ECE-486 24-hour "Extravaganza"
Spring 2001
Due 12:00 (high noon), Tuesday May 8

- 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!
- 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.

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

Passband:	$0 < \|f\| < 0.025$	Gain = 20 dB ± 0.1 dB
Stopband:	$\|f\| > 0.04$	Gain ≤ -60 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. 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. Design a 6th order bandpass filter with passband $0.31 \leq f \leq 0.33$ and passband gain between -0.3 dB and 0 dB. Set the minimum stopband attenuation to 80 dB, and make the transition between passband and stopband as narrow as possible.

   Specify your solution as a cascade of three second-order sections, all with real filter coefficients.

3. Design a linear phase filter to be used in an audio equalizer over the band of frequencies $0 < F < 20$ kHz. The sample rate for the equalizer is $F_s = 44.1$ ksps. The filter should have magnitude response which approximates (within 0.2 dB) the function

$$G(f) = 10\left(\frac{1}{2} - \frac{1}{2}\cos(4\pi f)\right)^8 \ \text{dB}$$

   over the band of frequencies $|f| < 20/44.1 = 0.4535$. Frequencies above $|f| = 21/44.1 = 0.4762$ should be attenuated by at least 40 dB. That is, the designed filter should satisfy

$$G(f) - 0.2 < 20\log_{10}(|H(f)|) < G(f) + 0.2 \qquad |f| < 0.4535$$

   and

$$20\log_{10}(|H(f)|) < -40 \qquad |f| > 0.4762$$

   Turn in a description of your design procedure, and a plot showing your filter response. Submit your design for grading using the MATLAB command

```
ece486_submit('lastname_equalizer',...);
```

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

4. A signal of interest $m(t)$ has been double-sideband (DSB) modulated using a carrier frequency of $F_c = 35$ kHz to give the signal

$$s(t) = Am(t)\cos(2\pi F_c t)$$

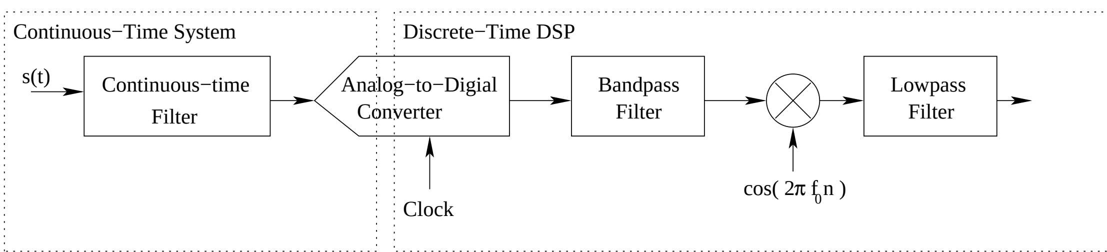
Assume that $m(t)$ has bandwidth 3 kHz, so that $S(F)$ occupies the band of frequencies $32 < F < 38$ kHz. Potentially interfering signals are assumed at frequencies below 31 kHz or above 39 kHz (these signals should be attenuated by at least 60 dB by the demodulator).

A continuous-time DSB demodulator 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 $s(t)$ by the cosine gives

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

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.



Assume that the analog-to-digital converter operates at a sampling frequency of 100 ksps. Provide design specifications for the above system assuming that frequencies within the band of interest should be preserved by the demodulator ± 0.2 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.

5. Specify whether the following statement is true or false. Justify your answer.

   If the bilinear transformation is used to transform a continuous-time all-pass system to a discrete-time system, the resulting discrete-time system will also be an all-pass system.