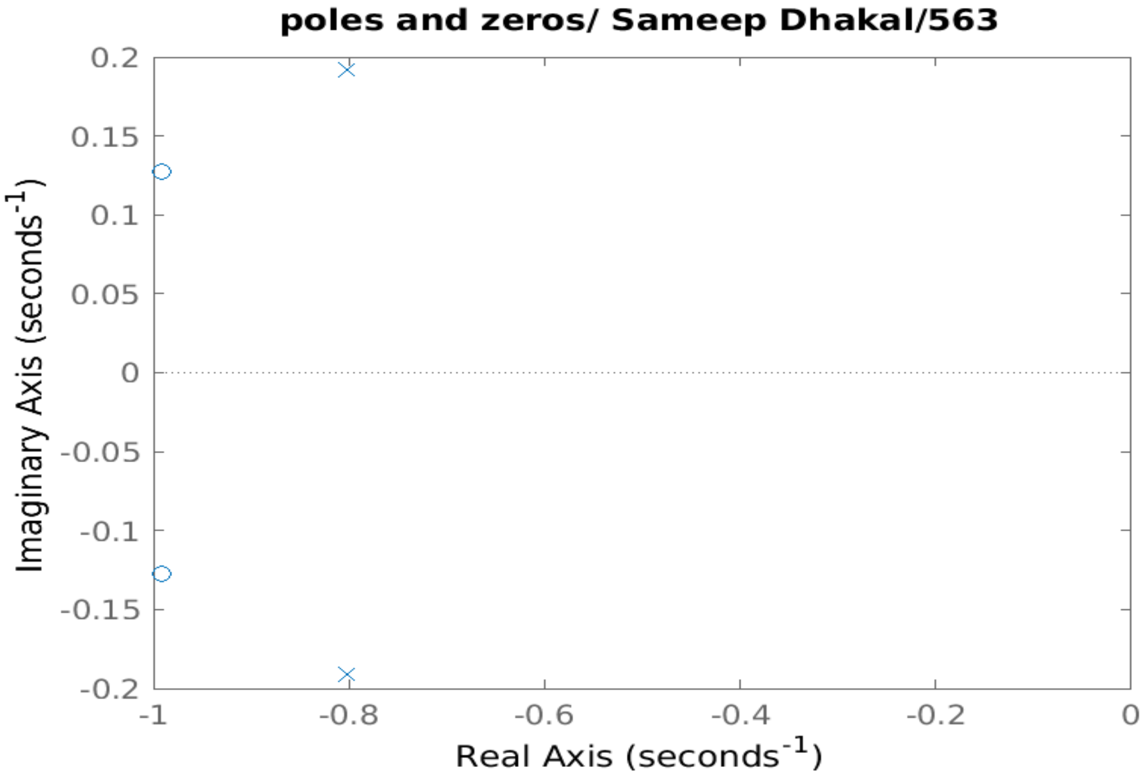

Impulse Response
Amplitude
Time (seconds)
0.12
0.1
0.08
0.06
0.04
0.02
0
-0.02
0
20
40
60
80
100
120
140
160
180
200

