UNIVERSITY OF CALIFORNIA

College of Engineering

Electrical Engineering and Computer Sciences Department

145M Microcomputer Interfacing Lab

Final Exam Solutions May 18, 1992

1A **Nyquist Theorem:** To recover a waveform from its sampled values, the highest frequency present must $\leq$ one-half the sampling frequency.

1B **Discrete Fourier Transform:** Transform for determining the amplitude of the frequency components of a periodically sampled waveform.

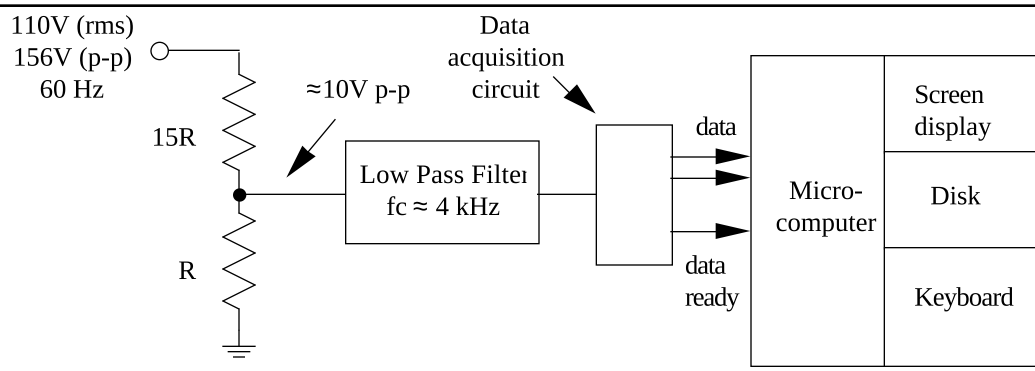
1C **Differential Linearity Error** (of an A/D converter): Difference between the spacing of neighboring transition voltages and their average spacing. [4 points off if step size or transition voltage not mentioned] [4 points off if absolute or relative accuracy was defined]

1D **Anti-Aliasing Filter:** Low-pass filter used to block frequencies above one-half the sampling frequency and thereby prevent aliasing.

1E **Power Amplifier:** Amplifier having high power or current output, and required to drive an actuator such as a speaker or heater. [2 points off if high power or current output not mentioned]

1F **Digital Filter:** Filter whose output is a linear combination of previous input and output values.

2A



[3 points off if 156 V p-p sent directly into acquisition circuit]
[3 points off if low-pass filter omitted]

2B $\Delta f = 0.01$ Hz, $S = 1/\Delta f = 100$ sec, $N = 100$ sec x 10 kHz $= 10^6$ samples
(S = 50 sec, $N = 5 \times 10^5$ samples also acceptable)

2C F_0 corresponds to 0 Hz or dc.

2D 60 Hz corresponds to F_{6000} and F_{N-6000} where $N = 10^6$

2E Since the distortion has a period of 60 Hz, only multiples of 60 Hz will be present.
The highest harmonic n_{max} that can pass the anti-aliasing filter is $\approx$ 5000 Hz/ 60Hz $\approx$ 80.
The nth harmonic will be at F_{6000n} and $F_{N-6000n}$
[3 points off if only frequencies given]
[6 points off if only F_{6000} and F_{N-6000} given]
[6 points off if answer says that all Fourier amplitudes are non-zero]

2F
1) Data acquisition circuit samples waveform, digitizes, and sets data ready line
2) When program detects data ready line, it reads data, stores data, and resets data ready line
3) Loop back to step 1 until 10^6 values taken
4) Multiply values by Hanning window
5) Compute the FFT

[2 points off for each step missing]

3A The first harmonic could be anywhere in the 59.9 to 60.1 Hz range, which corresponds to 20 potentially non-zero Fourier coefficients (40 filters) from F_{5990} to F_{6010}.

The range of the 80th harmonic will involve 40 x 80 = 3200 filters.

The average number of filters per harmonic = 1600.

The total number of filters is then 1600 x 80 = 128,000 (much less than the total number of 500,000 real plus 500,000 imaginary components)

[4 points off for an answer of 2-5 filters]

[3 points off for answers of 10-40 filters or 10^6 filters]

[2 points off for an answer from 3000 to 7000 filters]

[2 points off if the 59.9 to 60.1 Hz range was neglected]

3B The real FIR filter is given by

$$F_n = \sum_{k=0}^{N-1} f_k \cos(2\pi nk/N)$$

where n = 100f and N = 10^6.

[2 points off if frequency f missing]

The real IIR filter is given by

$$F_n(t+\Delta t) = \left[F_n(t) - f_0 + f_N\right]\cos(2\pi n/N)$$

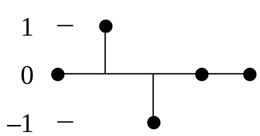
where n = 100f and N = 10^6.

