UNIVERSITY OF CALIFORNIA
College of Engineering
Electrical Engineering and Computer Sciences Department

145M Microcomputer Interfacing Lab

Final Exam Solutions May 18, 2001

1a The **transparent latch** input and output are digital while the **sample and hold amplifier** input and output are analog. Both have similar control lines whose state determines whether the output is equal to the input or held at its last value.

[1 point off for describing both but not stating what is different]

[4 points off for describing both without mentioning analog or digital]

[4 points off for stating that the transparent latch is always transparent but the S/H can be either sample or hold]

1b

successive approximation	**flash**
# steps = # bits (N)	continuous conversion (1 step)
requires S/H for steady input	does not require S/H
larger number of bits (12-16 typical)	smaller number of bits (8-10 typical)
uses a D/A	uses $2^N - 1$ comparators

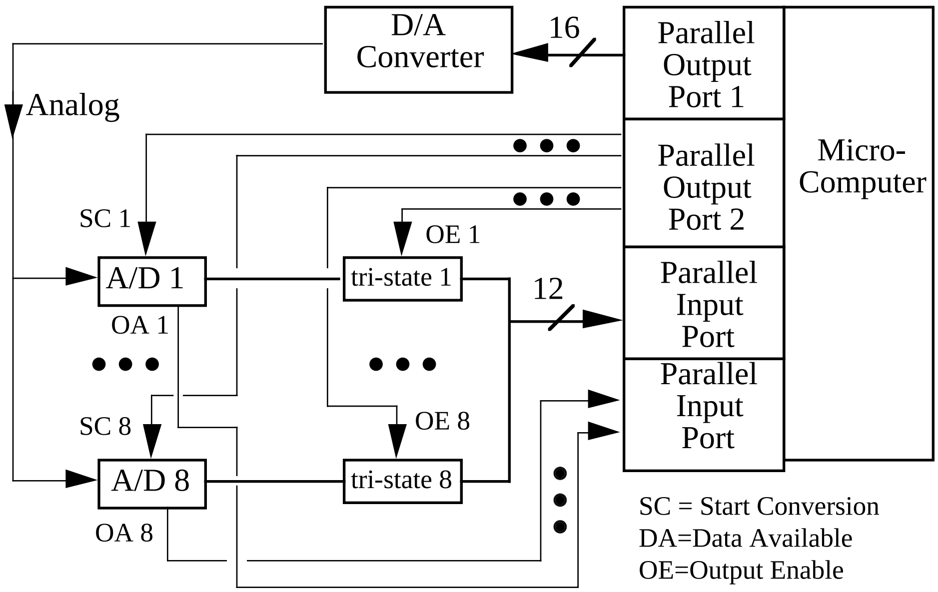
[5 points off for # steps missing] [no points off for missing one other item] [2 points off for each additional missing item]

1c **Frequency aliasing** is caused by insufficient sampling and causes frequencies above one-half the sampling frequency to appear as lower frequencies. **Spectral leakage** is caused by sampling a non-integer number of cycles, which produces a discontinuity at the edges of the sampling window, which adds erroneous contributions to many Fourier coefficients. (More precisely, the observed Fourier transform is the true Fourier transform convolved with the Fourier transform of the sampling window)

2a The following are essential [3 points off for each omitted]:

- Connect 16 bits of the parallel output port to the input of the 16 bit D/A converter
- Connect the analog output of the D/A to the analog inputs of all A/Ds
- Connect the 12 bit A/Ds to separate tri-state drivers
- Connect the outputs of the tri-state drivers together to form a data bus
- Connect the data bus to 12 bits of the parallel input port
- Provide 8 separate output port lines for initiating conversion of the 8 A/Ds
- Provide 8 separate output port lines for enabling the 8 separate tri-state drivers
- Provide input for 8 separate input port lines to indicate when individual A/Ds have completed conversion

(OK to connect the start conversion and output enable signals for each A/D signal path, provided the signal timing was properly designed)



2b The steps needed to measure the first transition voltage V(0,1) for the first A/D converter.

1 set SC1 low, disable all tri-state drivers, set N = 0

2 Put N on D/A

3 wait 10µs until D/A has settled [using wait(10)]

4 Put low-high edge on SC1 output line to start conversion

5 Wait until output data available

6 enable tri-state 1 (disable all others)

7 read input port to get value M

8 Put high-low edge on SC1 output port to end conversion cycle

9 if M=0, increase N by one and loop back to step 2

10 If M=1, save (D/A voltage step)(N-1/2) as the transition voltage

2c Send successive 16-bit numbers 0 to 2^{16}–1 to the D/A converter and convert the analog output with the A/D. Whenever the A/D output value changes, store the corresponding D/A value in a transition voltage table

To determine **absolute accuracy**, compare the D/A values for each A/D output transition with the ideal transition values. Since the step size of the D/A is 16 times finer than the average step size of the A/D, this design can measure the A/D transition voltages to an accuracy of 1/16 LSB.

