DSAP LAB5 075bct092

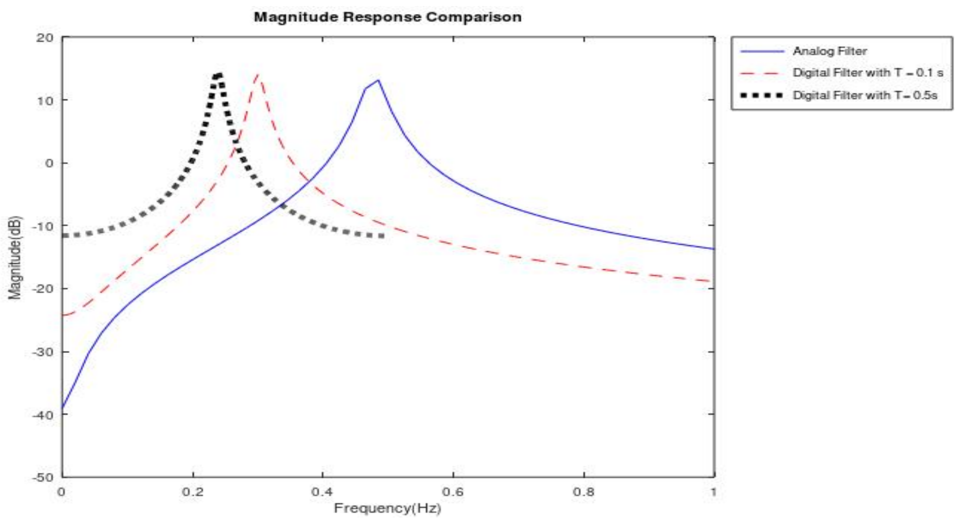
A.

1. Convert the analog filter $H_a(s) = \frac{s+0.1}{(s+0.1)^2+9}$ in to a digital IIR filter by means of the impulse invariance method. Plot the frequency response (magnitude) of the designed filter taking sampling interval (T) of 0.1, 0.5 seconds. Compare the response of the filter designed to that of the analog one. Comment on the effect of T on the response.

Code:

```
pkg load signal
b = [1 0.1];
a = [1 0.2 9.01];
Ts = 0.1;
fs = 1/Ts ;
Wa = linspace (0,8,64) ;
Ha = freqs (b,a,Wa);
% Converting the analog filter to discrete one
[bz,az] = impinvar (b,a,fs);
[ Hz , Wz ] = freqz (bz,az,512);
Ts1 = 0.5;
fs1 = 1/Ts1;
[ bz1, az1 ] = impinvar (b,a,fs1) ;
[ Hz1, Wz1 ] = freqz (bz1, az1, 512) ;
% Frequency Response Comparison
plot (Wa/(2* pi) ,20*log10(abs(Ha)),'b-',Wz ,20*log10(abs(Hz)),'r--',
        Wz1/(2* pi) ,20*log10(abs(Hz1)),'k:','LineWidth',2);
axis ([0 1 -50 20]) ;
xlabel ('Frequency(Hz)') , ylabel ('Magnitude(dB)') ;
title ('Magnitude Response Comparison') ;
legend('Analog Filter','Digital Filter with T = 0.1 s','Digital Filter with T=
        0.5s','Location','NorthEastOutside');
```

Output:

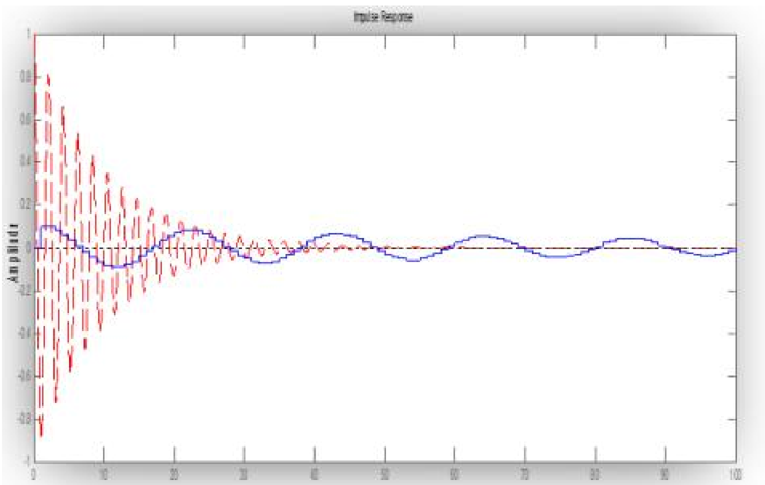


2. Compare the unit sample response of the designed digital IIR filter with the impulse response of analog filter for T=0.1 and 0.5.

