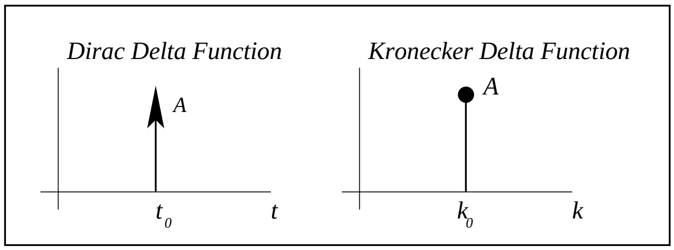
ECE486 Test 2, In Class Portion

April 20, 2006
Closed Book, Closed Notes, No Calculators or Computers

When you are finished with this portion of the exam, turn it in and go to work on the computer portion. After you turn this exam in, you may not return to this part of the test.

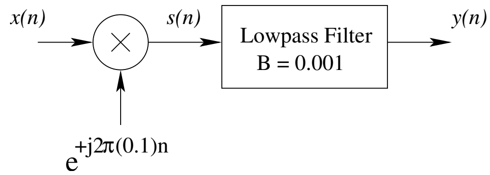
In all of your plots, please be sure to distinguish between the Dirac delta function (shown on the left, with zero width, infinite height, and area labeled next to the arrow head), and the Kronecker delta function (shown on the right, with finite height labeled next to the dot).



All frequency-domain plots should clearly label the frequencies and amplitudes of major peaks.

Opportunities (not problems!) begin on the next page...

1. For the system shown below, assume that $x(n)$ is obtained by sampling a continuous-time signal $x_c(t)$ using a sample frequency of 100 ksps. The system is designed to examine the band of frequencies $9.9 < |F| < 10.1$ kHz.

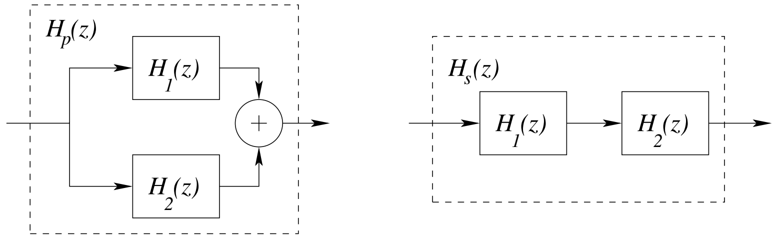


Assume that the input signal $x_c(t)$ is a cosine with (peak) amplitude A and frequency 10.001 kHz.

(a) Draw and carefully label the DTFT of $x(n)$, $s(n)$, and $y(n)$.

(b) Describe the (time-domain) behavior of the output signal $y(n)$. Give as many details as you can.

2. Let $H_1(z)$ and $H_2(z)$ denote two linear phase FIR filters with symmetric impulse responses $h_1(n)$ and $h_2(n)$. Define $H_p(z)$ and $H_s(z)$ as the transfer functions associated with the parallel and series combinations of these filters.



(a) Express the impulse responses $h_p(n)$ and $h_s(n)$ in terms of $h_1(n)$ and $h_2(n)$.

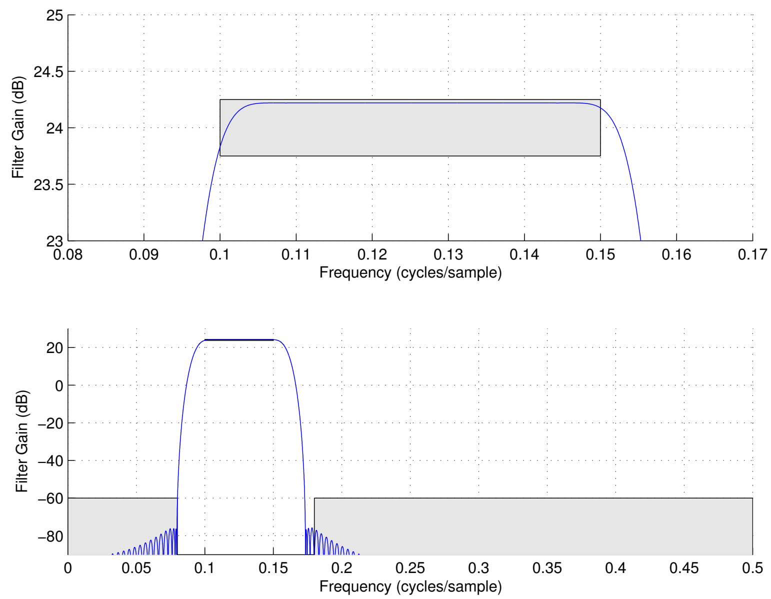
(b) Give the transfer functions $H_p(z)$ and $H_s(z)$ in terms of $H_1(z)$ and $H_2(z)$

(c) Is $H_p(z)$ necessarily a linear-phase filter? If so, prove it. If not, give an example of two linear-phase filters $H_1(z)$ and $H_2(z)$ for which the parallel combination is *not* linear phase.

