Name: Key

Section: 2111

EE432 Fall 2011 Exam 2

Problem	Possible Points	Score
1	20	
2	40	
3	40	
Total	100	

- Turn this in Friday, Nov. 4, 2011 at the beginning of class.
- This is an open book, open notes and open computer exam. USE ONLY YOUR COURSE TEXTBOOK ("Fundamentals of Digital Signal Processing"), *your* notes from this course, and MATLAB HELP.
- You ***must show your work*** to get full credit for problems. Use additional sheets as necessary.
- Label your plots carefully, and turn in all of your code when using MATLAB. Put all the code into one document to turn it in to save paper.
- **On any additional sheets you turn in (including sheets with MATLAB code), be sure to indicate which problem(s) are being addressed. If what you turn in is disorganized and not clearly labeled, you may lose points.**

- If you are stuck on a problem, you may ask for guidance…but it might cost you in points. You ask your question, and I will let you know how much it will cost. Then you can agree to obtain the guidance for the specified number of points off of your final score, if you wish.

1. (20 pts) Transfer Functions.

   a. Find the transfer function (H(z)) of a system that has a difference equation given by :
   $10y[n]+9y[n-1]-5y[n-5]=8x[n]+4x[n-1]-x[n-2]+2x[n-3]$.
   Put your answer into positive powers of z. Ensure that $a_0 = 1$.

$$y[n]+0.9y[n-1]-0.5y[n-5]=0.8x[n]+0.4x[n-1]-0.1x[n-2]+0.2x[n-3]$$

$$Y(z)\left(1+0.9z^{-1}-0.5z^{-5}\right)=X(z)\left(0.8+0.4z^{-1}-0.1z^{-2}+0.2z^{-3}\right)$$

$$H(z)=\frac{Y(z)}{X(z)}=\frac{0.8+0.4z^{-1}-0.1z^{-2}+0.2z^{-3}}{1+0.9z^{-1}-0.5z^{-5}}\quad\frac{z^5}{z^5}$$

$$H(z)=\frac{0.8z^5+0.4z^4-0.1z^3+0.2z^2}{z^5+0.9z^4-0.5}$$

   b. Find the system's poles and zeros, and list them here.

Poles:

$-0.9488 \pm j0.4251$

$0.1724 \pm j0.7788$

0.7428

Zeros: $0, 0, -0.9263$

$0.2132 \pm j0.4738$

c. Using MATLAB, create a pole-zero plot for this system, and print out a copy of a plot of its transfer function (i.e., frequency response) with magnitude and phase on the same plot. Ensure that all your poles and zeros from step *b* appear. Is the system stable? Why or why not?

Not stable – 2 poles outside unit circle
plots attached.

d. Using long division, find the 1st 3 non-zero terms of the system's impulse response (h[n]).

$$\begin{array}{r|l} & 0.8 - 0.32z^{-1} + 0.188z^{-2} + \dots \\ \hline z^5 + 0.9z^4 - 0.5 & 0.8z^5 + 0.4z^4 - 0.1z^3 + 0.2z^2 \\ & \underline{0.8z^5 + 0.72z^4 \qquad\qquad -0.4} \\ & -0.32z^4 - 0.1z^3 + 0.2z^2 + \qquad +0.4 \\ & \underline{-0.32z^4 - 0.288z^3 + \dots} \\ & 0.188z^3 + \dots \end{array}$$

