ME (Embedded Systems)

DIGITAL SIGNAL PROCESSING

Lab Assignment 2

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1. Write a Matlab program to Design an ideal linear phase bandpass FIR filter with cutoff frequencies pi/6 rads and pi/3 rads, using frequency sampling technique. Assume 25 tap coefficients.

Code :

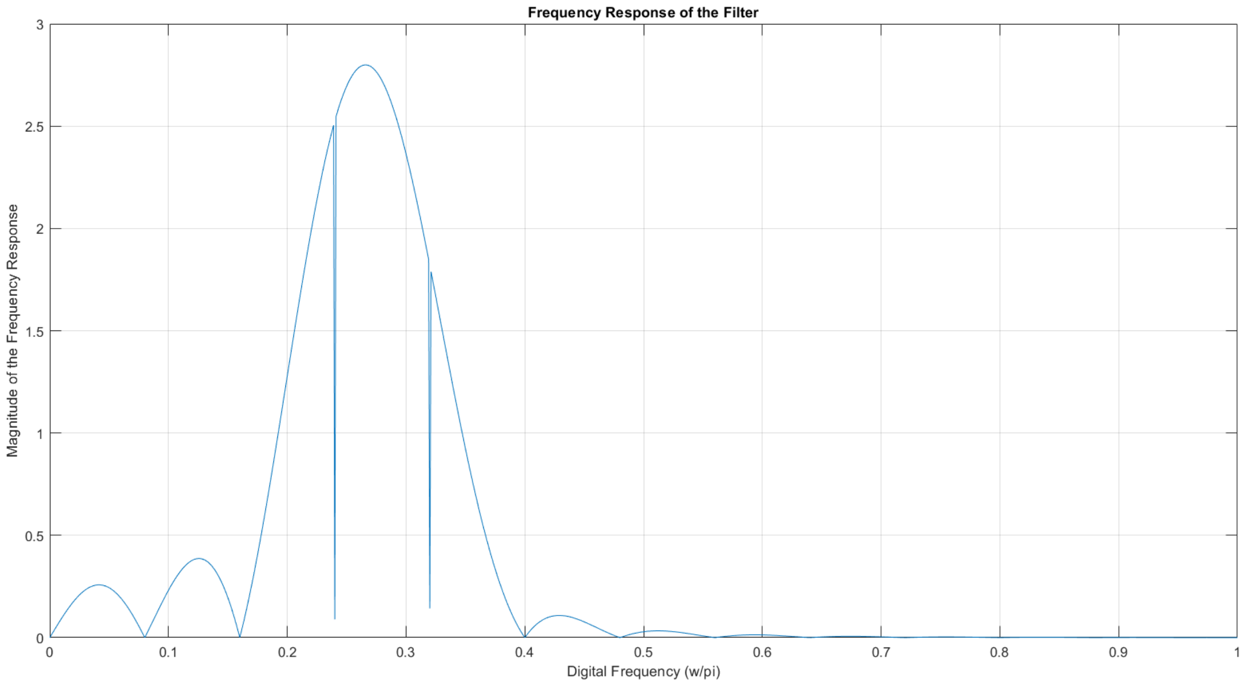
```
clc;
clear all;

wc1=pi/6;
wc2=pi/3;
N=25;

%linear phase factor
alpha=(N-1)/2;

%dft samples indexed with k
hk=[zeros(1,N)];
k=0:1:N-1;

for i=1:N
    w=2*pi*k(i)/N;
    if(w>=wc1 && w<=wc2)
        hk(i)=exp(-1i*w*alpha);
    end
end

%magnitude and phase of DFT samples
Hmag=abs(hk);
Hphase=phase(hk);

%comb filter
```

```
num=[1,zeros(1,N-1),-1];
den=N;
Hc=tf(num,den,0.01,'Variable','z^-1'); %assuming
sampling time for input as 0.01s
disp("Comb filter transfer function:");
Hc

%Resonator
%Since N=25 there is no N/2 term
% H(0)
num=[Hmag(1)];
den=[1,-1];
Hr=tf(num,den,0.01,'Variable','z^-1');

% find Hk(z) for values of k=1 to N-1/2 and add to Hr
for i=1:(N-1)/2
    if(Hmag(i+1)~=0)
        num=[cos(Hphase(i+1)),cos(Hphase(i+1)-
2*pi*i/N)];
        den=[1,-2*cos(2*pi*i/N),1];
        Hkz=tf(num,den,0.01,'Variable','z^-1');
        Hr=Hr+2*Hmag(i+1)*Hkz;
    end
end
disp("Resonator Tranfer function");
Hr

%Multiply tranfer function of comb filter and
resonator
Hz=Hc*Hr;
disp("Final transfer function:");
Hz

[num,den]=tfdata(Hz,'v');

w=0:.001*pi:pi;
Hw=freqz(num,den,w);
Domega=w/pi;
plot(Domega,abs(Hw));
title('Frequency Response of the Filter')
xlabel('Digital Frequency (w/pi)')
ylabel('Magnitude of the Frequency Response')
grid on
```

