EECS 145M: Microcomputer Interfacing Lab
Spring 1993 Final Exam (May 20)

Problem 1 (total 36 points):

Define the following terms (30 words or less)

a. (6 points) R-2R Ladder (of a D/A converter)

b. (6 points) Average Step Size (of an A/D converter)

c. (6 points) Relative Accuracy (of an A/D converter)

d. (6 points) Handshaking (between a microcomputer and a digital circuit)

e. (6 points) Two's Complement Operation

f. (6 points) Tri-state buffer

Problem 2 (35 points)

Describe how you would measure the frequency of the 60 Hz, 120 V power line to an accuracy of 0.001 Hz over a measurement period of 1 second. Assume that the frequency is in the 59 to 61 Hz range and that you have the following components.

- A microcomputer with a 9513 timer/counter circuit (like the one used in laboratory exercise 2) and an 8-bit parallel I/O port (like the one used in laboratory exercise 3).
- A comparator circuit.
- Additional components as needed, but keep it simple.

Do the following:

a. (12 points) Draw a block diagram of your measurement system, showing and labeling all essential components and signal lines.

b. (12 points) Describe the operation of your system (hardware and software) in a flow chart or pseudo-code.

c. (11 points) Draw a timing diagram for the important signals.

Problem 3 (44 points)

Design a microcomputer-based system for two-track (stereo) recording of music at a sampling rate of 50 kHz. Both tracks are to be sampled simultaneously, but the digital data must be read by the computer sequentially **because you have only one parallel input port**. You have the following components at your disposal:

- A microcomputer able to transfer data to and from its parallel I/O ports in 2 microseconds.
- A single 16-bit parallel input port. A BISTROBE line can be asserted by an external circuit and read by the program as a signal that new data are ready to be read. A low-to-high transition on an external

BIHOLD line latches data onto the 16 internal registers. (Similar to the port used in the 145M lab.)
- A 16-bit parallel output port, configured for transparent operation.
- A single timer that can be configured by the microcomputer to produce pulses on an external line that are uniformly spaced in time. The pulse sense (high or low), width, and interval can be set by the computer program.
- Two sample-and-hold amplifiers. High = hold, low = sample.
- Two 16-bit successive approximation A/D converters with a conversion speed of 15 microseconds. Conversion is initiated with a low-to-high transition. "End of conversion" is signaled by a low-to-high transition.
- Two 16-bit registers with tri-state output buffers. External data are latched onto internal registers with a low-to-high edge on a strobe line. When an additional "select" line is high, the 16 outputs agree with the 16 inputs. When the "select" line is low, the 16 outputs are in a high impedance state.
- One-shots and delays as needed.
- Additional components as needed, but keep it simple.

a. (20 points) Draw a block diagram of your recording system, showing and labeling all essential components and signal lines.

b. (12 points) Describe the operation of your system in flowchart of pseudo code.

c. (12 points) Show a timing diagram for one complete sampling cycle.

Problem 4 (total 55 points):

In concert halls and outdoor arenas, the sound that the that the audience hears is not necessarily the sound produced by the performers. The reason is that hte limited rseponse of the loud speakers, absorption by surrounding surfaces, resonances within enclosing volumes, and feedback effects can greatly alter the frequency spectrum. One spectacular feedback effect is the loud whine that occurs at particular frequency when the gain loop (from the recording microphone to the amplifier to the speakers and back through the air to the recording microphone) is greater than 1. At high volumes, it would be desirable to attenuate the signal at such resonant frequencies.

Design a system for frequency filtering in a sound system so that the sound that reaches a sensing microphone in the audience has the same frequency content as the sound that reaches a recording microphone on the stage. Do not worry about phase, seperate stereo or quadriphonic channels, or trying to correct the sound at any other place in the audience.

Assume the following:

- The recording microphone and the power amplifiers that drive the loudspeakers have a flat response from 20 Hz to 20 kHz.
- The sampling microphone placed in the audience records all frequencies between 20 Hz and 20 kHz equally.

- You plan to use an 8-pole Butterworth filter in your sampling microphone circuit with a corner frequency f_c = 25 kHz. The frequency response of the n-pole Butterworth filter is given by: $G(f) = 1/(1 + (f/f_c)^{2n})^{1/2}$

Part 1 - determining the acoustical response of the hall:

- You decide to determine the acoustical response of the hall by sending a series of sharp pulses into the power amplifiers witha repetition period of 20 Hz and recording the response with the sensing microphone.
- The pulses are so narrow that the non-zero values of their frequency spectrum are equal from 20 Hz to 20 kHz.
- You avoid windowing effects by sampling the output of the recording microphone in the audience for exactly 10 repetition periods of the 20 Hz pulses at a sampling frequency of 100 kHz.
- You then take the FFT to determine the frequency response of the hall.

a. (5 points) How many samples will you be taking?

b. (5 points) To what frequency does the F1 Fourier coefficient correspond?

c. (5 points) What Fourier coefficient corresponds to 20 Hz?

d. (10 points) Which Fourier coefficients are non-zero?

e. (5 points) What is the gain of the Butterworth filter at $25/(2)^{1/2}$ kHz = 17.67 kHz?
(Hint: $1/(1 + e) \sim= 1 - e/2$ for small e, where e is the greek symbol epsilon)

f. (5 points) What is the gain of the Butterworth filter at 50 kHz?

Problem 5 (total 30 points):

Design a system for converting digital audio data from compact digital disk (sampled at 44 kHz) to digital audiotape (to be played at 48 kHz). Simply reading the 16-bit digital words from the compact disk and writing them to digital tape will not work because on playback all frequencies will be shifted upward by about 10%.

Note that there is no single correct answer - but note the following:

- The waveform produced by the D/A converter in the digital audiotape player must be an accurate replica (bothe in phase and frequency) of the waveform originally sampled to produce the digital disk.
- The conversion process can be performed either in batch mode or continuous mode.

a. (15 points) Sketch the system. Show and label all essential components and signal lines.

b. (15 points) Describe how the system works.