stop here

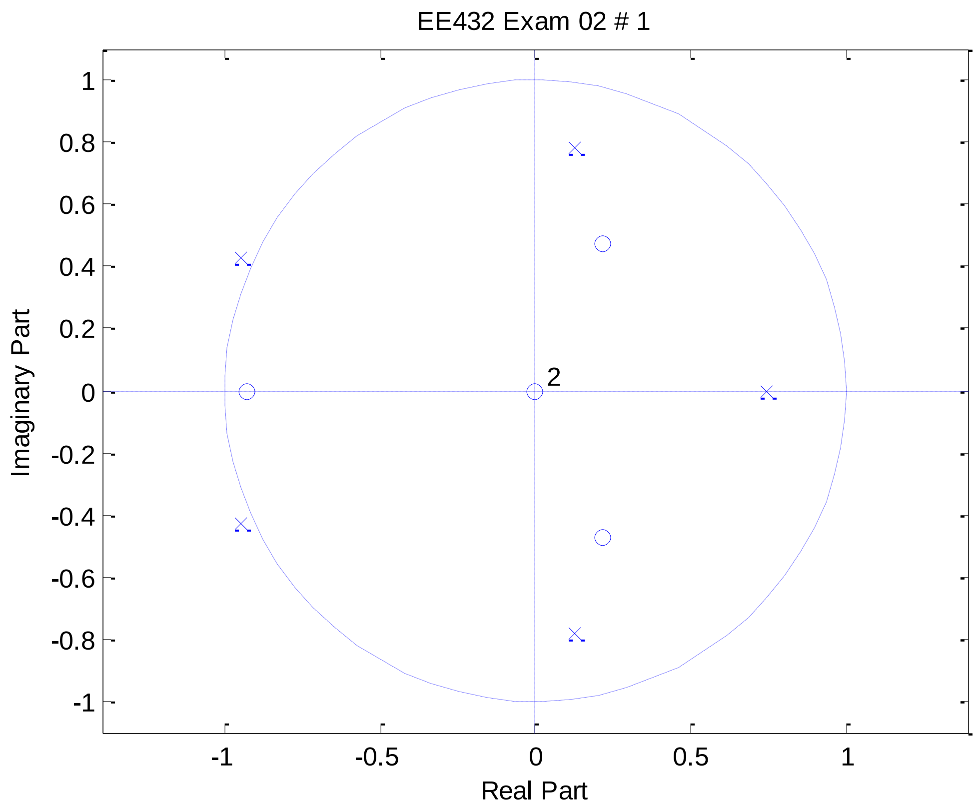
$$H(z) = 0.8 - 0.32z^{-1} + 0.188z^{-2} + \dots$$

So

$$\boxed{h[n] = 0.8\delta[n] - 0.32\delta[n-1] + 0.188\delta[n-2] + \dots}$$

```
% Problem 1:
b=[0.8 0.4 -0.1 0.2 0 0];
a=[1 0.9 0 0 0 -0.5];
disp('zeros: ')
roots(b)
disp('poles: ')
roots(a)

figure(1), zplane(b,a),title('EE432 Exam 02 # 1')
fvtool(b,a),title('EE432 Exam 02 # 1')
```



40

(~~[illegible]~~ pts) Frequency response.

a. A system has the following poles and zeros. Using MATLAB tools, determine the equation for its frequency response $H(e^{j\Omega})$. Your answer must consist of a single numerator polynomial and a single denominator polynomial (i.e., not factored).

<u>*zeros*</u>:

$z_1 = -0.85,$

$z_2, z_3 = 1.01e^{\pm j\frac{\pi}{3}}$

<u>*poles*</u>:

$p_1, p_2 = 0.35 \pm j0.95$

$p_3, p_4 = 0.8e^{\pm j\frac{2\pi}{3}}$

$p_5 = 0.75$

$p_6 = -0.9$[illegible]

From MATLAB,

$b = [1 \quad -0.16 \quad 0.1616 \quad 0.8671]$

$a = [1 \quad 0.25 \quad 0.445 \quad 0.4703 \quad -0.0341 \quad -0.1527 \quad -0.4428$

$$H(e^{j\Omega}) = \frac{1 - 0.16e^{-j\Omega} + 0.1616e^{-j2\Omega} + 0.8671e^{-j3\Omega}}{1 + 0.25e^{-j\Omega} + 0.445e^{-j2\Omega} + 0.4703e^{-j3\Omega} - 0.0341e^{-j4\Omega} - 0.1527e^{-j5\Omega} - 0.4428e^{-j6\Omega}}$$

b. Generate and turn in a pole-zero plot for this filter using MATLAB. Is the filter stable? Why or why not?

- Plot + code attached
- <u>Not</u> stable - 2 poles outside unit circle

c. <u>**Without**</u> using *fvtool*, generate a plot of the magnitude (NOT in dB) and phase (in radians) of the frequency response of this filter on a 2x1 subplot. Using your plot, determine the equation for the output of the filter if the input $x[n] = 10\ \sin(0.3\pi n + 1.1)$.

$$y[n] = 10 \cdot |H(0.3\pi)| \cdot \sin(0.3\pi n + 1.1 + \angle H(0.3\pi))$$

note: $0.3\pi = 0.9425$ rad

$$= 10\,(0.24)\sin(0.3\pi n + 1.1 - 0.083) = \boxed{2.4\sin(0.3\pi n + 1.017)}$$

d. When a signal is input to this filter, will any of the input frequency content <u>disappear</u> as the signal passes through the filter? If so, with a sample frequency of 44.1 kHz, what frequency(ies) will disappear? If not, with a sample frequency of 44.1 kHz, what frequency corresponds to the highest gain?

