NAME (please print) ____

STUDENT (SID) NUMBER ___________________________

UNIVERSITY OF CALIFORNIA, BERKELEY

College of Engineering
Electrical Engineering and Computer Sciences

EECS 145M: Microcomputer Interfacing Lab

LAB REPORTS:

1 ________________	2 ________________	3 ________________
8 ________________	9 ________________	10 ________________
21 ________________	22 ________________	23 ________________
24a ________________	24b ________________	

Total of top 4 Long Lab Grades	______________	(400 max)
Total of top 4 Short Lab Grades	______________	(100 max)
Lab Participation	______________	(100 max)
Mid-Term #1	______________	(100 max)
Mid-Term #2	______________	(100 max)
Final Exam	______________	(200 max)
Total Course Grade	______________	(1000 max)

COURSE LETTER GRADE

Spring 2006 FINAL EXAM (May 13)

Answer the questions on the following pages completely, but as concisely as possible. The exam is to be taken ***closed book***. Use the reverse side of the exam sheets if you need more space. Calculators are OK.

Partial credit can only be given if you show your work.

FINAL EXAM GRADE :

1 __________ (40 max) 2 __________ (20 max) 3 __________ (45 max)

4 __________ (50 max) 5 __________ (45 max)

TOTAL __________ (200 max)

Initials ____

PROBLEM 1 (total 40 points) Briefly define the following terms:

1.1 (10 points) Tri-State Buffer

1.2 (10 points) Fourier Convolution Theorem

1.3 (10 points) Transition Voltages (of the A/D converter)

1.4 (10 points) Infinite Impulse Response Digital Filter

PROBLEM 2 (20 points)
You take two sets of measurements, find that their average values are *a* and *b*, and that the standard deviations of the averages are $\sigma_a = 0.01a$ and $\sigma_b = 0.01b$. If $r = a/b$, derive the standard deviation of r (σ_r) as a function of *r*.

Initials ____

PROBLEM 3 (total 45 points)

3.1 (10 points) Using the Fourier Convolution Theorem, explain why a periodic waveform contains only discrete frequencies. You may want to use a diagram to aid the explanation.

3.2 (15 points) Using the Fourier Frequency Convolution Theorem, explain how aliasing occurs. When designing a system for sampling arbitrary waveforms, describe how you can reduce aliasing to an acceptable level. You may want to use a diagram to aid the explanation.

3.3 (10 points) When sampling an arbitrary waveform, explain how multiplying the data by a windowing function reduces the appearance of unwanted frequencies (spectral leakage) in the sampled values. You may want to use a diagram to aid the explanation.

3.4 (10 points) Assuming that you do not use a windowing function, use the Fourier Frequency Convolution Theorem to relate the frequency content of the sampled values with the frequency content of the original waveform. You may want to use a diagram to aid the explanation.

PROBLEM 4 (total 50 points)

Design an interface that allows a single computer with a single digital I/O board to read data from 256 external sensor circuits once per second. These circuits continuously measure physical quantities such as time, temperature, pressure, speed, etc.

Assume that

- The I/O board has data lines for 32 bits, and these can be set individually for input or output
- The sensors produce 16-bit digital outputs that update frequently and at unpredictable times. They do not have handshaking lines.
- When a sensor circuit output changes, it takes a maximum of 20 ns for the bits to become stable.
- One of the external circuits counts the elapsed number of milliseconds and is set to zero at the end of every minute (it would overflow at 65.5 seconds); another external circuit counts minutes

4.1 (20 points) Sketch your design, showing and labeling all essential components and lines.

4.2 (10 points) Describe the events that take place in your circuit when one of the sensor circuit output changes.

4.3 (20 points) Describe the program steps that the computer must use to input the 256 sensor values and store them along with the approximate time (within a few ms) at which they were taken.

PROBLEM 5 (total 45 points):

Design a system for determining the frequency content of the notes produced by a musical instrument, assuming the following:

- The instrument is a bowed string or wind instrument or the human voice that can play a sustained note.
- When the instrument plays a sustained note, the sound (waveform) contains harmonics with constant frequencies and amplitudes
- The frequency of the first harmonic is the frequency of the note being played.

5.1 (15 points) Describe how for any sustained note you would sample the waveform in the frequency range from 20 to 20,000 Hz to an accuracy of 1%, while reducing the amplitude of any aliased frequencies by a factor of 1000.

5.2 (10 points) Describe how you would use the waveform data from part 5.1 to determine the frequencies and relative amplitudes of the harmonic components (up to the 100^{th} harmonic) to an accuracy of 1% in frequency and 1% in amplitude.

5.3 (5 points) For a single note, how many numbers need to be stored in part 5.1 to be able to reproduce the waveform at a later time?

5.4 (5 points) For a single note, how many numbers need to be stored in part 5.2 to reproduce the waveform at a later time?

5.5 (10 points) Design a computer system that uses part 5.2 to allow a musician to play notes that sound like those from the original instrument.