Output:

Command Window

```
Comb filter transfer function:

Hc =

  1 - z^-25
  ---------
     25

Sample time: 0.01 seconds
Discrete-time transfer function.

Resonator Tranfer function

Hr =

  -0.1069 - 0.6693 z^-1 - 0.6693 z^-2 - 0.1069 z^-3
  -------------------------------------------------
    1 - 2.53 z^-1 + 3.562 z^-2 - 2.53 z^-3 + z^-4

Sample time: 0.01 seconds
Discrete-time transfer function.

Final transfer function:

Hz =

  -0.1069 - 0.6693 z^-1 - 0.6693 z^-2 - 0.1069 z^-3 + 0.1069 z^-25 + 0.6693 z^-26 + 0.6693 z^-27 + 0.1069 z^-28
  ------------------------------------------------------------------------------------------------------------
                               25 - 63.24 z^-1 + 89.06 z^-2 - 63.24 z^-3 + 25 z^-4

Sample time: 0.01 seconds
Discrete-time transfer function.
```



2. Write a Matlab program to Design a digital Butterworth filter using Impulse Invariance transformation to meet the following specification.

(Do not use built-in functions to design the analog filter)

$$0 \geq |H(e^{j\Omega})|_{dB} \geq -1; \qquad \text{for} \qquad 0 \leq \Omega \leq 20 \text{ rad/sec.}$$

$$|H(e^{j\Omega})|_{dB} < -60; \qquad \text{for} \qquad \Omega \geq 200 \text{ rad/sec.}$$

Assume a sampling period be = 0.01 sec.

<u>**Code**</u>**:**

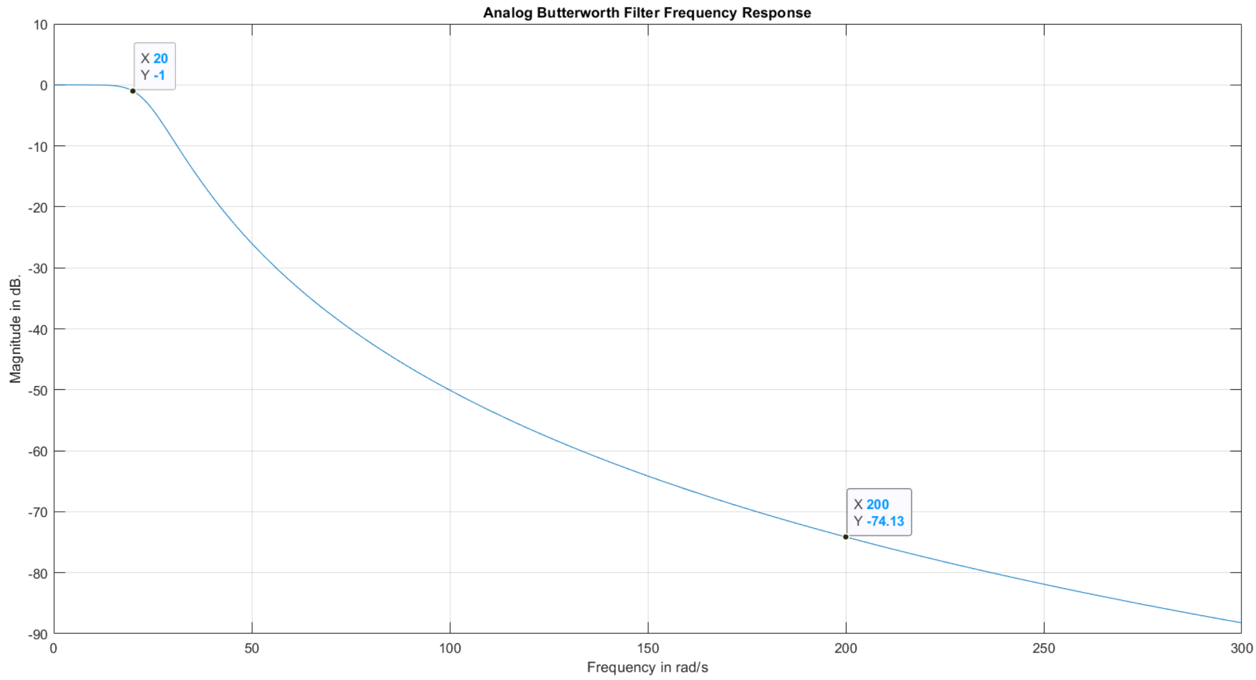
```
clc;
clear all;
ap=-1;
as=-60;
wp=20;
ws=200;
T=0.01;
Fs=1/T;

%calculate N
N=ceil(log10((10^(-ap/10)-1)/(10^(-as/10)-
1))/(2*log10(wp/ws)));
disp("N:");
disp(N);
%calculate cutoff frequency
wc=wp/((10^(-ap/10)-1)^(1/(2*N)));
disp("Cutoff Freq:");
disp(wc);

% transfer function by finding poles
sk=[]
hs=1;
for k=0:N-1
 sk(k+1)=wc*exp(-1i*(2*k+1+N)*pi/(2*N));
 tf(abs(sk(k+1)),[1,-sk(k+1)])
 hs=hs*tf(abs(sk(k+1)),[1,-sk(k+1)]);
end
disp("poles:");
disp(sk);
hs

%frequency response of analog filter
```