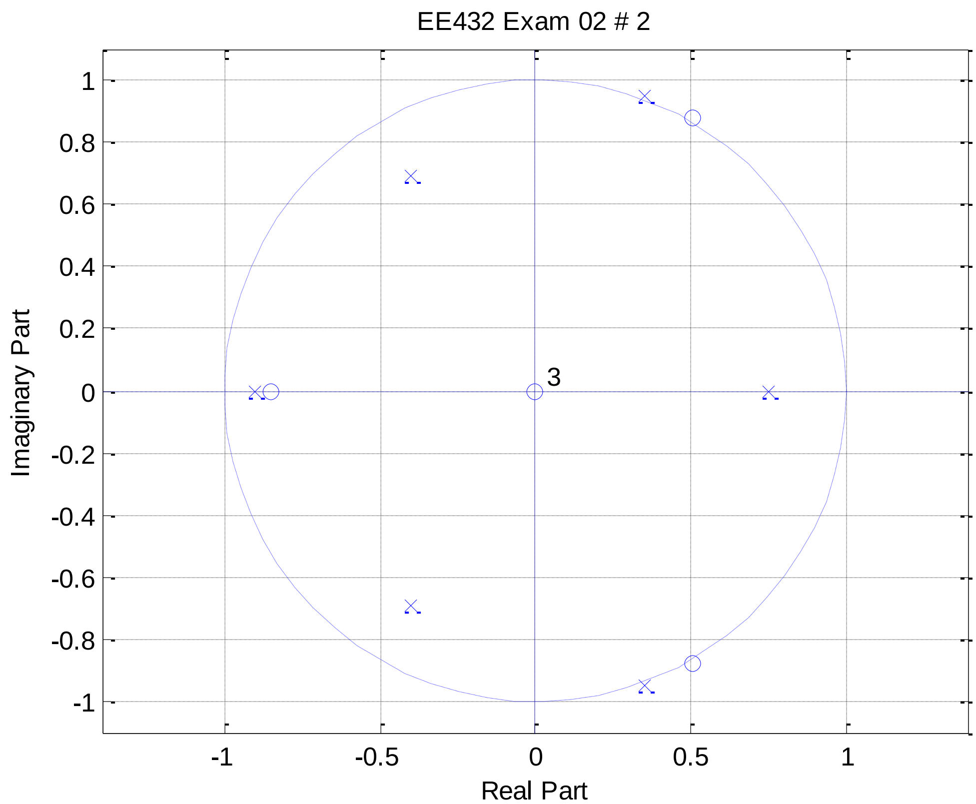
- No zeros on unit circle, so no freqs disappear
- max gain at $\Omega = 1.219$ rad, or $f = \frac{1.219(44.1\text{kHz})}{2\pi} = \boxed{8.56\text{kHz}}$

