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

145M Microcomputer Interfacing Lab

Final Exam Solutions May 17, 1990

1A **Sample and Hold Amplifier:** Analog device that either amplifies an input signal or holds its output at a constant value, depending on a digital control signal.

1B **Transition Voltages:** The analog input voltages at which the output changes by one least significant bit.

1C **Frequency Aliasing:** Erroneous lower frequencies that arise when a waveform is periodically sampled at less than twice its maximum frequency.

1D **Glitch:** Brief erroneous spike that occurs in the output of a D/A converter when >1 input bits change at slightly different times.

1E **Digital Filter:** Method for transforming a series of digital values where each output value depends on some previous input and output values. (2 points off if dependence on previous inputs and outputs not mentioned.)

1F **Power Supply Sensitivity:** Ratio of % change in output voltage to a % change in power supply voltage.

2A **Successive Approximation:**
1) Set all N output bits to zero
2) For n = N to 1, repeat steps 3 and 4 (bit N is the MSB, bit 1 is the LSB)
3) Set bit n to one and send the N-bit number to a D/A converter
4) Compare the input with the D/A output. If greater, set bit n to zero

Number of steps = N

(3 points off if number of steps omitted or wrong, 4 points off if description of operation omitted)

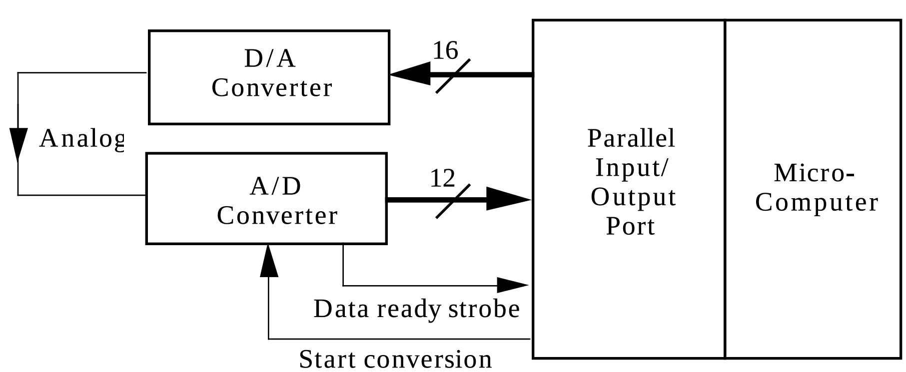
2B **Flash:** Input is sent to the V_+ input of $2^N - 1$ comparators. A series of resistors produces an ascending series of reference voltages, one to each V_- input of the comparators. The lower comparators with $V_+ > V_-$ have a logical output of one. The upper comparators with $V_+ < V_-$ have a logical output of zero. Fast digital logic generates the number of the comparator at the boundary, and this number is the digital representation of the analog input.

Number of steps = 1 Note: 3 points off if number of steps omitted or wrong

2C **Tracking:** The input to be converted is compared to the output of a D/A converter. The input of the D/A converter is the output of an up/down counter. If the D/A output is low, add one to the counter. If the D/A output is high, subtract one from the counter.

Number of steps = 2^N (Considers worst case Nyquist limit, where input swings between minimum and maximum values between samples.
Note: 3 points off if number of steps omitted or wrong

3A



3B Vary the digital input to the 16-bit D/A and convert the D/A output with the A/D converter being tested. Read the output of the A/D and determine the D/A input values where the A/D output changes by one bit. These correspond to the transition voltages. The absolute accuracy is the agreement of all the transition voltages with their ideal values. Alternatively, you could compute the center of the "steps" and compare those with their ideal values.

Note: 2 points off if you did not determine the transition voltages. If you simply compare the A/D output with the D/A input, then even a "perfect" A/D will have an absolute accuracy error due to quantization error. This point was covered in lecture and in the midterm solution sheets.

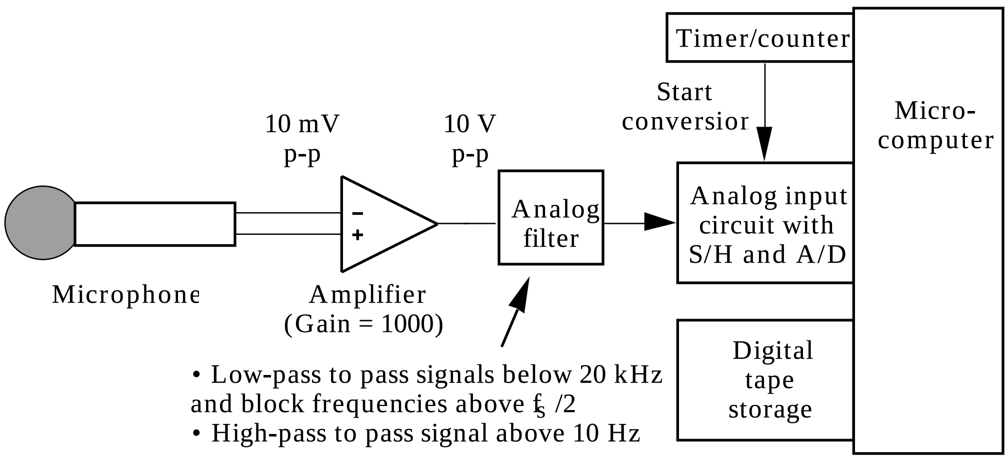
3C Compare the transition voltages measured in part 3B with a straight line passing through the lowest V(0,1) and highest $V(2^N-2, 2^N-1)$ transition voltages.

