ECE-486 24-hour "Extravaganza"
Spring 2002
Due 2:00 PM, Saturday May 4

- Do your own work. Do not discuss the problems with any other person. Do not talk about the problems with any other student until after the due-time. Don't even say things like "Number 2 seems hard", or "Did number 3 seem trivial to you?". Do not incur the wrath of the instructor! Be an honest person.
- You may use MATLAB or MATHCAD for any of the problems. *Please do so.* Numerical errors will be harshly graded. Check your answers!
- You can use your own notes, and any published material.
- Please check the course www site frequently. Any test corrections or clarifications will be posted there. http://www.eece.maine.edu/~hummels/classes/ece486
- For each problem, make (labeled) plots illustrating that your filter designs have met the stated specifications. Indicate on your plots the filter order used for the design.
- Make sure that your submitted filter coefficients are real.

1. Design minimum order FIR and IIR bandpass filters which meet the following specifications.

Passband:	$0.13 < \|f\| < 0.2$	Gain $= 30$ dB ± 0.5 dB
Stopband:	$0 < \|f\| < 0.1$ or $0.22 < \|f\| < 0.5$	Gain ≤ -50 dB

   Indicate the order of your filter, the design method used (including how you selected filter design parameters) and give plots showing that you've met the desired specs. Your FIR filter should be a linear phase filter. Evaluate the number of multiplications required to calculate each filter output.

   Submit your FIR and IIR designs for grading using the MATLAB commands

```
ece486_submit('lastname_FIR',h);
ece486_submit('lastname_IIR',b,a);
```

   Use the `ece486_check()` program to check you submission.

2. An FIR filter is to be designed to compensate for a linear distortion characteristic that is introduced by a transmission line. The desired magnitude and phase characteristic for the filter in the band $0.7 < \omega < 1.3$ are described by

$$
\begin{aligned}
|H(\omega)| &= 0.4 + 2|\omega - 1| - 2(\omega - 1)^2 \\
\angle H(\omega) &= 0.9\cos(3(\omega - 1))
\end{aligned}
$$

   In addition to approximating the above characteristic for $0.7 < \omega < 1.3$, the filter is to reject the bands $\omega < 0.6$ and $\omega > 1.4$ with at least 60 dB of attenuation.

   Design an filter that accomplish this task. The filter should introduce the above magnitude and phase characteristic in addition to the linear phase characteristic associated with the filter delay. If $H_{FIR}(\omega)$ is the transfer function for your FIR filter, and the filter output is delayed D samples from the input, we require that

$$|H(\omega)e^{-j\omega D} - H_{FIR}(\omega)| < 0.05 \qquad 0.7 < \omega < 1.3$$

   Also, the filter must meet the stopband specification

$$20\log_{10}(|H_{FIR}(\omega)|) \leq -60 \qquad 0 < |\omega| < 0.6 \text{ or } 1.4 < |\omega| < \pi$$

   (a) Turn in a description of your design procedure, and a plot showing your filter response.

(b) Indicate the delay associated with your design (the value of "D").

(c) Is your design a linear phase filter? Justify your answer.

(d) Submit your design for grading using the MATLAB command

```
ece486_submit('lastname_problem2',h);
```

where "h" is the name of the vector containing your filter coefficients. Use the `ece486_check()` program to check you submission.

3. To transmit stereo signals for broadcast FM, a composite signal containing both the left and right channels is created at the transmitter. The signal is described by

$$s(t) = m_s(t) + m_d(t)\cos(2\pi F_c t) + \cos(2\pi(F_c/2)t)$$

where $m_s(t)$ denotes the sum of the left and right channels, and $m_d(t)$ denotes the difference of the left and right channels. The audio signals all have bandwidth 15 kHz, and the value of F_c is 38 kHz. The $m_d(t)$ term above has been modulated to the 38 kHz carrier so that it does not interfere with the sum term $m_s(t)$. A monaural receiver would simply filter out the (baseband) 15 kHz signal $m_s(t)$, and drive the speaker with the sum of the left and right channels. A stereo receiver must also demodulate the $m_d(t)$ term, and then recreate the left(right) channel by adding(or subtracting) $m_s(t)$ and $m_d(t)$. The 19 kHz cosine in $s(t)$ can be used by the receiver to aid in synchronizing the receiver and transmitter carrier signals.

