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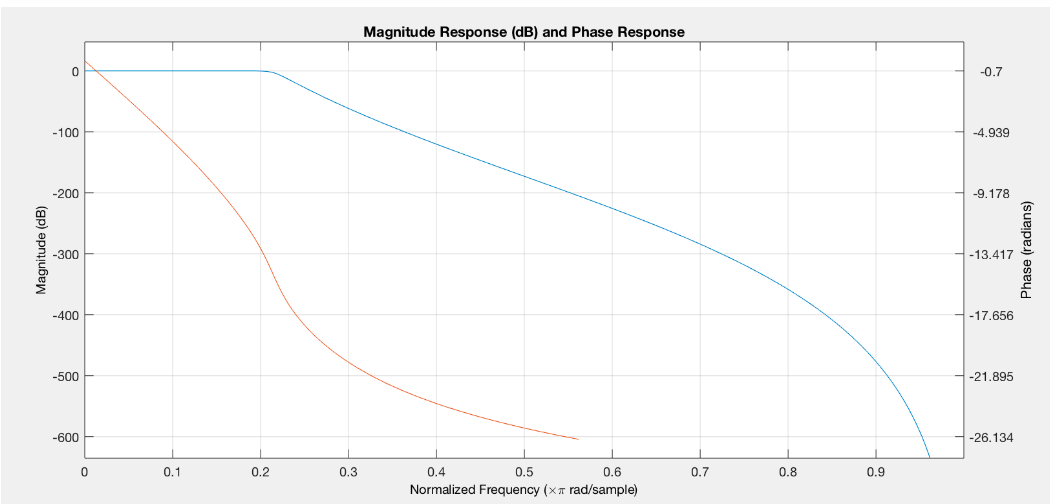
EE312 LAB REPORT#4

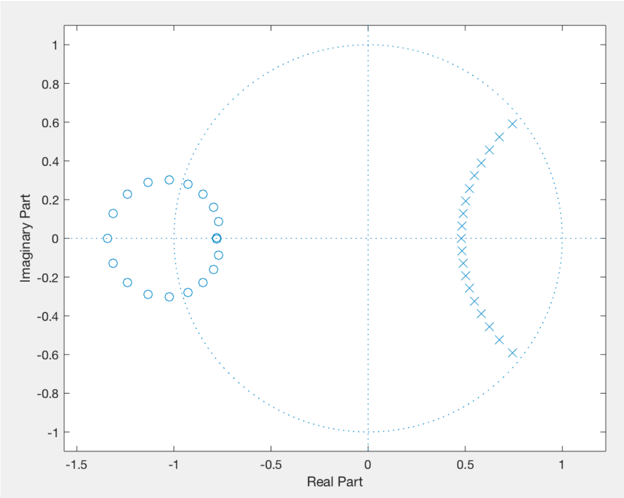
SPRING 2020

Problem 4.1

a) In this part, we adjusted pass-band frequencies of our low pass filter. After that, we determined the minimum order of the filter (N) as 19 by buttord command. Then, we plot their magnitude and phase response by fvtool command. According to magnitude and z-plane plots, it is obviously low pass filter and its phase response is not linear which is because of it is IIR filter.

```
[x,fs] =audioread('music2.wav');
Wp = (2*pi*500)/fs;
Ws = (2*pi*700)/fs;

Rp= 0.2;
Rs= 45 ;

[N, Wn] = buttord(Wp, Ws, Rp, Rs);
[b,a] = butter(N,Wn,'low'); %Low-pass

[z,p,k] = butter(N,Wn,'low');
sos = zp2sos(z,p,k);
figure(1);
fvtool(sos,'Analysis','freq')

figure(1);
zplane(b,a);

% figure(2);
% freqz(b,a);

x1= filter(b,a,x);
soundsc(x1,fs);
```

