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

145M Microcomputer Interfacing Lab

Final Exam Solutions May 20, 1993

1A **R-2R Ladder** (of a D/A converter): Resistor network consisting only of resistors of value R and 2R that permits N switches to control a binary sequence of currents $I_i = I_0 2^i$.

1B **Average Step Size** (of an A/D converter): Average difference between all neighboring transition voltages. Also, the last transition voltage $V(2^N-2, 2^N-1)$ minus the first transition voltage V(0,1), divided by 2^N-2.

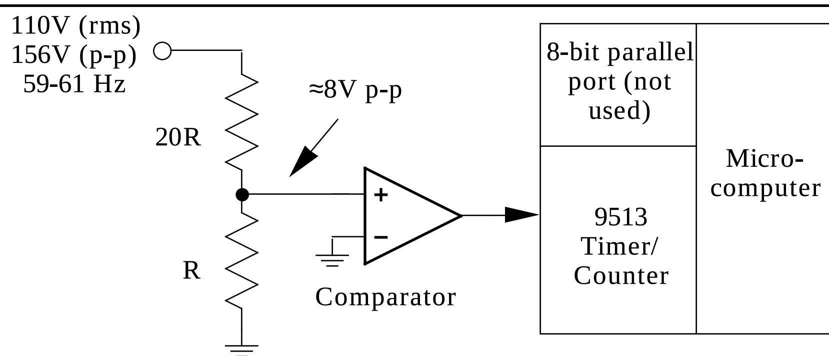
1C **Relative Accuracy** (of an A/D converter): Agreement between transition voltages V(n, n+1) plotted as a function of n and a straight line passing through the first and last transition voltages.

1D **Handshaking** (between a microcomputer and a digital circuit): Signal lines and protocol for ensuring accurate data transmission. Typical signals are “ready to send”, “ready to receive”, “data sent”, “data received”.

1E **Two’s Complement Operation:** Complement each bit and add 1.

1F **Tri-State Buffer:** Digital circuit with input and output lines, and a select line. When the select line is asserted, output = input. Otherwise the output is in a high-impedance state.