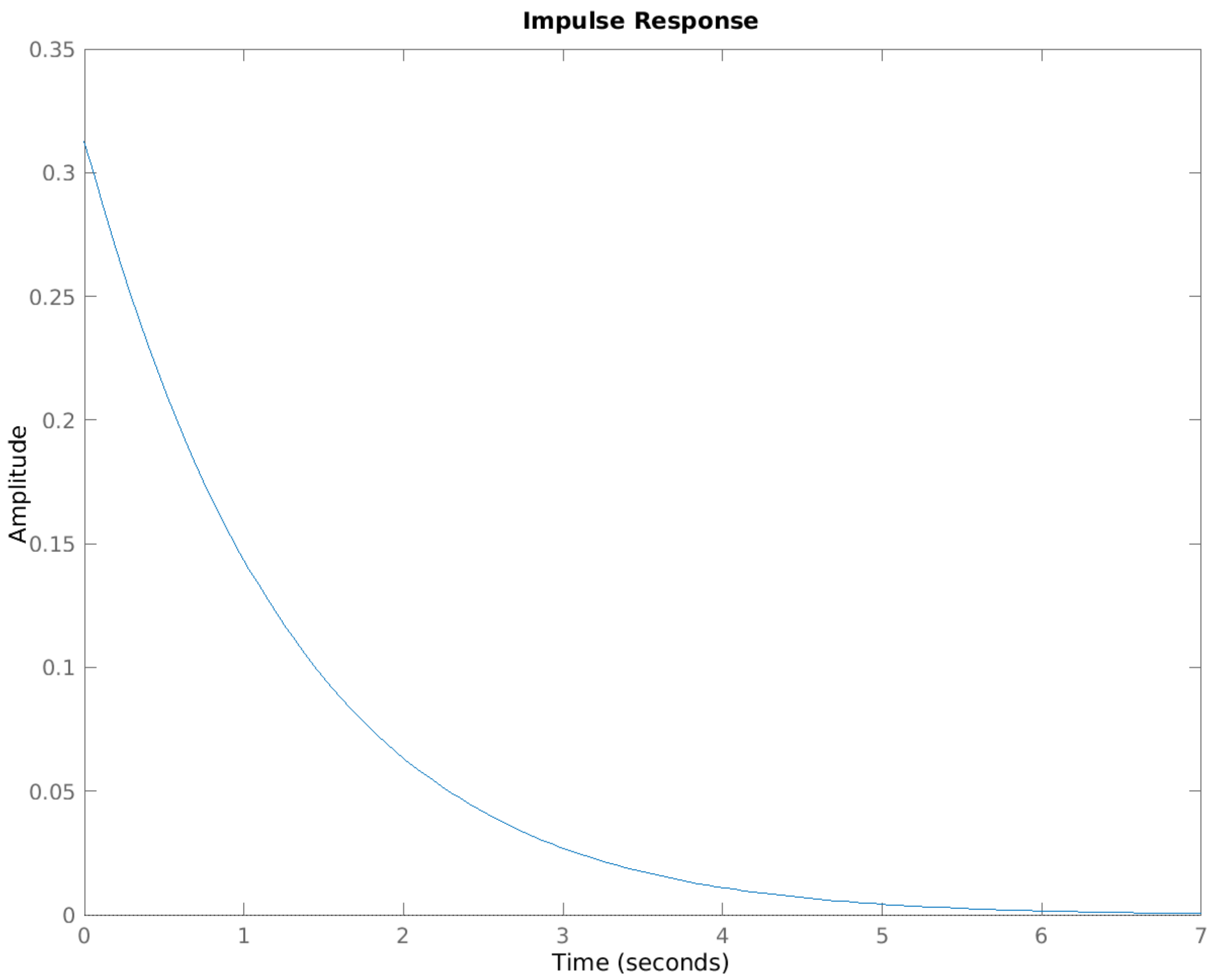

Step Response
Amplitude
Time (seconds)
0
0.2
0.4
0.6
0.8
1
1.2
0
20
40
60
80
100
120
140
160
180

