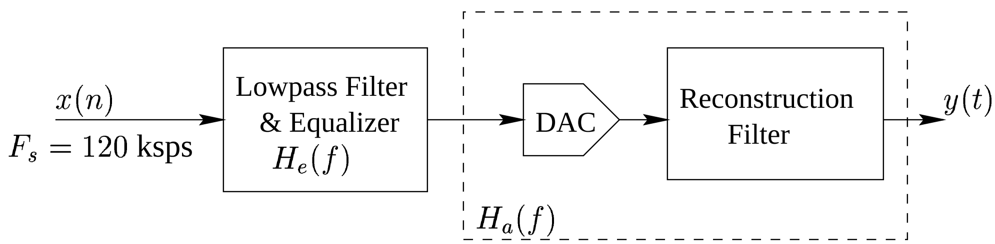
ECE-486 Test 2
Fall 1999
Due 5:00 PM, Dec. 7

- Do your own work. Do not discuss the problems with any other student. Do not talk about the test 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.
- Unless otherwise stated, 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.

1. Design a 6th order lowpass filter with passband $0 \leq f \leq 0.05$ and passband gain between -0.2 dB and 0 dB. Set the minimum stopband attenuation to 60 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.

2. A 17 kHz bandwidth analog signal is to be reconstructed from a discrete-time signal $x(n)$ (sample rate $F_s = 120$ ksps) using the system shown below.



The signal $x(n)$ contains the desired signal as well as undesired signal components at frequencies above 23 kHz. Because of hardware constraints, the DAC and reconstruction filter may be shown to have magnitude response

$$|H_a(F)| = \left(\frac{\sin(\pi F T_s)}{\pi F T_s}\right)\left(\frac{1}{1+(F/20000)^8}\right)^{1/2} \qquad T_s = 1/F_s = 8.33\ \mu\text{s}$$

The lowpass filter passband response is to be modified to "equalize" the linear distortion which the hardware will introduce. Design a linear phase filter $H_e(f)$ such that the composite system ($H_e(f)$ and $H_a(F)$) is flat (± 0.1 dB) in the passband ($0 \leq F \leq 17$ kHz), and so that all signals at frequencies above 23 kHz will be attenuated by at least 70 dB.

3. A signal $x_a(t)$ has been sampled using a sample frequency of $F_x = 120$ ksps. The signal contains a lowpass signal of interest with bandwidth 3 kHz, and a number of undesired signals throughout the 0-60 kHz band.

   A "downsampler" is to be designed which reduces the sample frequency by a factor of 16, to $F_y = 120/16 = 7.5$ ksps. This problem involves attempting the design using several structures, and evaluating each of the approaches. Each of the structures must leave the 0-3 kHz band unchanged (± 0.1 dB), and must attenuate any terms which lie above the new Nyquist frequency of 3.75 kHz by at least 80 dB.

   (a) First, design the downsampler using a single decimate-by-16 stage. Use an FIR filter in your implementation.

   (b) Second, design the downsampler by following a decimate-by-8 stage with a decimate-by-2 stage. Again, use FIR filters for your implementation

   (c) Third, design the downsampler by implementing an IIR filter as the anti-aliasing filter running at the original 120 ksps sample rate. No FIR filter is then needed for the decimator.

   In each case, show how you designed the required filters (including how you selected filter design parameters), give the filter orders that are required for each filter, and provide magnitude frequency-response plots for all required filters. Evaluate the computational complexity of each of the above designs by specifying the number of multiply/accumulate operations required *per (decimated) output sample*. Which of the above approaches is most efficient?

4. 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.