(d) Is $H_s(z)$ necessarily a linear-phase filter? If so, prove it. If not, give an example of two linear-phase filters $H_1(z)$ and $H_2(z)$ for which the series combination is *not* linear phase.

3. The plot below shows the magnitude response of a linear phase FIR filter which has been designed using the "window method" to meet the following specification:

Passband:	$0.1 < \|f\| < 0.15$
Passband Gain:	24 ± 0.25 dB
Stopbands:	$\|f\| < 0.08$ and $0.18 < \|f\| < 0.5$
Stopband Attenuation:	≥ 60 dB



The specification was met using 240 filter coefficients utilizing a Kaiser window with parameter $\beta = 10$.

Suggest how the design might be modified to achieve a lower order filter (still using the window design method).

4. The transfer function of a continuous-time, first order "all-pass" filter is given by

$$H_c(s) = \frac{s-a}{s+a}$$

where a is real and $a \geq 0$. One can verify that $|H_c(j\Omega)| = 1$ for all Ω. The filter provides a frequency dependent phase shift, taking a value of $\pi/2$ radians at $\Omega = a$ rad/sec.

Use the bilinear transform to design a first order discrete-time all-pass system. Design your system to provide a phase shift of $\pi/2$ radians at $f = 0.1$ cycles/sample. The system is to use a sample frequency of 50 ksps. Describe your system using a difference equation, and show how to find all required filter coefficients.

ECE486 Test 2, Computer Portion

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Open Book, Open Notes
Calculators and Computers allowed

You may use your own printed or written reference material. Communication with other people is not allowed. Web browsers should be closed. Use of chat rooms, bulletin boards, firstclass etc. is not allowed.

Results submitted after the end of the test will not be accepted or graded.

Do not send output to the printer. Printed output will not be accepted or graded.

All submitted filter designs should have real coefficients. All FIR filter designs should be linear phase filters. In all filter designs, try to minimize the number of filter coefficients.

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1. Design an equiripple, linear phase lowpass filter which meets the following specifications:

Passband:	$0 < \|f\| < 0.04$
Passband Gain:	18.1 dB $\pm$ 0.3 dB
Stopband:	$0.05 < \|f\| < 0.5$
Stopband Attenuation:	$\geq$ 60 dB

Submit your design for grading using:

```
ece486_submit('lastname_test2_1',h);
```

In the space below, give the filter order and the number of filter coefficients. Indicate the delay associated with the filter.

2. Find an IIR filter design which meets the same specifications as given in problem 1. Submit your design for grading using

```
ece486_submit('lastname_test2_2',b,a);
```

In the space below, give the filter order and the number of filter coefficients. Indicate the delay associated with the filter.

3. A discrete-time system uses sampling frequency $F_s = 48$ ksps. Design a linear phase FIR 90°phase-shift filter which operates over the band of frequencies 3 kHz $< |F| <$ 5 kHz. Over this band, the designed filter should approximate the transfer function $10j\,\mathrm{sign}(F)$ (with an appropriate constant delay phase shift). Design your filter so that the delay associated with the filter is an integer number of samples.

   Your filter should maintain a gain of 20 ± 0.2 dB over the passband. In addition, the FIR filter is to attenuate frequencies below 1 kHz and above 8 kHz by at least 50 dB.

   Submit your design for grading using the MATLAB command

```
ece486_submit('lastname_test2_3',h);
```

   where "h" is the name of the vector containing your filter coefficients.

   In the space below, give the filter order and the number of filter coefficients. Indicate the delay associated with the filter.

4. In MATLAB, type the command:

```
load N:\test2data.mat
```

   This should load a vector (named "`test2data`") which contains 1024 samples of a signal which has been captured using a sampling frequency of 40 Msps. You may assume that the samples are in units of volts, and that the original signal was band-limited to 20 MHz, so that aliasing is not an issue. The (noisy) signal contains two sinusoids at closely spaced frequencies (and dramatically different amplitudes).

   Determine (as accurately as possible) the amplitude and frequency of both of the sinusoids. Give your results, and describe how you determined your answers in the space below.