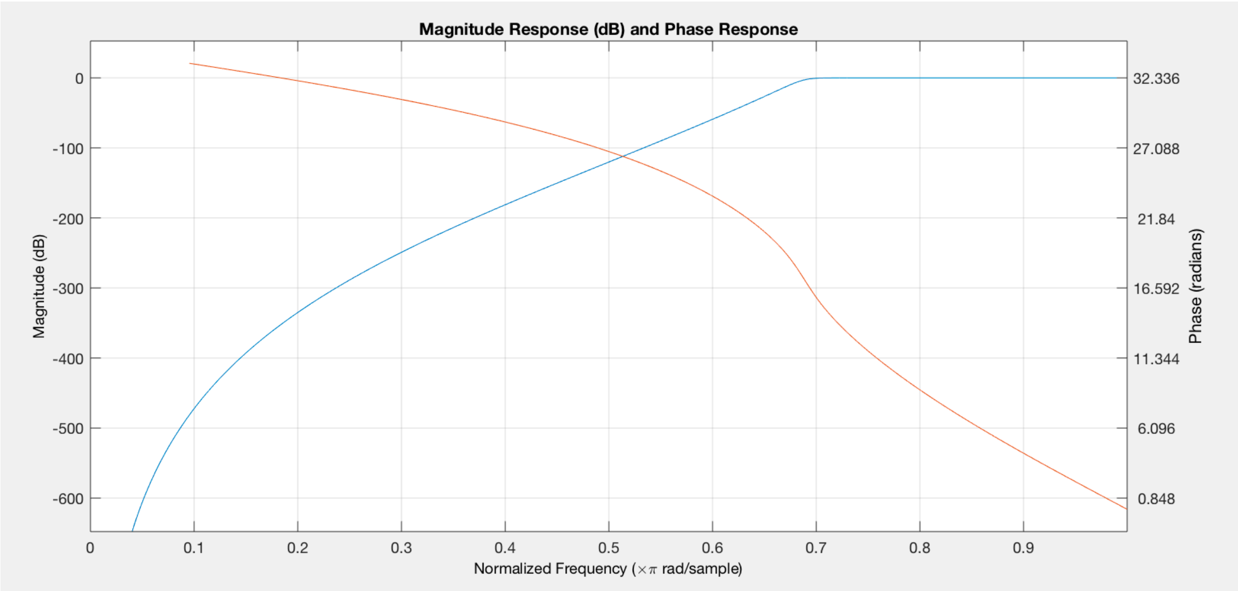


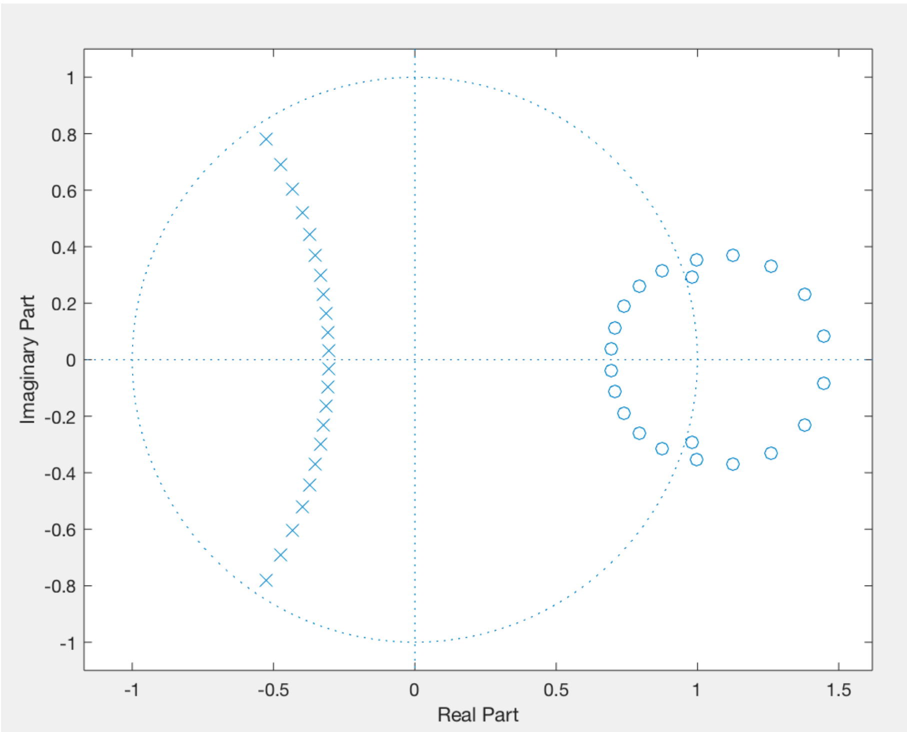


b	*1x20 double*
fs	16000
k	3.4358e-11
N	19
p	*19x1 complex double*
Rp	0.2000
Rs	45
sos	*10x6 double*