2A

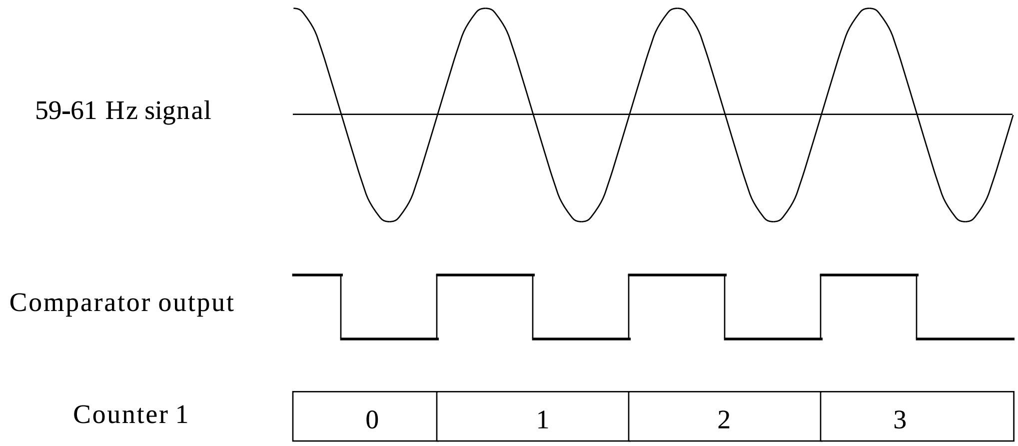


2B

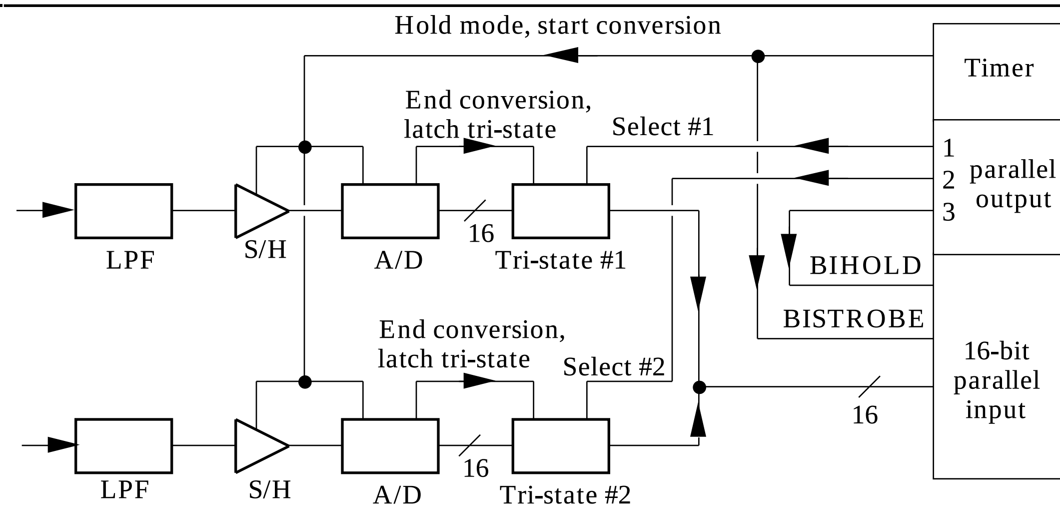
1. Set counter 1 for external input- increment on rising edge.
2. Set counters 2 and 3 for cascaded upward counting at 1 MHz
3. Zero and arm all three counters
4. Latch and read counter 1 in a loop until it reads 1
5. Latch and read counters 2 and 3, pack bytes to produce a time T1
6. Latch and read counter 1 in a loop until it reads 61
7. Latch and read counters 2 and 3, pack bytes to produce a time T2
8. Compute power line frequency as $60 \times 10^6/(T2 - T1)$

[4 points off if the exact time of zero crossings not used. Counting the number of cycles in 1 sec gives only integer answers 59, 60, and 61- this is not accurate to 0.001 Hz.