3D Compute the step sizes as the differences between neighboring transition voltages measured in part 3B. The differential linearity error is the differences between the step sizes and their average value.

3E The typical accuracy of the procedures in 3B, 3C, and 3D is determined by the ±1/2 LSB of the D/A. Since each A/D step is 16 times larger (12 bits vs 16 bits), the D/A accuracy is equivalent to 1/32 LSB of the A/D. Both 1/16 and 1/64 were accepted.

Note: some students apparently misinterpreted the question to read "what is the typical **value** of the quantities measured in parts A, B, and C" rather than ". . . the typical **accuracy** . . .".

4A



To keep the time interval between samples constant to one part in 10^6, a timer/counter unit must supply the "start conversion" pulses (i.e. hardware triggering). There is no way that such constancy can be obtained with software triggering. Note that timers can be very flexible pulse generators, where the pulse width and period can be set by a computer program.

Note: 4 points off if no A/D, 4 points off if no amplifier, 4 points off if no timer used to start conversion, 4 points off if no digital storage, 3 points off if no S/H, 3 points off if no anti-aliasing filter, 2 points off if no computer.

4B $f_s = 2 \times 20$ kHz = 40 kHz

4C An anti-aliasing filter is needed to suppress frequencies above $f_s/2$ while passing signal frequencies from 10 Hz to 20 kHz. Assuming that you use a filter that drops from nearly unity gain at 20 kHz to a low value at 30 kHz, you could design for f_s = 60 kHz. The four-pole filter you used in the lab exercise would only drop a factor of about 5 in amplitude from 20 to 30 kHz and an 8-pole filter would drop by a factor of about 25.

Note: Answers between 44 and 120 kHz were acceptable. 2 points off if the answer was 40 kHz and a good argument for an anti-aliasing filter with a perfectly sharp response at 20 kHz was not given.

4D A resolution of 2×10^{-5} is one part in 50,000, which requires 16 bits. (2^{16} = 64k). Conversion time must be < 16 µs for f_s = 60 kHz, < 10 µs for f_s = 100 kHz, etc.

Note: several students erroneously interpreted "the time interval between samples must be **constant** to one part in 10^6" to mean that the time interval had to be 10^{-6} s.

4E Integrating and tracking would be accurate enough but too slow. Flash would be more than fast enough, but 16 bits would require 65,535 comparators – not yet within available technology. **Successive approximation** would satisfy the requirements for speed and accuracy with minimum cost. (And in fact is used for digital recording)

4F (60 k samples/s) (3600 s/hr) (2 bytes/sample) = 432 megabytes. Answers from 288 to 720 megabytes were acceptable.

4G $T = 1/(20 \text{ kHz } \pi\, 2^{17}) = 0.121$ ns. Note: formula was on last page of exam.

5A F_0 corresponds to 0 Hz (dc). F_1 corresponds to 0.5 Hz (one cycle per S = 2 sec)

5B $F_{32,768}$ corresponds to the highest frequency of 16,384 Hz.

5C F_{200} corresponds to the pure 100 Hz tone. The complex conjugate appears at $F_{65,336}$. All other Fourier components should be zero.

Note: F_0 was optional. 2 points off if no complex conjugate.

5D Non-zero values are at F_{200}, F_{400}, F_{600}, . . . , $F_{32,400}$, $F_{32,600}$, $F_{32,736}$, $F_{32,936}$, . . . , $F_{65,136}$, $F_{65,336}$. This could also be written F_{200n} with complex conjugates at $F_{65,536\text{-}200n}$, where n = 1, 2, 3, etc.

Note: 2 points off if complex conjugates missing.

5E F_{200} and F_{201} would be equal because the pure 100.25 Hz tone lies exactly between them. Their complex conjugates would be at $F_{65,335}$ and $F_{65,336}$. Neighboring values are small but non-zero due to spectral leakage.

Note: 2 points off if harmonics such as F_{400} listed. 2 points off if small values called zero. 2 points off if no complex conjugates. 1 point off if F_{201} not listed.

145M Undergraduate Numerical Grades:

	7/8 x Lab	Midterm	Final	Total
Averages	652	93	164	909 (B+)
rms	35	6	25	56

145M Letter Grade Distribution

Letter Grade	Course Totals (1000 max)
A+	979, 982, 982
A	943, 946, 948*, 953, 954, 954, 954, 955, 961
A–	922, 923, 925,931,933, 936, 938
B+	913, 916
B	888, 891, 892, 896, 897
B–	841, 852, 854, 862, 872
C	779, 795
C–	766

*Graduate student