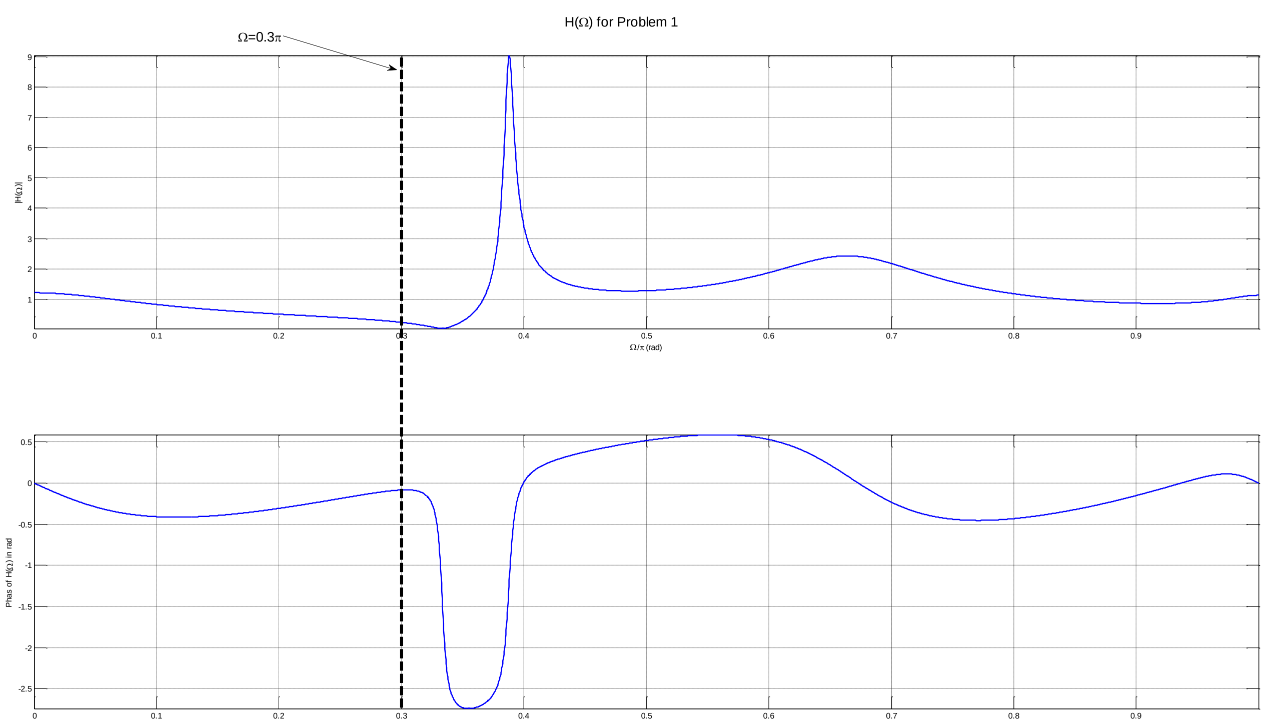
$\Omega = \frac{2\pi f}{f_s} \quad f = \frac{\Omega f_s}{2\pi}$

```matlab
% Problem 2:
z1=-0.85;
z2=1.01*exp(j*pi/3);
z3=1.01*exp(-j*pi/3);
p1=0.35+j*0.95;
p2=0.35-j*0.95;
p3=0.8*exp(j*2*pi/3);
p4=0.8*exp(-j*2*pi/3);
p5=0.75;
p6=-0.9;

b=conv([1 -z1],[1 -z2]);
b=conv(b,[1 -z3]);
a=conv([1 -p1],[1 -p2]);
a=conv(a,[1 -p3]);
a=conv(a,[1 -p4]);
a=conv(a,[1 -p5]);
a=conv(a,[1 -p6]);

figure(2),zplane(b,a),title('EE432 Exam 02 # 2'),grid on
Om=0:0.001:pi;  % Omega
num=0;
for k=1:length(b)
    num=num+b(k)*exp(-j*(k-1)*Om);
end
denom=0;
for k=1:length(a)
    denom=denom+a(k)*exp(-j*(k-1)*Om);
end
%denom=1+0.349*exp(-j*Om)+0.3807*exp(-j*2*Om)+0.5722*exp(-j*3*Om)-0.0793*exp(-
j*4*Om)-0.1153*exp(-j*5*Om)-0.4915*exp(-j*6*Om);
H=num./denom;
figure(3),subplot(2,1,1),plot(Om,abs(H)),xlabel('\Omega
(rad)'),ylabel('|H(\Omega)|'),grid on
subplot(2,1,2),plot(Om,angle(H)),xlabel('\Omega (rad)'),ylabel('Phas of H(\Omega)
in rad'),grid on
```


EE432 Exam 02 # 2
Imaginary Part
Real Part
3
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
-1
-0.5
0
0.5
1


H(Ω) for Problem 1
Ω=0.3π
|H(Ω)|
Ω/π (rad)
Phas of H(Ω) in rad
Ω/π (rad)

(40 pts) Download the file called "EE432-Exam02.wav" from the course website. This is a music clip from Queen (*Another One Bites the Dust*) corrupted by noise (some very annoying tones). If you read in the signal using *wavread*, then run the *fvtool*(1,1,*a*) command (if *a* is the input signal), then *fvtool* will display the frequency content of the signal *a* and the annoying tones should be apparent.

Design a windowed FIR low pass filter to remove or attenuate the noise found in the signal, while keeping as much frequency content of the actual signal as possible, to maximize the quality of the filtered signal. The goal is to filter the signal so you cannot hear the annoying tones.