b) **In this part, we adjusted pass-band frequencies of our high pass filter. After that, we determined the minimum order of the filter (N) as 22 by buttord command. Then, we plot their magnitude and phase response by fvtool command. According to magnitude and z-plane plots, it is obviously high pass filter and its phase response is not linear which is because of it is IIR filter.**

```
% IIR HIGH-PASS
[x,fs] = audioread('music2.wav');

Wp = (2*pi*1700)/fs;
Ws = (2*pi*1900)/fs;
Rp = 0.2;
Rs = 45;

[N,Wn] = buttord(Wp,Ws,Rp,Rs); % Min order calculated
[b,a] = butter(N,Wn,'high'); % High-pass

[z,p,k] = butter(N,Wn,'high');
sos = zp2sos(z,p,k);
figure(1); fvtool(sos,'Analysis','freq')

% figure(2);
% freqz(b,a);

figure(2);
zplane(b,a);

x1= filter(b,a,x);
soundsc(x1,fs);
```





b	*1x23 double*
fs	16000
k	6.9178e-10
N	22
p	*22x1 complex double*
Rp	0.2000
Rs	45
sos	*11x6 double*

d) 1&2

```matlab
% IIR HIGH-PASS
[x,fs] = audioread('music2.wav');

Wp = (2*pi*1700)/fs;
Ws = (2*pi*1900)/fs;
Rp = 0.2;
Rs = 45;

[N,Wn] = buttord(Wp,Ws,Rp,Rs); % Min order calc
[b,a] = butter(N,Wn,'high'); % High-pass

[z,p,k] = butter(N,Wn,'high');
sos = zp2sos(z,p,k);
figure(1); fvtool(sos,'Analysis','freq')

% figure(2);
% freqz(b,a);

figure(2);
zplane(b,a);

x1= filter(b,a,x);
soundsc(x1,fs);
```

```matlab
[x,fs] =audioread('music2.wav');
Wp = (2*pi*500)/fs;
Ws = (2*pi*700)/fs;

Rp= 0.2;
Rs= 45 ;

[N, Wn] = buttord(Wp, Ws, Rp, Rs);
[b,a] = butter(N,Wn,'low'); %Low-pass

[z,p,k] = butter(N,Wn,'low');
sos = zp2sos(z,p,k);
figure(1);
fvtool(sos,'Analysis','freq')

figure(1);
zplane(b,a);

% figure(2);
% freqz(b,a);

x1= filter(b,a,x);
soundsc(x1,fs);
```

We used filter command to prevent from distortion while we are listening. If we examine for low pass IIR filter, high frequency components are suppressed by filter, we hear low frequency as dominant voice. For high pass filter, low frequency components are suppressed by filter, we hear high frequency as dominant voice.

=> We see same attributes also in FIR filters.

Problem 4.2

***For this part, we need to translate the specs given above to the FIR filter input specs in MATLAB. We referred this link to calculate dev.**