To summarize, the FM transmitted signal has three components:

- A 15 kHz bandwidth "sum" term $m_s(t)$ occupying the band $|F| < 15$ kHz.
- A 30 kHz bandwidth modulated "difference" term occupying the band 15 kHz on either side of the 38 kHz carrier (23 kHz$<|F|<$53 kHz).
- A pure cosine at 19 kHz, which may be used for synchronization, but should be filtered out of any reconstructed audio signal.

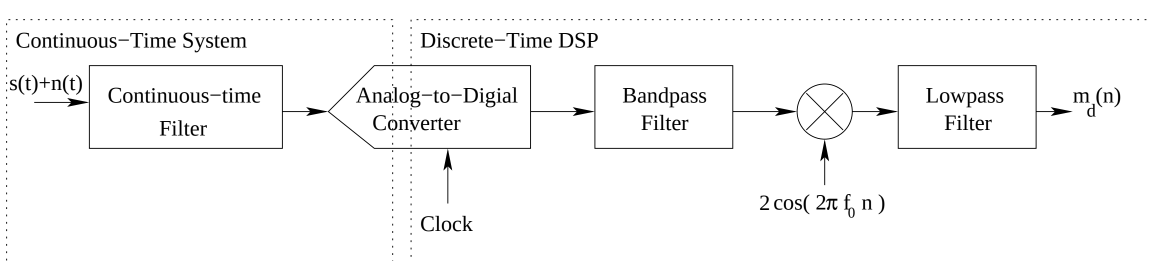
This problem deals with the recovery of the difference signal $m_d(t)$.

A continuous-time demodulator to recover $m_d(t)$ operates by bandpass filtering the received signal to remove undesired signals, multiplying by a cosine at the carrier frequency, and lowpass filtering the result. Multiplying the above difference term by the cosine gives

$$\begin{aligned} x(t) &= m_d(t)\cos^2(2\pi F_c t) \\ &= m_d(t)\left(\frac{1}{2} + \frac{1}{2}\cos(4\pi F_c t)\right) \\ &= \frac{1}{2}m_d(t) + \frac{1}{2}m_d(t)\cos(4\pi F_c t) \end{aligned}$$

The first term of the above equation represents the portion passed by the lowpass filter, and provides the recovered version of $m(t)$.

For this problem, provide the specifications which are needed to implement the above demodulator using a discrete-time processor. the demodulator is to have the structure shown below.



The input to the demodulator is assumed to include noise, which has energy uniformly distributed from DC to 200 kHz. Assume that the analog-to-digital converter operates at a sampling frequency of 120 ksps. Provide design specifications for the above system assuming that frequencies within the band of interest should be preserved by the demodulator ± 0.1 dB. Undesired signals should be attenuated by at least 80 dB. If possible, relax the constraints on the continuous-time filter design in favor of more stringent discrete-time filter specifications. Your solution should indicate the following:

(a) *Continuous-Time Filter:* Is the filter required? If so, specify the passband and stopband edges, passband gain (with tolerances) and minimum stopband attenuation.

(b) *Discrete-Time Bandpass Filter:* Is the filter required? If so, specify the passband and stopband edges, passband gain (with tolerances) and minimum stopband attenuation.

(c) *Mixer Frequency:* Specify the required (discrete-time) frequency f_0.

(d) *Discrete-Time Lowpass Filter:* Is the filter required? If so, specify the passband and stopband edges, passband gain (with tolerances) and minimum stopband attenuation.

4. This problem deals with the calculation of the spectrum of two sinusoidal signals measured in the presence of noise.

   Let $x_1(t)$ be a sinusoidal signal with frequency 17.25 MHz, and let $x_2(t)$ be a sinusoidal signal with the *same amplitude* and frequency 17.27 MHz. The signals are each sampled in the presence of noise using a sampling frequency of 51.2 Msps, giving discrete-time signals $x_1(n)$ and $x_2(n)$. Blocks of 1024 samples are collected, and the FFTs of the two signals are calculated. The resulting magnitudes of the two FFTs are plotted in Figure 1.

   The above experiment was repeated, but this time the collected data was "windowed" prior to calculation of the spectrum. In this case, we set

$$y_1(n) = x_1(n)w(n) \qquad\qquad y_2(n) = x_2(n)w(n)$$

   where $w(n)$ was a 1024-sample Kaiser window function generated using the matlab call "kaiser(1024,7)". The frequencies of the two signals were the same as those shown in Figure 1, but the amplitudes of both signals were changed. The magnitudes of the two FFTs of the windowed signals are plotted in Figure 2.

