Barak Barclay

Dr. Al Batten

ECE2610-001

Lab 5

**Section 3.1:**

(a)

filter 1: b0=1, b1=-2\*cos(0.44\*pi)=0.374, b2=1

filter 2: b0=1, b1=-2\*cos(0.7\*pi)= 1.176, b2=1

(b)

Code:

n=0:149;

xn=5\*cos(0.3\*pi.\*n)+22\*cos(0.44\*pi.\*n-pi/3)+22\*cos(0.7\*pi.\*n-pi/4);

(c)

Code:

f1=[1 -2\*cos(0.44\*pi) 1];

wn=firfilt(f1,xn);

f2=[1 -2\*cos(0.7\*pi) 1];

yn=firfilt(f2,wn);

(d)

Code:

n=0:39;

xn=5\*cos(0.3\*pi.\*n)+22\*cos(0.44\*pi.\*n-pi/3)+22\*cos(0.7\*pi.\*n-pi/4);

subplot(2,1,1),stem(n,xn(n+1));

f1=[1 -2\*cos(0.44\*pi) 1];

wn=firfilt(f1,xn);

f2=[1 -2\*cos(0.7\*pi) 1];

yn=firfilt(f2,wn);

subplot(2,1,2),stem(n,yn(n+1));

Yss[n]=9.412\*cos(0.3\*pi.\*n-0.6\*pi)+0.00572\*cos(0.44\*pi.\*n-1.21\*pi)+0.033\*cos(0.7\*pi.\*n-1.65\*pi)

(e)

Code:

n=0:39;

xn=5\*cos(0.3\*pi.\*n)+22\*cos(0.44\*pi.\*n-pi/3)+22\*cos(0.7\*pi.\*n-pi/4);

f1=[1 -2\*cos(0.44\*pi) 1];

wn=firfilt(f1,xn);

f2=[1 -2\*cos(0.7\*pi) 1];

yn=firfilt(f2,wn);

stem(n,yn(n+1));

hold on

n=5:40;

yn2=9.412\*cos(0.3\*pi.\*n-0.6\*pi)+0.00572\*cos(0.44\*pi.\*n-1.21\*pi)+0.033\*cos(0.7\*pi.\*n-1.65\*pi);

stem(n,yn2)



(f)

Since the cascaded filter has a length of 5, the input signal has to run through 5 start-up points before the all of the coefficients in the filter are being used.

**Section 3.2:**

(a)

Code:

n=0:9;

f=(2/10).\*cos(0.44\*pi\*n);

w=-pi:(pi/500):pi;

H=freqz(f,1,w);

plot(w,abs(H))



When w=0.3pi, gain=0.285\*5=1.425

When w=0.44pi, gain=1.095\*22=24.09

When w=0.7pi, gain=0.285\*22=6.26

(b)

Code:

n=0:19;

f=(2/20).\*cos(0.44\*pi\*n);

w=-pi:(pi/500):pi;

H=freqz(f,1,w);

plot(w,abs(H))

Graph for L=10 in part a.

For L=20:



For L=40:



Other Information:

When L=10, passband width = 0.162pi

When L=20, passband width = 0.088pi

When L=40, passband width = 0.042pi

L has an inverse relationship with passband width.

(c)

When an input signal goes through a filter, the magnitude of the frequency response is multiplied by the amplitude of the input signal causing some frequencies to be increased and/or decreased depending on the filter. Looking at graph of the magnitude of the frequency response in part a, you can see how one component can passed while reducing or rejecting others.

(d)

Code:

done=0;

L=2;

while (~done)

L=L+1;

n=0:(L-1);

f=2/L.\*cos(0.44\*pi\*n);

w=-pi:pi/500:pi;

H=freqz(f,1,w);

jkl=[1:150 350:650 850:1001];

done=max(abs(H(jkl)))<max(abs(H)\*0.1);

end

L-1

Other information:

Shortest filter length 🡪 L=38

(e)

Code:

L=38;

n=0:23;

f=(2/L).\*cos(0.44\*pi\*n);

n2=0:99;

xn=5\*cos(0.3\*pi.\*n2)+22\*cos(0.44\*pi.\*n2-pi/3)+22\*cos(0.7\*pi.\*n2-pi/4);

subplot(2,1,1);

stem(xn(n2+1))

yn=conv(f,xn);

subplot(2,1,2);

stem(yn(n2+1))



Other Information:

The filter has removed and reduced some components by multiplying the amplitude of the input signal by the magnitude of the frequency response which is either zero or a really low number at those frequencies.

(f)

Code:

L=38;

n=0:23;

f=(2/L).\*cos(0.44\*pi\*n);

w=-pi:(pi/500):pi;

H=freqz(f,1,w);

plot(w,abs(H))



Other Information:

In the plot above, it shows the magnitude of the frequency response to be .6561 at +/-0.44pi rad. At |w|=<.3pi rad, the magnitude is less than 0.1. At 0.7pi=<|w|=<pi the magnitude is less than 0.1. Since the magnitude of the frequency response is multiplied by the amplitude of the input signal to create the output signal, the output signal will have reduced amplitudes at the frequencies where the magnitude is 0.1 or less, mentioned above, and have only slightly reduced frequencies at +/-0.44pi rad.