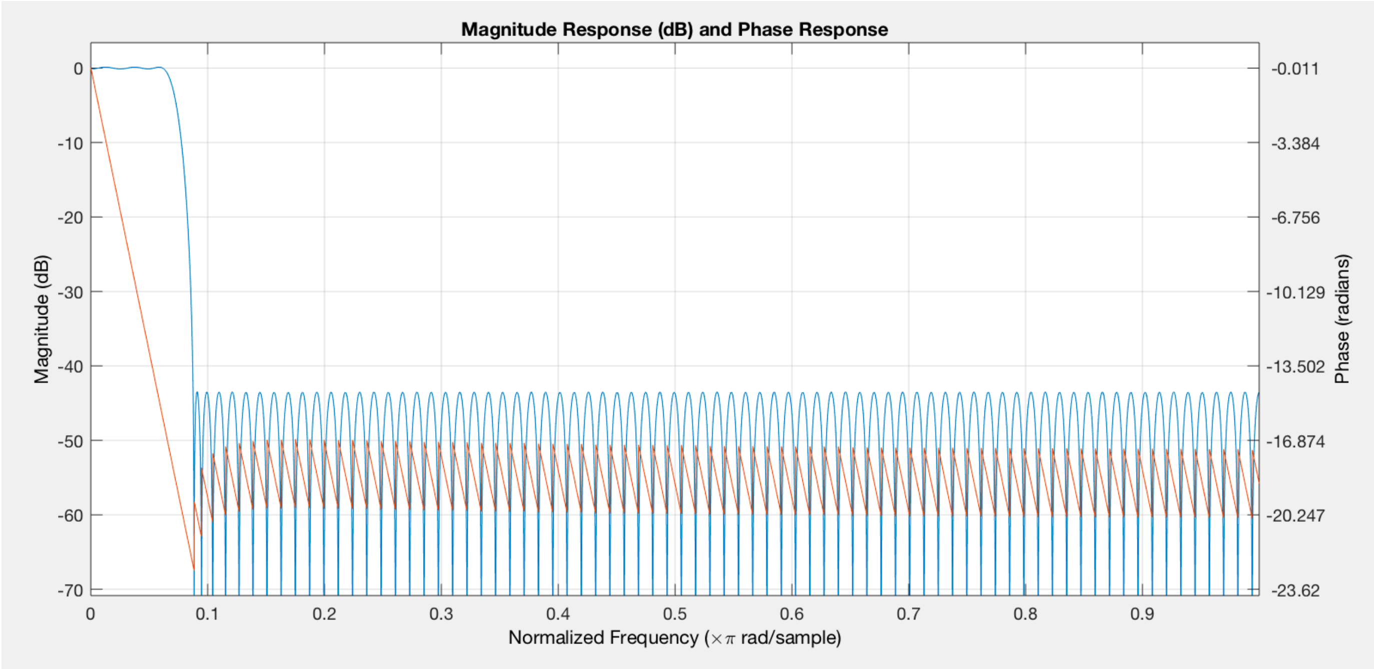
http://matlab.izmiran.ru/help/toolbox/signal/firpmord.html

For FIR_LP:

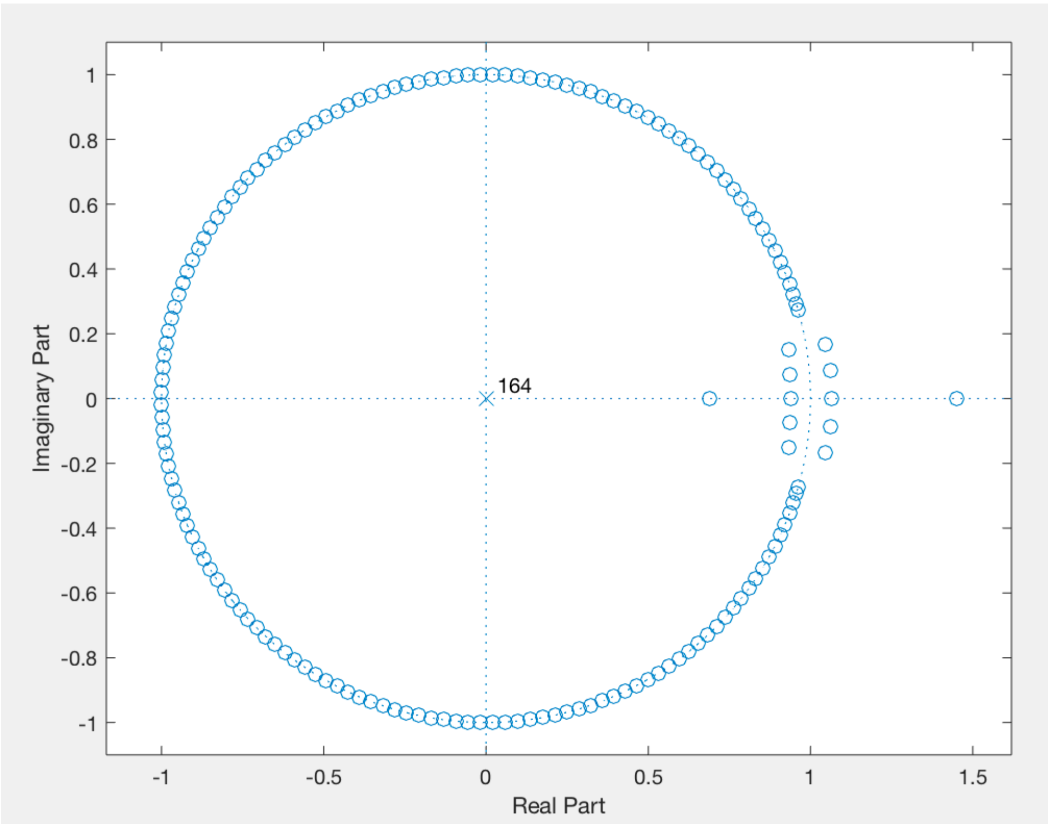
Actually, we think that , in our case, voice of FIR filter is more obvious and listenable when compared with IIR filter. Again, we adjusted the pass-band frequencies and we found minimum order as 164. Also, we determined amplitude as well in this part. Additionally, we adjusted dev parameter because of MATLAB working principle. After that, we got the plot of the magnitude, phase response and zero plots.

```
% FIR-LP
[x,fs] =audioread('music2.wav');
f = [500 700]; % Cut-off frequencies
Rp= 0.2 ;
Rs= 45;
a = [1 0]; % Desired amplitude

dev = [(10^(Rp/20)-1)/(10^(Rp/20)+1) 10^(-Rs/20)]; % Translating into FIR specs
[n,fo,ao,w] = firpmord(f,a,dev,fs);
b = firpm(n,fo,ao,w);

% figure(1); freqz(b,fs); % Plots the magnitude responses
figure(1); h = fvtool(b);
figure(2); zplane(b);

x1=filter(b,1,x);
soundsc(x1,fs);
```