Design constraints: Your filter should have as small a transition width as possible, must have fewer than 200 coefficients, and must suppress side lobes (stop band attenuation) > 50 dB.

- Describe how you designed your filter(s) on a separate page (i.e., walkthrough the steps from Table 9.4 from the textbook). Include your determination of which tones needed to be removed.
- Create a plot of the magnitude of the frequency response of your filter (in dB) using *fvtool*, and on the same plot, display the frequency content of the *corrupted* signal. **Turn in this plot.**
- Fill in the information below from your design:

  Frequencies (Hz) of annoying tones: 12.1275 kHz, 13.7813 kHz, 15.9863 kHz

  Pass band edge frequency = 11.1275 k Hz

  Transition width = 1 kHz

  f_1 = 11.6275 kHz Ω_1 = 0.5289π rad

  Window type: Hamming # of coefficients: 151

  Stop band attenuation (from your filter design process): 55 dB
- Fill in the information below from your frequency response magnitude plot:

  Actual stop band attenuation: -53.46 dB

  -3dB Bandwidth: 11.545 kHz Hz

My design - yours will probably be different.

(Go to next page)

(3) LPF design

- 3 annoying tones at 0.55π, 0.625π, 0.725π based on using fvtool(1,1,x)
- Since $f_s = 44.1$ kHz, these correspond to:

$$f_1 = 0.55\pi \frac{f_s}{2\pi} = 12.1275 \text{ kHz}$$

$$f_2 = 0.625\pi \frac{f_s}{2\pi} = 13.7813 \text{ kHz}$$

$$f_3 = 0.725\pi \frac{f_s}{2\pi} = 15.9863 \text{ kHz}$$

- Design filter to remove freqs ≥ 12.1275 kHz
- Since stop band attenuation > 50 dB, can use a Hamming, Blackman or Kaiser window.
- For the same transition width, the Hamming window uses the fewest # coefficients.
- let: stop band edge freq = 12.1275 kHz

  desired pass band edge freq = 11.1275 kHz
  (gives a 1 kHz transition width)

(1) $f_1 = 11.1275 \text{kHz} + \frac{TW}{2} = 11.6275 \text{ kHz}$

(2) $$\Omega_1 = \frac{2\pi f_1}{f_s} = 2\pi \frac{11.6275 \times 10^3}{44.1 \times 10^3} = 0.5289\pi$$

$$h_1[n] = \frac{\sin(0.5289\pi n)}{n\pi}$$

(3) window = Hamming

$$\# \text{ terms} = \frac{3.44 \cdot f_s}{T_W} = 151.7 \rightarrow 151 \text{ terms}$$

(4) $h[n] = h_1[n]\, w[n]$ where $w[n] = 0.54 + 0.46 \cos \frac{2\pi n}{150})$

For $-75 \leq n \leq 75$

TOPS 35500

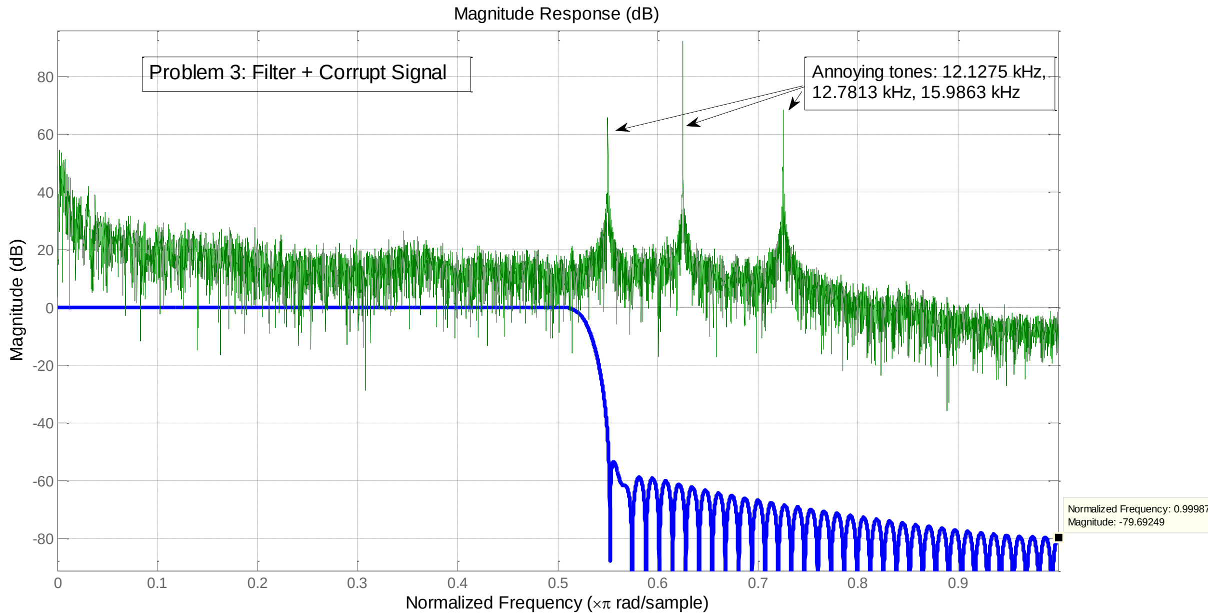
- Filter the corrupt signal from "EE432-Exam02.wav" with your LPF.

