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)
Monday, April 16, 2001

PROBLEM 1 (20 points)

1a (10 points) Sketch a block diagram of the main components of the **successive approximation** analog to digital converter.

1b (10 points) Describe in words or a list of steps the operation of the **successive approximation** analog to digital converter.

PROBLEM 2 (20 points)

2a (10 points) Sketch a block diagram of the main components of the **flash** analog to digital converter.

2b (10 points) Describe in words or a list of steps the operation of the **flash** analog to digital converter.

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Problem 3 (20 points) The integral Fourier transform of a single square pulse of width T_0 is H(f) = sin($\pi T_0 f$)/($\pi T_0 f$). (See equation sheet for a table of values.)

3a (10 points) Using the Fourier convolution theorem, write the equation for the integral Fourier transform of an infinite periodic series of square pulses of width T_0 and period T_r.

3a (10 points) Sketch the integral Fourier transform $H(f)$ of an infinite series of square waves of 1 μs width and 1 ms period from f = 0 Hz to 1 MHz.

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Problem 4 (total 40 points) You have been assigned the task of determining the gain $|V_{out}/V_{in}|$ of a "black box" circuit from 0 Hz to 100 kHz in 100 Hz steps with 1% accuracy. The circuit has an input V_{in} and an output V_{out} but you do not know what is in the black box.

- Rather than measuring the response of the black box at 1000 separate frequencies, you decide to input a periodic series of 1 μs pulses. As seen in problem 3, **this allows you to input all frequencies of interest simultaneously**.
- You then sample the output and perform the FFT to determine the frequency content of the output
- Since the periodic pulses also contain frequencies well above the frequencies of interest, you decide to use an 8-pole Butterworth filter with gain >0.999 for frequencies below 100 kHz and gain < 0.01 for frequencies that could alias below 100 kHz. (Hint: use the equation sheet filter gain table with $n = 8$)
- Assume that you have a circuit that can produce 1 μs-wide pulses with a period T_r of your choosing
- Assume that you have a computer with a data acquisition circuit that can take M samples of a waveform at a sampling frequency f_s of your choosing.

4a. (10 points) Draw a block diagram of your system, showing all essential elements.

4b. (10 points) Determine (1) the corner frequency f_c of the Butterworth filter and (2) the sampling frequency f_s of the data acquisition circuit that meet the anti-aliasing requirements.

4c. (10 points) If you minimize the number of samples *M* (where *M* is power of 2), what is your sampling frequency f_s, pulse repetition rate T_r and the value of *M*?

4d. (10 points) To what frequency f_n does the Fourier coefficient H_n correspond, and how would you compute the black box circuit gain $|V_{out}/V_{in}|$ at that frequency?