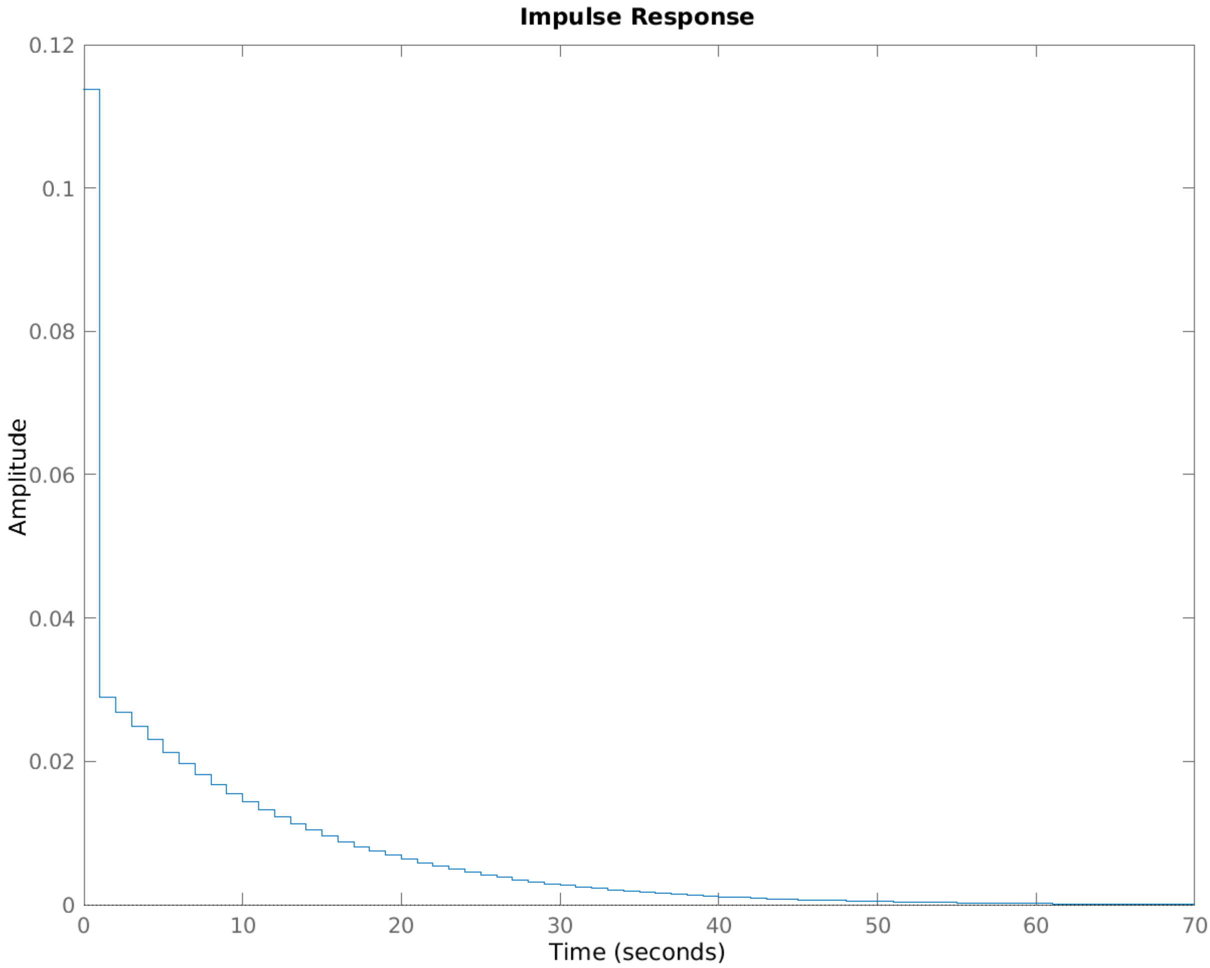
b. Chebyshev II

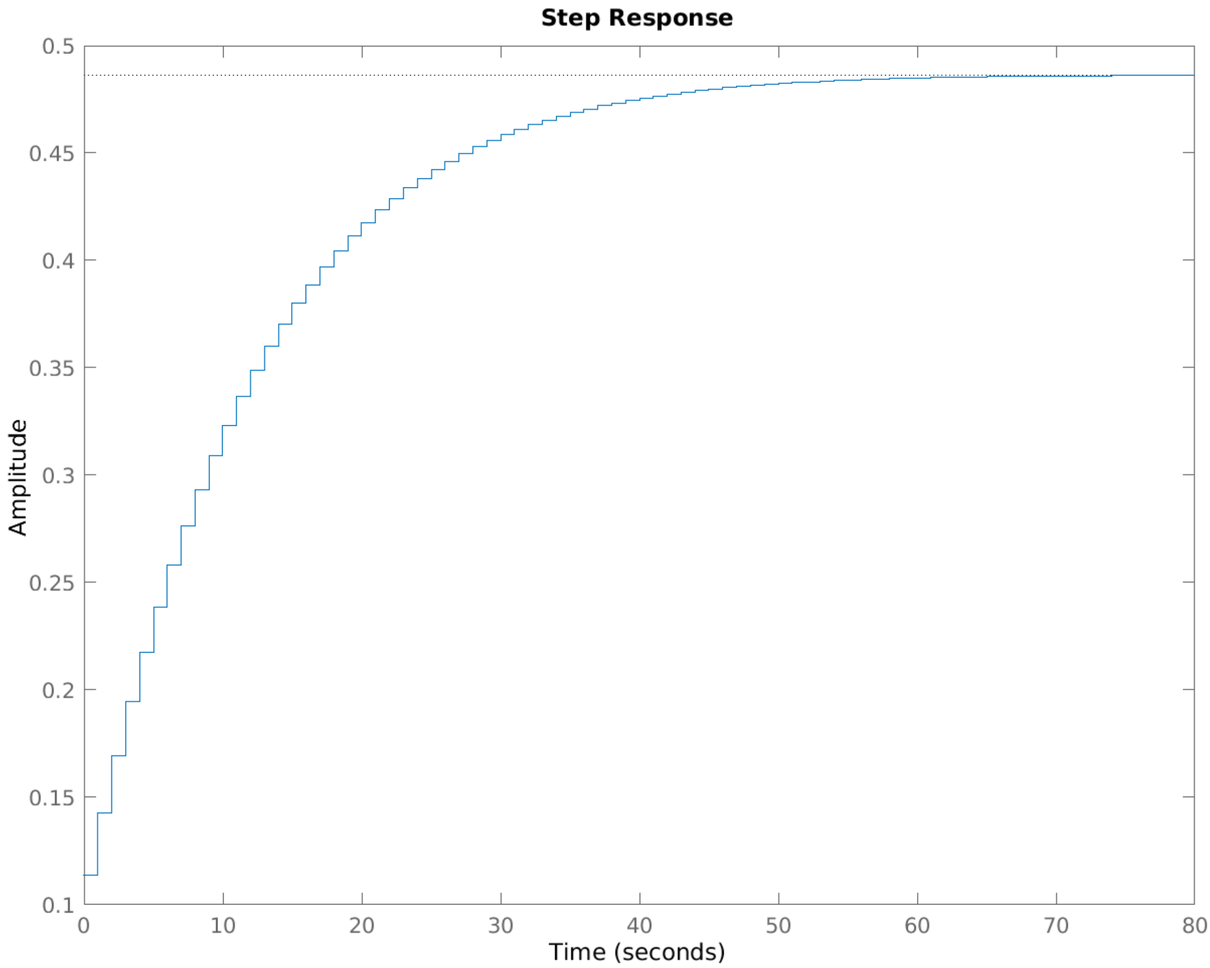
```
clc;
clear all;
close all;
wp= 0.2*pi;
ws=0.3*pi;
rp= 1;
rs= 15;
[n,wn]= cheb2ord(wp,ws,rp,rs);
[b,a]= cheby2(n,rs,wn);
sys= tf(b,a);
figure;
freqs(b,a);
title('frequency response/ Sameep Dhakal/563');
figure;
pzmap(sys);
title('poles and zeros/ Sameep Dhakal/563');
figure;
impulse(b,a);
[bz,az]= impinvar(b,a,10);
figure;
dimpulse(bz,az);
figure;
dstep(bz,az);
```




poles and zeros/ Sameep Dhakal/563
Imaginary Axis (seconds^{-1})
Real Axis (seconds^{-1})
0.2
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
-0.2
-1
-0.8
-0.6
-0.4
-0.2
0


Impulse Response
Amplitude
Time (seconds)
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
0
1
2
3
4
5
6
7


Impulse Response
Amplitude
0.12
0.1
0.08
0.06
0.04
0.02
0
0
10
20
30
40
50
60
70
Time (seconds)