3C
1 Take 10^6 samples (f_0 to f_{N-1}) and store in memory
2 Compute all needed coefficients in a loop
3 Start conversion and read new sample when ready
4 Delete oldest value (f_0), store new value (f_N) after last value (f_{N-1}), and shift all f_k to f_{k-1}
5 Loop back to step 2

[3 points off for each missing step]

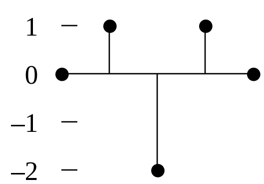
4A

$y_0 = 0$ $y_1 = 1$ $y_2 = -1$
$y_3 = 0$ $y_4 = 0$



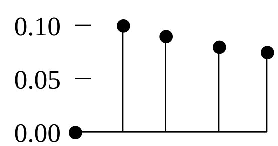
4B

$y_0 = 0$ $y_1 = 1$ $y_2 = -2$
$y_3 = 1$ $y_4 = 0$



4C

$y_0 = 0$ $y_1 = 0.100$ $y_2 = 0.091$
$y_3 = 0.081$ $y_4 = 0.073$



4D

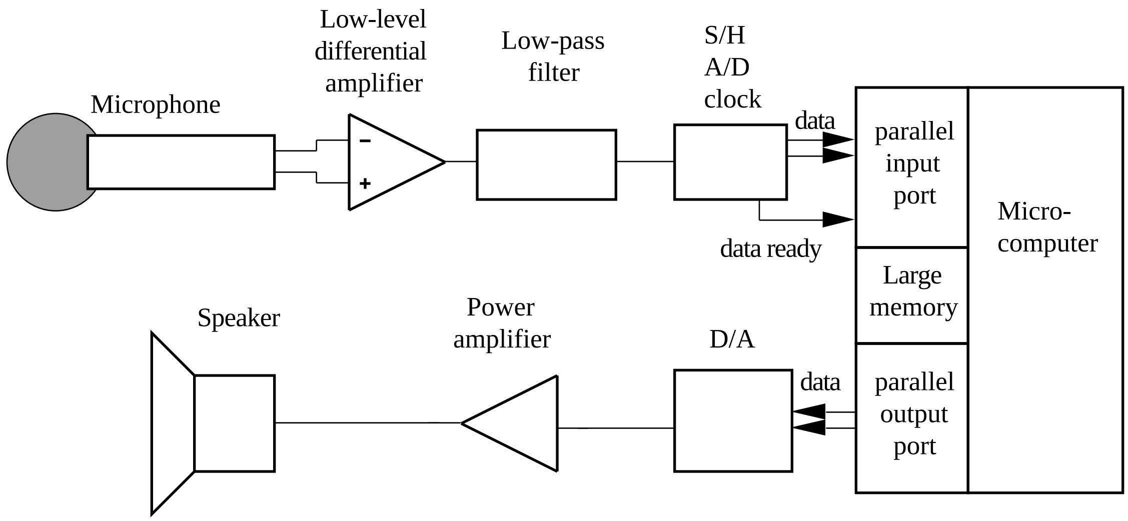
Second derivative — b
First derivative — a
Low pass filter — c

a — FIR
b — FIR
c — IIR

5A
1 With an input impulse, sample the output waveform h(t) and perform the FFT.
2 Perform the FFT of the desired output y(t)
3 Compute the required input u(t) using

$$u(t) = FFT^{-1}\left(\frac{FFT(y)}{FFT(h)}\right)$$

5B



[1 point off for each amplifier missing]   [1 point off for missing low pass filter]

5C

The digital filter is given by

$y_i = (1-\alpha)x_{i-1} + \alpha y_{i-1} \qquad \alpha = \exp(-T/RC)$

where T is the cycle time (sampling x_{i-1} + computing y_i)

Gain vs. frequency is similar to the analog filter.

Phase is similar but shifted later by T.

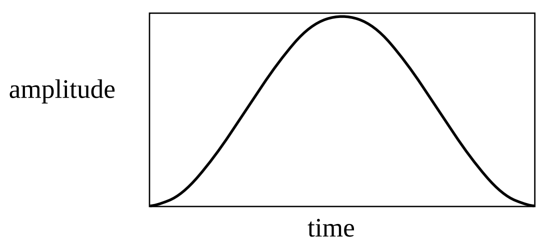
[1 point off for exp(-1/RC)]   [2 points off if processing delay omitted]

[3 points off if α not related to RC]

5D

The simple rectangular window allows a discontinuity between the ends of the sampled values.

To avoid this, need to multiply the time samples by a function that smoothly approaches zero at both ends:



145M Numerical Grades:

	6/9 x Lab	Lab Partic.	Midterm	Final	Total
Average	564	100	78	175	916 (B+)
rms	24	0	17	20	57
Maximum	600	100	100	200	1000

145M Letter Grade Distribution

Letter Grade	Course Totals (1000 max)
A+	980*, 969
A	968, 965, 960, 957, 949*
A–	none
B+	929, 919, 913, 904, 903
B	891, 884
B–	none
C	840
D	753

* Graduate students