f	[500,700]
fo	[0;0.0625;0.0875;1]
fs	16000
n	164
Rp	0.2000
Rs	45



=>Since filter is FIR, our phase response is linear.



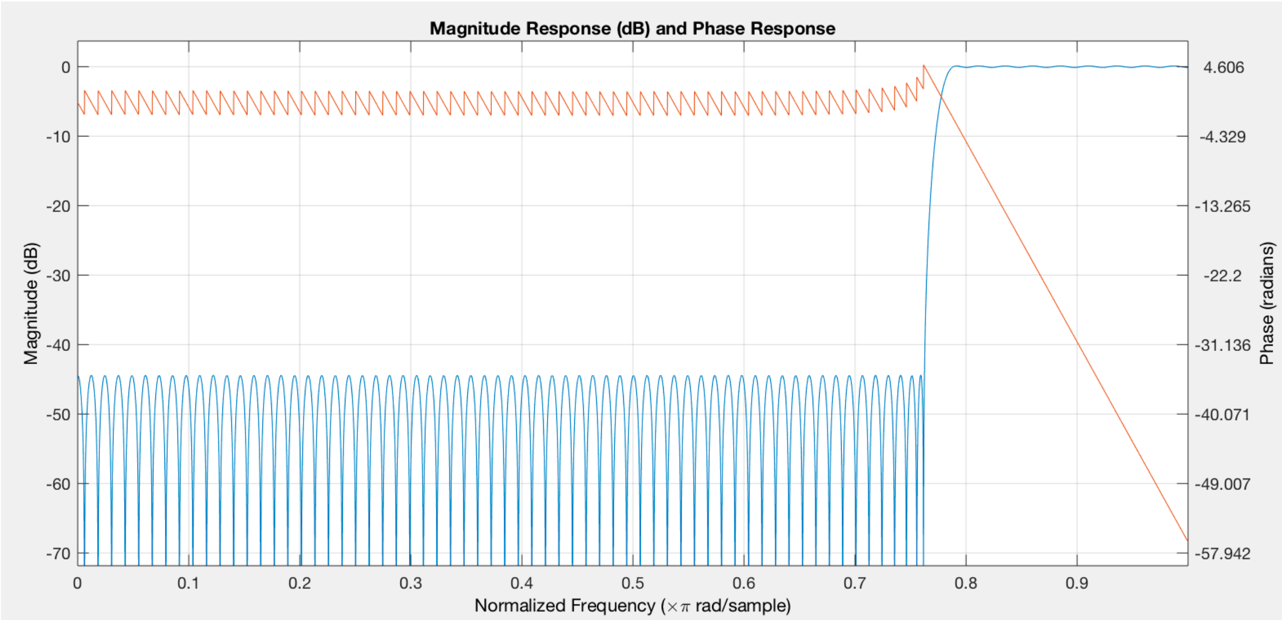
=>Since filter is FIR, zeros only exists at the origin.