[3 points off if only 12 D/A bits are varied. The transition voltages (or the center of the steps) cannot be determined accurately unless more than 12 bits of the accuracy of the 16-bit D/A is used.]

2d Determine the D/A values corresponding to first and last A/D transition voltages, and the equation of the line that passes through them. **Linearity** is a measure of how closely the other measured transition values pass through the line.

2e Determine the difference in D/A values between each A/D transition voltage. The **differential linearity** is a measure of the equality of those differences. Alternatively, the A/D step sizes could be determined as the number of successive D/A inputs that produce the same A/D output.

2f The method can determine the A/D accuracies to 1/16 LSB (±1/32 LSB was OK).

Note that 1 A/D LSB = 16 D/A LSBs.

[5 points off for an answer of 1/2 LSB]

3a Filter gain >0.99 for frequencies <78,400 Hz

3b Filter gain <0.01 for frequencies >177,800 Hz

3c $S = M\,\Delta t = M/f_s = 2^{16}/2^{18}$ Hz = 0.25 s

3d H_0 corresponds to 0 Hz (d.c.); H_1 corresponds to 1/S = 4 Hz

3e The FFT produces coefficients H_n, where n = 0 to M–1. Therefore, the coefficient with the highest index is H_{M-1} or $H_{65,535}$, which corresponds to 4 Hz.

[2 points off for H_M and 0 Hz] [3 points off for H_M and 2^{18} Hz]

3f The FFT coefficient that corresponds to the highest frequency is $H_{M/2}$ or $H_{32,768}$. The corresponding frequency is (M/2)/S = 131,072 Hz

3g For a 4,000 Hz sinewave, the primary FFT coefficients are H_{1000} and H_{M-1000}. Additional neighboring coefficients H_{999}, H_{1001}, H_{M-999}, and H_{M-1001} are non-zero (actually half the value of the primary coefficients) due to the side lobes produced by the Hanning window.

[1 point off for omitting side lobes]

3h For a 4,000 Hz symmetric square wave, a sequence of harmonics will appear at odd multiples of the 4,000 Hz fundamental. So H_{k1000} and $H_{M-k1000}$ would be non-zero, and the Hanning side lobes would be at $H_{k1000-1}$, $H_{k1000+1}$, $H_{M-k1000-1}$, and $H_{M-k1000+1}$ (k odd).

[1 point off for omitting side lobes] [4 points off for omitting harmonics]

3i For a 4,002 Hz sinewave, H_{1000}, H_{1001}, H_{M-1000}, and H_{M-1001} would be non-zero and of equal magnitude, and the Hanning side lobes would appear at H_{999}, H_{1002}, H_{M-999} and H_{M-1001}.

[2 points off for omitting side lobes] [2 points off for omitting H_{M-1000} and H_{M-1001}]

3j The primary 4,000 Hz sinewave would produce non-zero values at H_{999}, H_{1000}, and H_{1001}. A second smaller sinewave of slightly higher frequency 4,000 + 4m Hz would produce non-zero values at $H_{1000+m-1}$, H_{1000+m}, and $H_{1000+m+1}$ (there are also complex conjugate coefficients at H_{M-1000}, etc.). For the smaller sinewave to appear as a separate peak, the coefficient H_{1001} must be below the coefficient at $H_{1000-m-1}$, which requires 1001 < 1000 +m –1, or m >2. The smallest value of m we can have is 3, which corresponds to a frequency 12 Hz above 4,000 Hz.

[4 points off for 4 Hz] [3 points off for 8 Hz] [both 12 Hz and 16 Hz were accepted]

3k To reduce the answer to 3j by a factor of two (i.e. to 6 Hz), sample for twice as long.

[2 points off for doubling the sampling frequency, since even if twice as many samples are taken, Δf remains 1/S]

3l A sinewave of frequency M – 84,000 Hz ($M = 2^{18}$) = 178,144 Hz will produce non-zero coefficients at H_{20999}, H_{21000}, H_{21001}, $H_{M-20999}$, $H_{M-21000}$, and $H_{M-21001}$. A sinewave of frequency 84,000 Hz will produce non-zero coefficients at exactly the same frequency indexes. This is an example of how a higher frequency can alias to a lower frequency. However, the 84,000 Hz sinewave will be only slightly reduced by the anti-aliasing filter (gain >0.90, while the 178,144 Hz sinewave will be greatly reduced (gain ≈0.01). So the coefficients will be about 100 times smaller for the 178,144 Hz sinewave.