   (a) Write down (give the equations) and sketch the DTFT of $x_1(n)$ and $x_2(n)$.

   (b) Explain the differences between the two plots shown in Figure 1. Address the difference in the shape of the spectrum, and explain the difference in the displayed amplitudes for the two sinusoidal signals.

   (c) What is the amplitude of the sinusoidal signal contained in $x_1(n)$? Give your answer as a "peak value" (not RMS, and not peak-to-peak).

   (d) Explain the differences between Figures 1 and 2. How does windowing the data cause the observed differences.

   (e) What is the amplitude of the sinusoidal signal contained used for Figure 2? Give your answer as a "peak value" (not RMS, and not peak-to-peak).

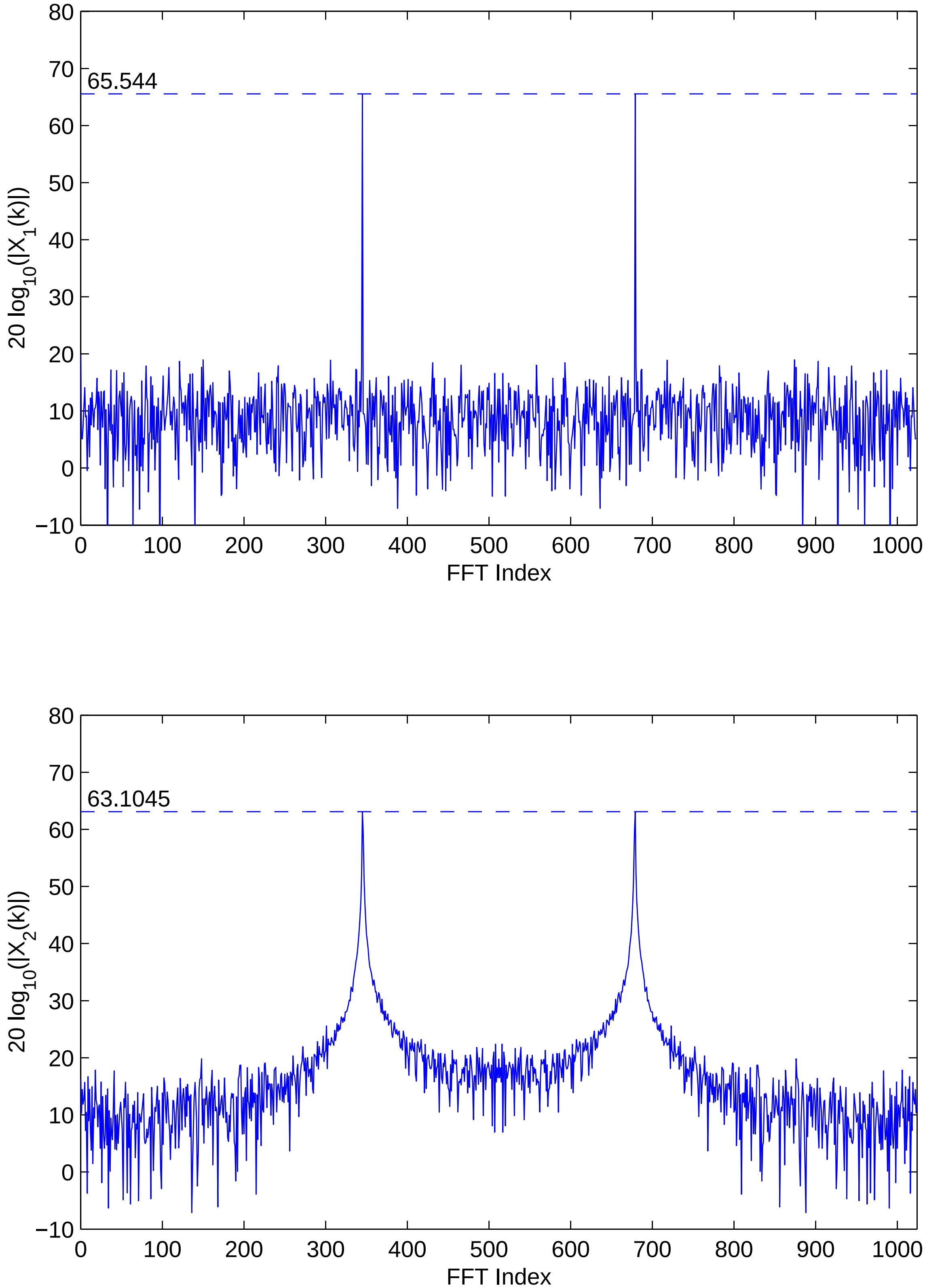


Figure 1: FFT Magnitude spectra plots for Problem 4

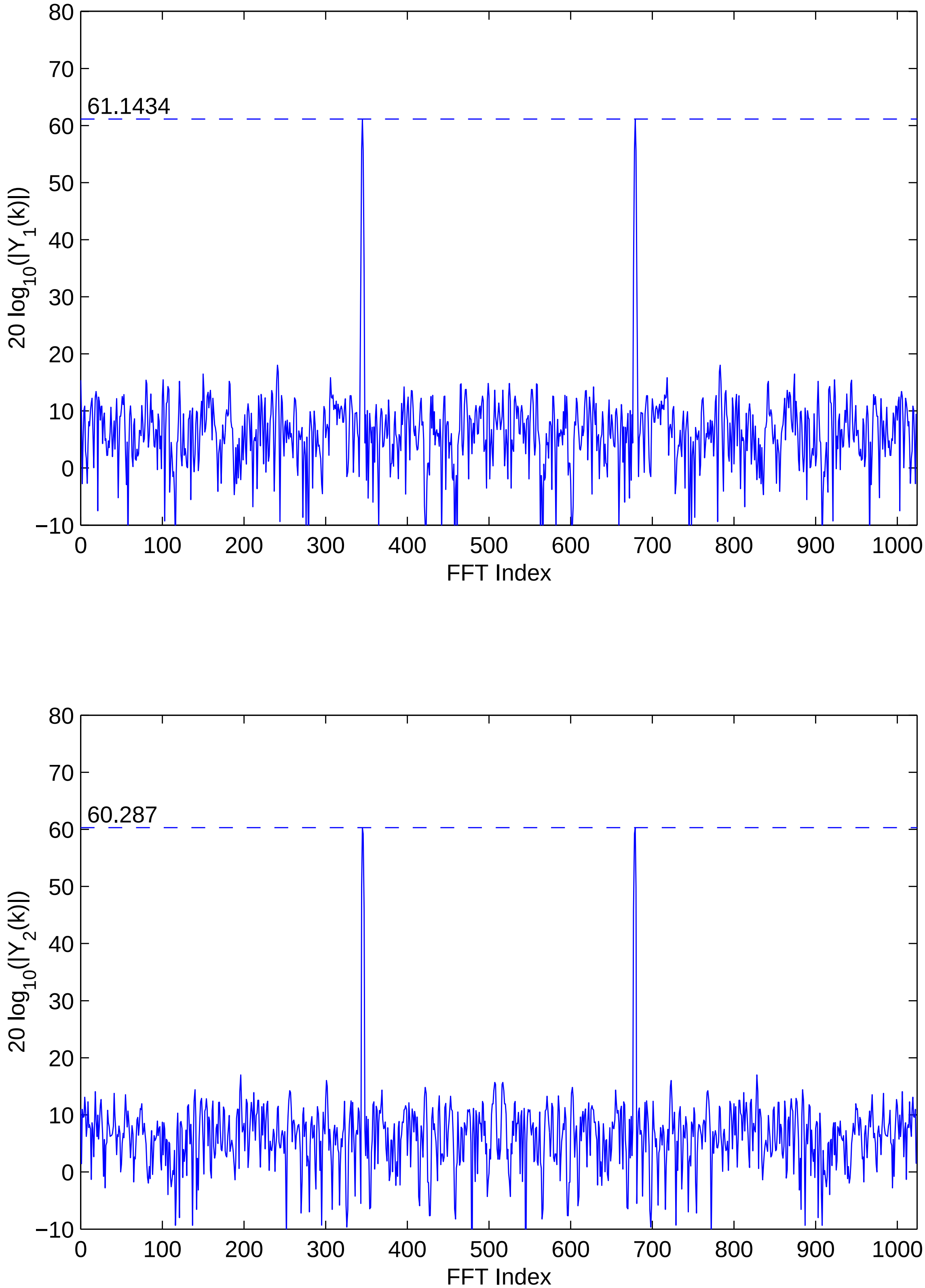


Figure 2: FFT Magnitude spectra plots for Problem 4