For FIR_HP:

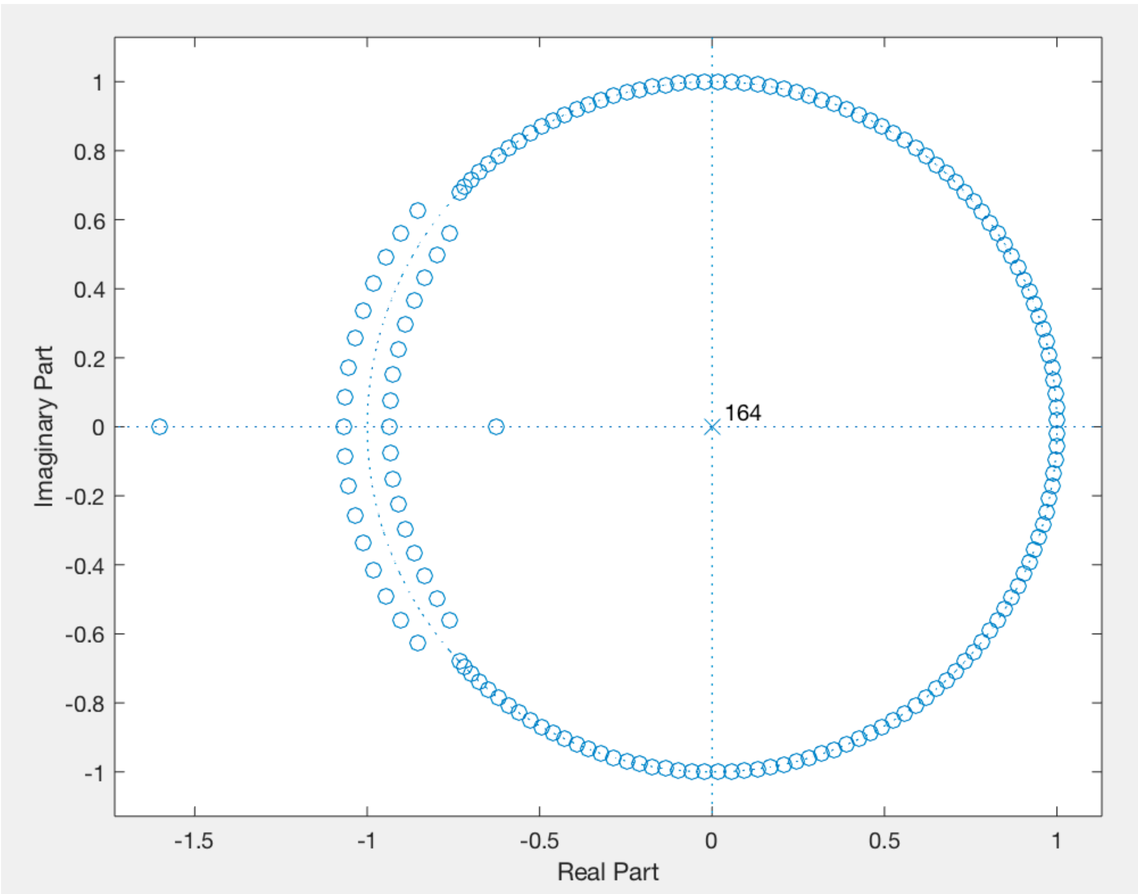
Actually, we think that in our case, filtered voice of FIR filter is more obvious and listenable when we compared with IIR filter. Again, we adjusted the pass-band frequencies and we found minimum order as 164. Also, we determined amplitude as well in this part. Additionally, we adjusted dev parameter because of MATLAB working principle. After that, we got the plot of the magnitude, phase response and zero plots.

```
% FIR HIGH-PASS
[x,fs] = audioread('music2.wav');

Rp = 0.2;
Rs = 45;
f = [1700 1900]; % Cut-off frequencies
a = [1 0]; % Desired amplitude

dev = [(10^(Rp/20)-1)/(10^(Rp/20)+1) 10^(-Rs/20)];
[n,fo,ao,w] = firpmord(f,a,dev,fs);
b = firpm(n,fo,ao,w);
z = firlp2hp(b); %Low-pass to High-pass

%figure(1); freqz(z,1,fs);% Plots the magnitude and phase respo
h = fvtool(z);
figure(2); zplane(z);% Pole-zero plot

x1=filter(z,1,x);
soundsc(x1,fs);
```

fs	16000
h	*1x1 fvtool*
n	164
Rp	0.2000
Rs	45



=>Since filter is FIR, our phase response is linear.



=>Since filter is FIR, zeros only exists at the origin.