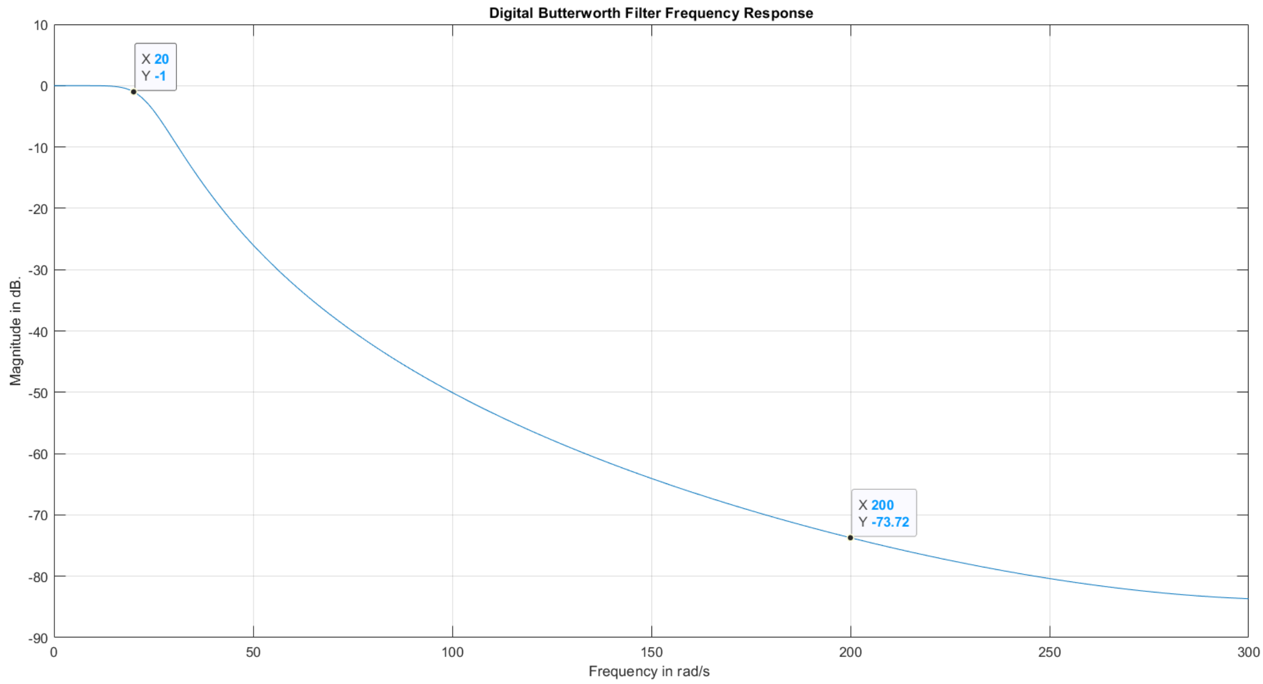
```
[num,den]=tfdata(hs,'v');
fre1=0:1:300;
[resps]=freqs(num,den,fre1);
mags=20*log10(abs(resps));
plot(fre1,mags);
title('Analog Butterworth Filter Frequency Response')
xlabel('Frequency in rad/s');
ylabel('Magnitude in dB.');
grid;
zoom;

%frequency response of digital filter
figure;
[BZ,AZ]=impinvar(num,den,Fs);
fre2=0:0.01:3;
[respz]=freqz(BZ,AZ,fre2);
magz=20*log10(abs(respz));
f2=fre2*Fs;
plot(f2,magz);
title('Digital Butterworth Filter Frequency Response')
xlabel('Frequency in rad/s ');
ylabel('Magnitude in dB.');
grid on;
```

Output:

```
Command Window
N:
     4

Cutoff Freq:
   23.6801
```

```
poles:
  -9.0620 -21.8775i -21.8775 - 9.0620i -21.8775 + 9.0620i  -9.0620 +21.8775i


hs =

                                               3.144e05
  ----------------------------------------------------------------------------------------------------
  s^4 + (61.88+1.066e-14i) s^3 + (1915+7.958e-13i) s^2 + (3.47e04+2.183e-11i) s + (3.144e05+2.91e-10i)

Continuous-time transfer function.
```


Analog Butterworth Filter Frequency Response
X 20
Y -1
X 200
Y -74.13
Magnitude in dB.
Frequency in rad/s
10
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
0
50
100
150
200
250
300


Digital Butterworth Filter Frequency Response
X 20
Y -1
X 200
Y -73.72
Magnitude in dB.
Frequency in rad/s
10
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
0
50
100
150
200
250
300