2C



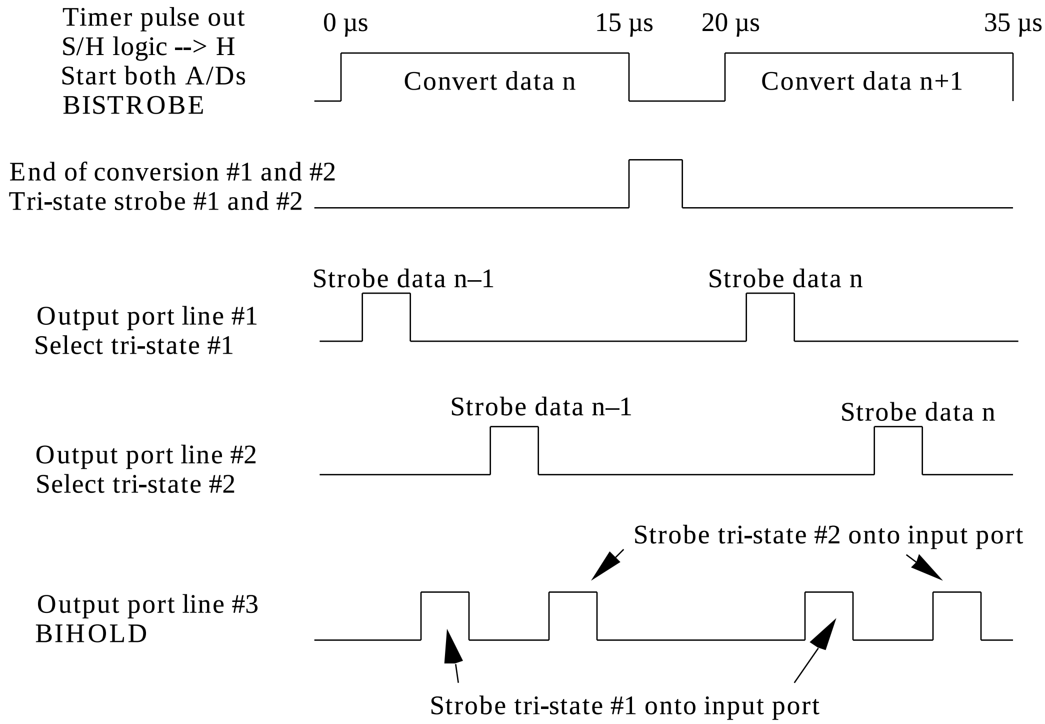
3A



3B

1. Initialize timer for positive external pulses, 15 μs wide, 50 kHz rep rate
2. Arm timer
3. Timer pulse puts both S/H in hold mode, starts both A/D converters
4. When conversion is complete, the two A/D converters strobe their data onto their respective tri-state registers.
5. The next timer pulse is detected by the program, which has been looping to sense BISTROBE
6. Program puts out a pulse in output line #1 to select tri-state buffer #1, which asserts its data onto the 16-bit parallel input port
7. After a short delay to allow the data to settle, the program puts a pulse on output line #3, which is connected to BIHOLD and latches the data onto the 16-bit parallel input port
8. The program reads the input port
9. The program puts a pulse on output line #2 to select tri-state buffer #2, which asserts its data onto the 16-bit parallel input port
10. repeat steps 7 and 8
11. Repeat steps 3-4 and 5-10 which occur in parallel, so that the program is reading old data from the tri-state buffers during the 15 μs that it takes the A/D converters to convert new data. Steps 3-4 take 15 μs while steps 5-10 should take about 6 μs.

3C



4A To sample 10 cycles of 20 Hz, S = 0.5 sec is required. 100 kHz x 0.5 sec = 50k samples

4B F_1 is the Fourier coefficient at a frequency of 1/S = 2 Hz

4C 20 Hz corresponds to $F_{10} = F_{49990}{}^*$

4D Because the signal is periodic and exactly 10 cycles were sampled, the only possible non-zero values are F_{10}, F_{20}, F_{30}, . . ., F_{50k-30}, F_{50k-20}, F_{50k-10} Note that the Butterworth filter suppresses the dc component, so F_0 =0 and that both even and odd harmonics can be present.

[6 points off for writing that all Fourier coefficients were non-zero; 5 points off for writing that all from F_0 to F_{10k} were non-zero; 4 points off for writing that all from F_0 to F_{10k} and all from F_{40k} to F_{50k} were non-zero; 3 points off for omitting F_{50-10n} to F_{50k}]

4E At 17.67 kHz, $f/f_c = 1/\sqrt{2}$ and $G = 1/\sqrt{1+2^{-8}} \approx 1 - 1/512 = 0.998$.

4F At 50 kHz, $f/f_c = 2$ and $G = 1/\sqrt{1+2^{16}} \approx 2^{-8} = 3.9 \times 10^{-3}$.

4G The Fourier transform taken in Part 1 of this problem is a measurement of the relative response of the auditorium as a function of frequency. At each frequency f_m of the graphic equalizer, determine the closest Fourier amplitude $|F_n|$, where $n \approx f_m/2$. Then the graphic equalizer gains should be $g_m = \max(F_n)/F_n$. [5 points off for formulas but no explanation; 3 points off if no relationship shown between n and f_m]

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5 Method 1 (batch mode):

1 Take the FFT of the 44 kHz compact disk values

2 Rescale the frequency axis so that 44 kHz becomes 48 kHz

3 Do the inverse FFT to create new digital values at 48 kHz sampling

Method 2 (continuous mode);

1 Play the compact disk data through a standard compact disk player, which interpolates the steps in the D/A output waveform and low-pass filters to give a smooth analog waveform
2 Sample this waveform at 48 kHz and digitize

[5 points off if a no interpolation or filtering used to smooth the higher harmonics caused by the abrupt edges of the 44 kHz D/A output steps. These higher frequencies can cause severe aliasing]

[15 points off if the DAT digital values are the same as the CD digital values]

145M Numerical Grades:

	5/9 x Lab	Lab Partic.	Midterm #1	Midterm #2	Final	Total
Average	478	100	84	92	162	921 (B+)
rms	23	0	12	4	28	53
Maximum	500	100	100	100	200	1000

145M Letter Grade Distribution

Letter Grade	Course Totals (1000 max)
A+	990
A	955, 958, 973
A–	935, 941, 943, 946, 946
B+	919, 921
B	868, 880, 886, 897
B–	none
C	772