NAME (please print) ____

STUDENT (SID) NUMBER _________________________

UNIVERSITY OF CALIFORNIA

College of Engineering
Electrical Engineering and Computer Sciences
Berkeley

EECS 145M: Microcomputer Interfacing Lab

LAB REPORTS:

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

Total of top 4 Lab Grades	______________	(400 max)
Total of top 4 Question Sections	______________	(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 2001 FINAL EXAM (May 18)

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. **In answering the problems, you are not limited to the particular equipment you used in the laboratory exercises.**

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

FINAL EXAM GRADE :

1 __________ (30 max)   2 __________ (70 max)

3 __________ (50 max)   4 __________ (50 max)

TOTAL __________ (200 max)

Initials ____________________

Problem 1 (total 30 points):

Briefly describe the essential differences between the following pairs of terms:

1a. (10 points) Transparent latch vs. sample and hold amplifier

1b. (10 points) Successive Approximation A/D converter vs. Flash A/D converter

1c. (10 points) Frequency aliasing vs. spectral leakage.

Initials ________________________

Problem 2. (70 points) Design a **computer-controlled** system for the assembly line testing of **eight** units of a new type of 12-bit A/D converter.

You are provided with the following:

- eight sample A/D converters (to be tested eight at a time)
- eight 16-bit tri-state drivers
- a microcomputer with the following:
  - a 16-bit D/A converter with 1/2 LSB absolute accuracy and 10 μs settling time
  - two 16-bit parallel input ports
  - two 16-bit parallel output ports

You may assume the following:

- The 16-bit parallel output port is in “transparent” mode (no handshaking). New data can be written to the port every 2 μs.
- You have a timer function wait(*N*), that can delay program execution for *N* μs.
- The A/D converter requires a “start conversion” low-to-high edge signal and after conversion provides an “output data available” low-to-high edge. The A/D converter sets “output data available” low and resets all internal functions when “start conversion” goes low.
- For highest possible reliability, you must wait until the A/D has signaled that its data are ready before reading its output.

Hint: Think about Laboratory Exercise 9 (A/D converters) and how you would automate the measurement and data analysis procedures.

2a. (20 points) Draw a block diagram of the major components, including two of the eight A/D converters being tested. Show and label all essential data and control lines.

Initials ______________________

2b. (10 points) List the steps your program must do to measure the first transition voltage V(0,1) of the first A/D converter (pseudocode is OK, so long as the logic is clear).

2c. (10 points) How would you determine the maximum absolute accuracy error of the A/D? (Explain the procedure in steps or with a flow diagram.)

2d. (10 points) How would you determine the maximum linearity error?

2e. (10 points) How would you determine the maximum differential linearity error?

2f. (10 points) With what accuracy could this system measure the quantities in parts **b.**, **c.**, and **d.** in units of 1 LSB of the A/D?

Initials ______________________

PROBLEM 3 (50 points)

You have designed and built a computer system to sample waveforms and perform the FFT.
It has the following characteristics:

- Sampling frequency = 2^{18} Hz = 262,144 Hz
- Number of samples = 2^{16} = 65,536
- Low-pass Butterworth anti-aliasing filter of order 8 and corner frequency f_c = 100,000 Hz
- Hanning (raised cosine) window

Answer the following questions:

3.a (3 points) For what frequency range does the anti-aliasing filter have gain >0.99?
(*Hint:* Use the Butterworth gain table on the equation sheet)

3.b (3 points) For what frequency range does the anti-aliasing filter have gain <0.01?

3.c (2 points) How long does it take to acquire the samples?

3.d (2 points) To what frequencies do the FFT coefficients H_0 and H_1 correspond?

3.e (4 points) What is the FFT coefficient with the highest index and to what frequency does it correspond?

3.f (4 points) What is the FFT coefficient that corresponds to the highest frequency and what is that frequency?

3.g (6 points) You sample a 4,000 Hz ***sinewave*** with the system and take the FFT. What FFT coefficients should be non-zero?

3.h (6 points) You sample a 4,000 Hz symmetric ***square*** wave with the system and take the FFT. What FFT coefficients should be non-zero? (symmetric means 50% high, 50% low)

3.i (6 points) You sample a 4,002 Hz ***sinewave*** with the system and take the FFT. What FFT coefficients should be non-zero?

3.j (6 points) You sample two sinewave signals, one with a frequency of 4,000 Hz and another at a nearby frequency and 10 times smaller in magnitude. How closely can the frequency of the second signal approach 4,000 Hz and still be resolved in the FFT coefficients as a separate peak?

3.k (2 points) How would you change the system to reduce the answer to the previous question 3.j by a factor of two?

3.l (6 points) You sample a sinewave of frequency 2^{18} – 84,000 Hz = 178,144 Hz and take the FFT. What FFT coefficients should be non-zero? How does the magnitude of the largest FFT coefficient compare with that you would get if you sampled an 84,000 Hz sinewave?

Initials ___________________________

Problem 4 (total 50 points):

You have the following:

- a computer with an analog input/output port that can operate at a chosen rate up to 10^6 samples/s
- a loudspeaker that can convert an electrical waveform into an acoustic waveform with a frequency response that is significantly above zero for all frequencies below 25 kHz and essentially zero for all frequencies above 50 kHz.
- a microphone that can accurately convert an acoustic waveform into an electrical waveform for all frequencies below 50 kHz.

4a (20 points) Describe how you would measure the acoustic response of the loudspeaker to a delta function electrical input. **Sketch** your setup and **list** the steps you would use to acquire a digitized version of the impulse response (sampled at 100 kHz) and take its discrete Fourier transform.

4b (10 points) Using the Fourier convolution theorem in the continuous time and frequency domains, what acoustic output waveform d(t) would result from using an arbitrary waveform a(t) as the input to a loudspeaker whose impulse response is c(t)?

4c (10.points) In the continuous time and frequency domains, determine a function b(t) so that if the convolution of a(t) and b(t) is used as the input to a loudspeaker with impulse response c(t), the loudspeaker will produce an acoustic waveform that is close to a(t)

4d (10.points) How would you apply the result of part 4c to a digitized version of a(t) (sampled at 100 kHz) to correct for the response of the loudspeaker and produce an acoustic waveform that is close to the digitized version of a(t).