Code:

```
pkg load signal
b =[1 0.1];
a =[1 0.2 9.01];
Ts = 0.1;
fs = 1/ Ts ;
[bz,az] = impinvar(b,a,fs) ;
Ts1 = 0.5;
fs1 = 1/ Ts1 ;
[bz1,az1] = impinvar(b,a,fs1) ;
impulse(b,a); hold on ;
dimpulse(bz,az); hold on;
dimpulse(bz1,az1) ;
legend ('Analog Filter','Digital IIR Filter with T = 0.1 s','Digital IIR Filter with T = 0.5 s');
```

Output:

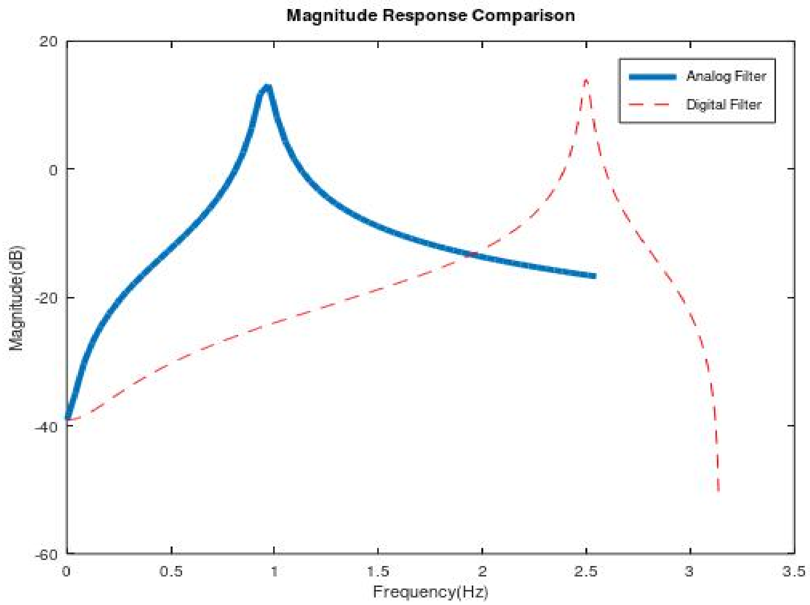


3. Convert the above analog filter in to a digital IIR filter by means of bilinear transformation and repeat all the procedures as specified in 1.1.

Code:

```
b = [1 0.1]; a = [1 0.2 9.01];
Ts = 0.5; fs = 1/ Ts ;
Wa = linspace (0 , 8 , 64) ;
Ha = freqs (b ,a , Wa);
plot ( Wa /pi ,20*log10(abs(Ha)),'LineWidth',2); hold on;
[bz,az]=bilinear(b,a,fs);
[Hz,Wz]=freqz(bz,az,512);
plot(Wz,20*log10(abs(Hz)) ,'r--') ;
xlabel('Frequency(Hz)'), ylabel('Magnitude(dB)');
title('Magnitude Response Comparison');
legend('Analog Filter','Digital Filter');
```

Output:



B.

An IIR digital low pass filter is required to meet the following specifications:

Pass band ripple (or peak to peak ripple): ≤ 0.5 dB

Passband edge: 1.2 kHz

Stopband attenuation:≥ 40 dB

Stopband edge: 2.0 kHz

Sample rate: 8.0 kHz

Use the MATLAB Signal Processing Toolbox functions to determine

- The required filter order,
- The cutoff frequency,
- The numerator and the denominator coefficients

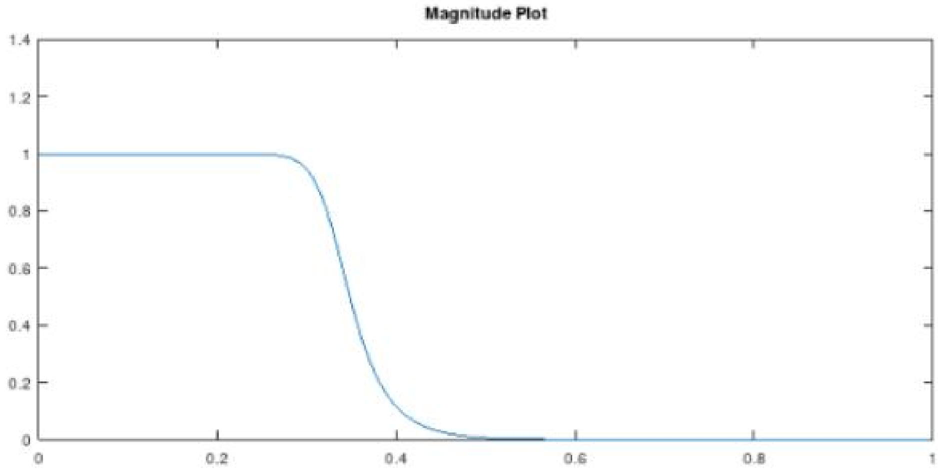
for the digital Butterwoth, digital Chebyshev and digital Elliptic filters. Also plot their frequency responses. Describe the nature of each response.