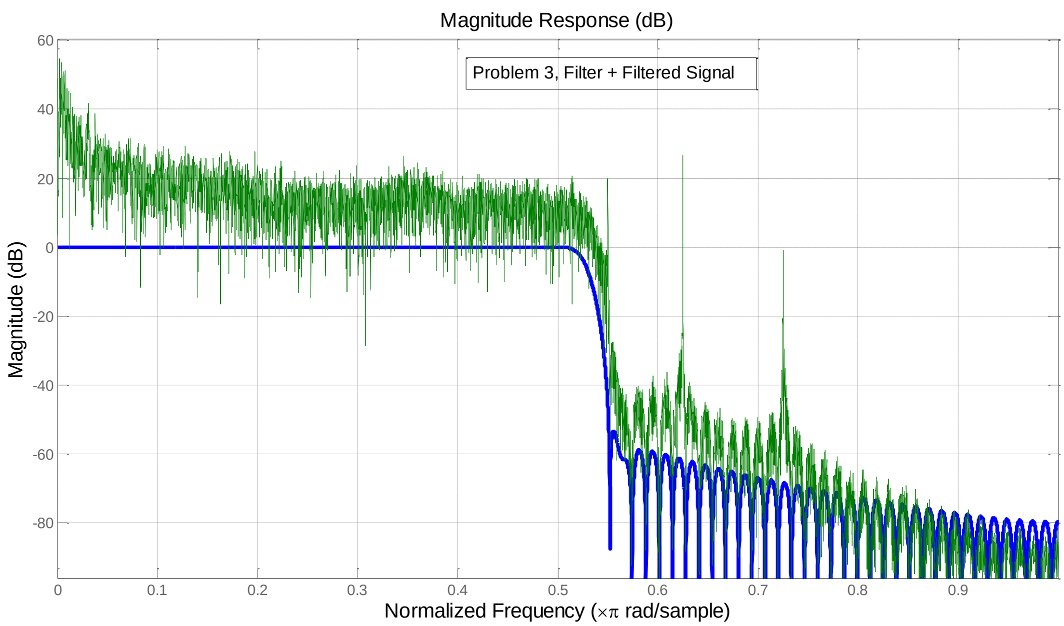
- Listen to the filtered signal…you should hear the music, but NOT the annoying tones.

- Create a plot of the magnitude of the magnitude of the frequency response of your filter (in dB) using *fvtool*, and on the same plot, display the frequency content of the *filtered* signal. **Turn in this plot.** Describe how well your filter seems to work, both audibly and based on the plot you just created.

  3 Annoying tones suppressed enough that I couldn't hear them. The peaks in the freq spectrum are reduced > 50 dB.

- Normalize your filtered signal using your *normalize432* function, write out the normalized signal using *wavwrite*, call the file "EE432-Exam02-fixed.wav," and email it to the professor.


Magnitude Response (dB)
Problem 3: Filter + Corrupt Signal
Annoying tones: 12.1275 kHz, 12.7813 kHz, 15.9863 kHz
Normalized Frequency: 0.99987
Magnitude: -79.69249
Magnitude (dB)
Normalized Frequency (×π rad/sample)


Magnitude Response (dB)
Problem 3, Filter + Filtered Signal
Magnitude (dB)
Normalized Frequency (×π rad/sample)