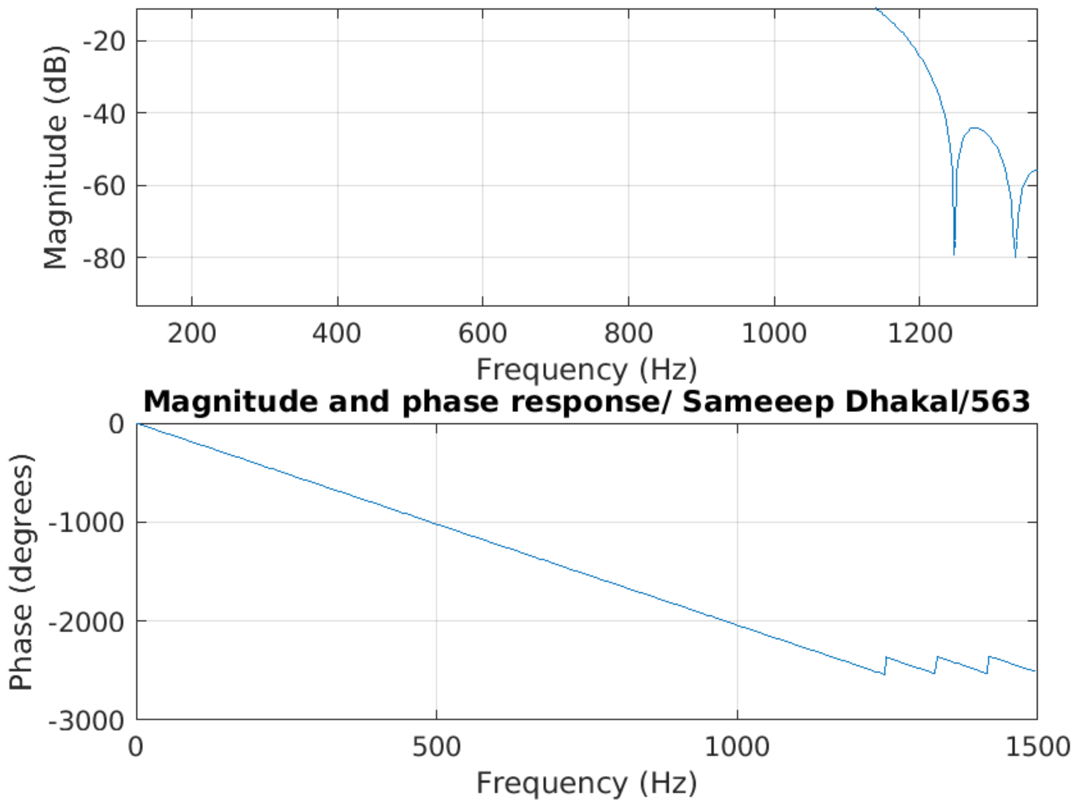

Step Response
Amplitude
0.5
0.45
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0
10
20
30
40
50
60
70
80
Time (seconds)

4. Finite Impulse Response

```
clc;
clear all;
close all;
fp=input('enter passband freq:');
fs=input('enter stopband freq:');
rs=input('enter passband attenuation:');
rp=input('enter stopband attenuation:');
f= input('Enter sampling freq:');
num= -20*log10(sqrt(rp*rs))-13;
dem= 14.6*(fp-fs)/f;
n= ceil(num/dem);
n=abs(n);
wp= 2*fp/f;
ws= 2*fs/f;
wn= (ws+wp)/2;
if(rem(n,2)==0)
    m= n+1;
else
    m=n;
    n=n-1;
end
w=hann(m);
b=fir1(n,wn,'low',w);
freqz(b,1,500,3000);
title('Magnitude and phase response/ Sameeep Dhakal/563');
```

Input:

```
enter passband freq:
1000
enter stopband freq:
1200
enter passband attenuation:
2
enter stopband attenuation:
45
Enter sampling freq:
3000
```



DISCUSSION AND CONCLUSION:

In this lab we worked on designing IRR filters and also low pass filters analyzed their output using matlab.