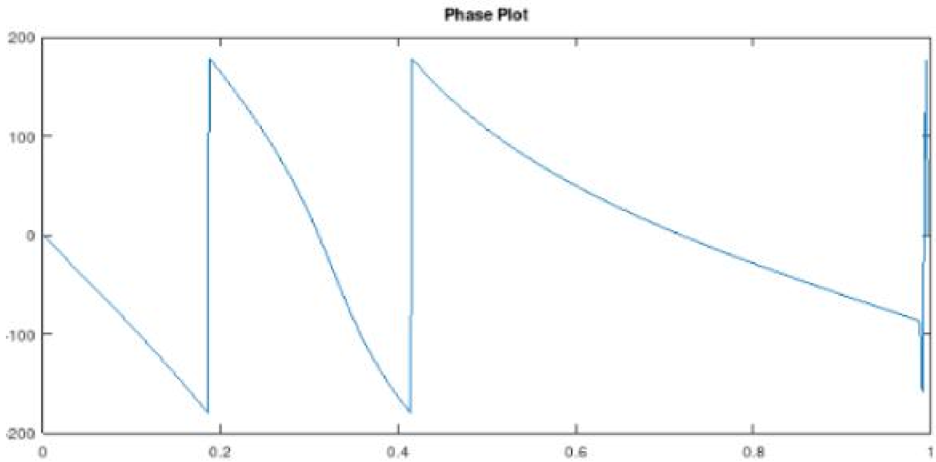
Code:

```
function [N, num1, den1]= b(fil_choice)
fs=8000; pb=1200;
sb=2000;Rp=0.5;
Rs=40;
fn=fs/2;
wp=pb/fn;
ws=sb/fn;
switch (fil_choice)
  case 'butter'
    [N,wc]=buttord(wp,ws,Rp,Rs);
     [num1,den1]=butter(N,wc);
  case 'cheby'
    [N,wc]=cheb1ord(wp,ws,Rp,Rs);
    [num1,den1]=cheby1(N,Rp,wc);
  case 'ellip'
    [N,wc]=ellipord(wp,ws,Rp,Rs);
    [num1,den1]=ellip(N,Rp, Rs, wc);
end
[Hd,wd]=freqz(num1,den1);
magd=abs(Hd);
phase=angle(Hd)*180/pi;
subplot(2,1,1);
plot(wd/pi,magd);title('Magnitude Plot')
subplot(2,1,2);
plot(wd/pi,phase);title('Phase Plot')
end
```

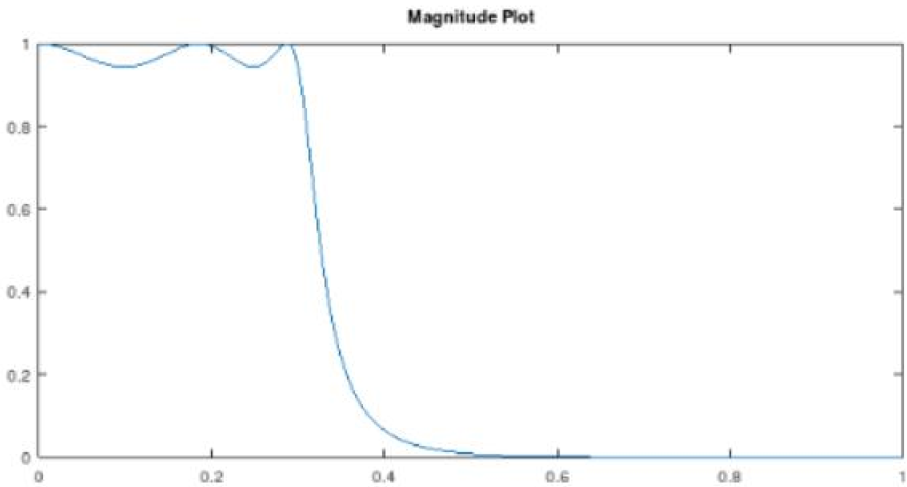
Output:

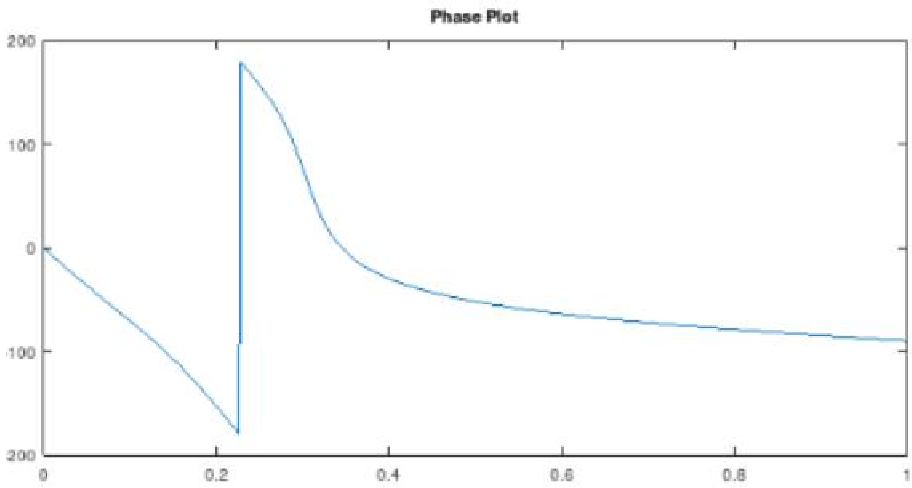
a. Butterworth



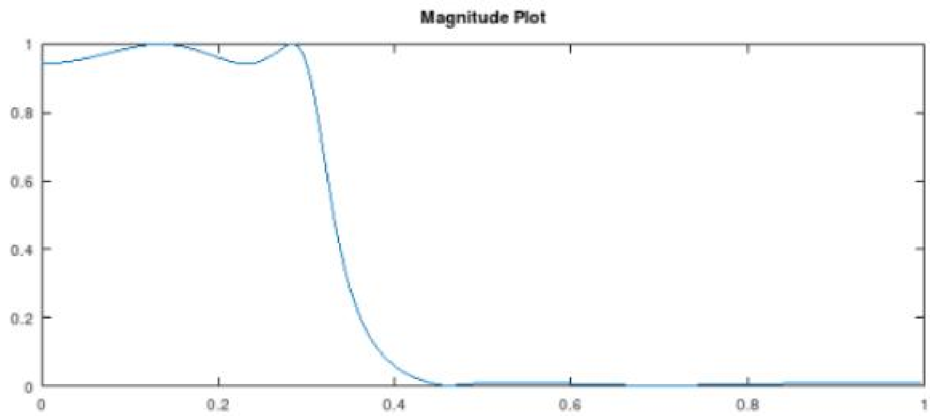


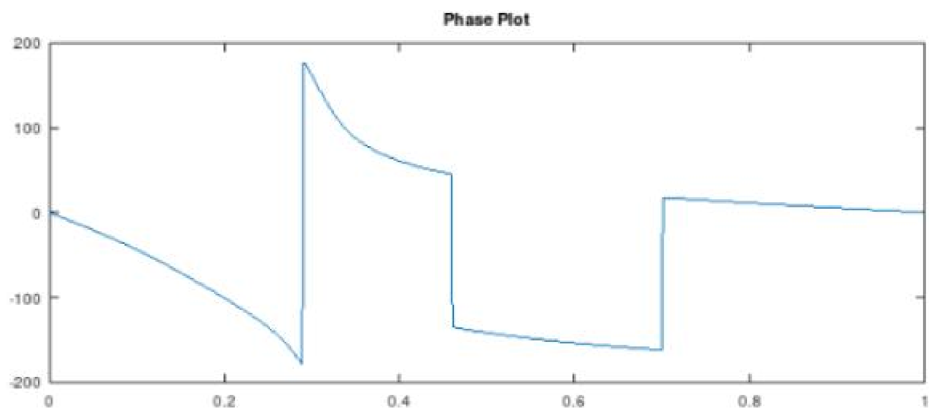
b. Chebyshev




Phase Plot
200
100
0
-100
-200
0
0.2
0.4
0.6
0.8
1

C. Elliptic


Magnitude Plot
1
0.8
0.6
0.4
0.2
0
0
0.2
0.4
0.6
0.8
1


Phase Plot
200
100
0
-100
-200
0
0.2
0.4
0.6
0.8
1