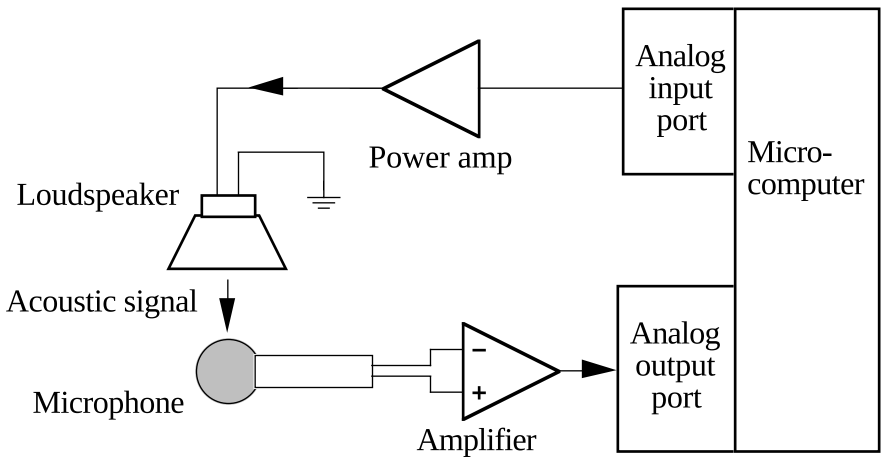
[3 points off for realizing that both frequencies produce the same non-zero magnitudes but stating that the magnitudes are the same]

[3 points off for giving a magnitude ratio of 100 but not giving the non-zero coefficients]

Note: a common mistake was to divide 178,144 Hz by 4 to get the frequency index- this is wrong because all frequencies above the Nyquist limit of 2^{17} Hz = 131,072 Hz are aliased to lower frequencies.

Another common mistake was that 178,144 Hz aliases to 178,144 Hz – 131,072 Hz = 47,072 Hz. Actually H_{M-m} aliases to H_m so 178,144 Hz aliases to 84,000 Hz.

4a



1 Generate an impulse (maximum voltage, width 1 µs) at the input of the power amplifier

2 immediately sample the output at 100 kHz to acquire the digitized impulse response c_i (number of samples is a power of 2).

3 Perform the FFT on c_i.

4b

$d(t) = a(t) \bullet c(t)$ $\quad$ • indicates convolution

$\mathcal{F}(d) = \mathcal{F}(a) \times \mathcal{F}(c)$ $\quad$ × indicates simple multiplication

$d(t) = \mathcal{F}^{-1}(\mathcal{F}(a) \times \mathcal{F}(c))$

[5 points off for having first equation only] [2 points off for having first two equations only]

4c

$a(t) = [a(t) \bullet b(t)] \bullet c(t)$ $\quad$ • indicates convolution

$\mathcal{F}(a) = \mathcal{F}(a) \times \mathcal{F}(b) \times \mathcal{F}(c)$ $\quad$ × indicates simple multiplication

$$b(t) = \mathcal{F}^{-1}\left[\frac{1}{\mathcal{F}(c)}\right] = \mathcal{F}^{-1}\left[\frac{1}{C_r + jC_i}\right] = \mathcal{F}^{-1}\left[\frac{C_r - jC_i}{C_r^2 + C_i^2}\right]$$

[5 points off for having first equation only] [2 points off for having first two equations only]

4d

1 From the 100 kHz sampled impulse response c_i, compute $b_i = FFT^{-1}[1/FFT(c_i)]$

2 Use b_i as an FIR digital filter on the 100 kHz sampled input stream a_i. This convolves a_i with b_i to produce f_i.

$$f_i = \sum_k b_k a_{i-k}$$

3 Sending f_i through the loudspeaker convolves f_i with c_i, which will result in a signal similar to a_i.

$$a(t) = f(t) \bullet c(t) = [a(t) \bullet b(t)] \bullet c(t) = a(t) \bullet [b(t) \bullet c(t)]$$

Note that b(t)•c(t)=delta function since FFT(b) x FFT(c) = 1

145M Final Exam Grades:

Problem	1	2	3	4	Total
Average	27.0	61.8	40.6	40.6	170.0
rms	4.0	10.5	8.3	8.4	24.9
Maximum	30	70	50	50	200

145M Numerical Grades:

	Lab total	Lab Partic.	Midterm #1	Midterm #2	Final	**Total**
Average	457.0	97.7	82.1	71.5	170.0	**878.3**
rms	44.9	1.8	10.8	14.4	24.9	**81.1**
Maximum	500	100	100	100	200	**1000**

145M Letter Grade Distribution

Letter Grade	Course Totals (1000 max)
A+	983*; 965; 935*
A	951; 923; 916; 909
A–	894; 887
B+	872; 868
B	833
B-	805
C+	
C	736
C–	698
D+	
D	

* graduate students