Name (Last, First) ____________________________ Student ID number _________________

UNIVERSITY OF CALIFORNIA

College of Engineering
Electrical Engineering and Computer Sciences Department

EECS 145M: Microcomputer Interfacing Laboratory

Spring Midterm #2 (Closed book- equation sheet provided- calculators OK)
Full credit can only be given if you show your work.
Wednesday, April 22, 2009

PROBLEM 1 (30 points)

1.1 (6 points) State the Fourier convolution theorem

1.2 (6 points) Use the Fourier convolution theorem to show that the Integral Fourier Transform of a periodic waveform contains only discrete frequencies.

1.3 (6 points) State the Nyquist sampling theorem

1.4 (6 points) State the Fourier frequency convolution theorem

1.5 (6 points) Use the Fourier frequency convolution theorem to explain the Nyquist sampling theorem in the frequency domain

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PROBLEM 2 (total 25 points):

You have a data acquisition system that has a sampling rate of 2^{16} = 65,536 Hz and an 8th order Butterworth anti-aliasing filter with a corner frequency of 20 kHz. A Hann window is used so that arbitrary waveforms can be sampled and analyzed.

2.1 (5 points) Over what frequency range is the Butterworth filter gain greater than 0.99?

2.2 (5 points) Over what frequency range is the Butterworth filter gain less than 0.001?

2.3 (10 points) If the input consists of two 1 V peak-to-peak sinewaves with frequencies of 20 kHz and 40 kHz, how would these appear in an FFT of 2^{16} = 65,536 sampled values?

2.4 (5 points) How would the answer to the previous part change if the Hann window was not used?

PROBLEM 3 (total 45 points):

You want to design a digital filter to ***continuously*** compensate for the limited frequency response of a loudspeaker (like Lab 24a). Before you can do this, you need to measure the frequency response of the loudspeaker.

Assume:

- You have a computer with analog I/O.
- The analog I/O has a range from –5V to +5V.
- You will excite the speaker with a pseudo-random waveform that you have stored as a floating point vector of 16,384 values that range from –5V to +5V.
- You have a microphone that has a uniform response over the 20 to 20 kHz frequency range of interest and an output from –5mV to +5mV.

3.1 (10 points) Sketch a block diagram of your system. Include the computer, the I/O ports, the loudspeaker, the microphone, and any necessary amplification.

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You send the pseudo-random waveform to the loudspeaker ***once*** at a frequency of 65,536 Hz and simultaneously sample the microphone waveform at the same frequency. You then take the FFT of the microphone samples.

3.2 (5 points) To what frequencies do the Fourier amplitudes H_1 and H_{1000} correspond?

3.3 (10 points) How would you use the pseudo-random waveform and the loudspeaker response to calculate the digital filter that compensates for the limited frequency response of the loud speaker?

3.4 (10 points) How would you check that the digital filter worked as planned?

3.5 (10 points) You notice that the microphone signal has random electronic noise and that this degrades the measurement of the frequency response of the loudspeaker. How would you use 100 cycles of the pseudo random waveform through the system to compute the frequency response with minimum noise?