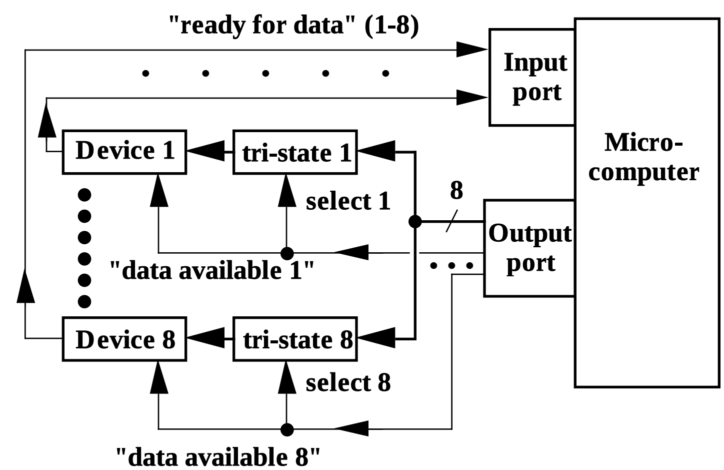
**UNIVERSITY OF CALIFORNIA**
**College of Engineering**
**Electrical Engineering and Computer Sciences Department**

# 145M Microcomputer Interfacing Lab

## Final Exam Solutions May 19, 1995

**Problem 1a**




**[3 points off if no tri-state between output bus and each device or no select line to each device]**
**[2 points off for each if "data available" or "data taken" lines missing]**

**Problem 1b**
**Use 8 input lines for "ready for data" from each device.**
**Use 8 output lines for "data available" to each device and to select the corresponding the octal tri-state.**
**Use 8 output lines for the output data bus**
**The "full handshaking" steps are:**

1. The computer sets all 8 "data available"/select lines FALSE
2. The computer checks that device n (the one we want to write to) has "ready for data" TRUE
3. The computer writes the 8 data bits to the inputs of all octal tri-states
4. The computer asserts "data available"/select output line n TRUE (or the computer asserts "data available" and the device selects an octal tri-state driver.)
5. The device detects "data available" and sets "ready for data" FALSE
6. The device reads the data and sets "ready for data" TRUE
7. The computer detects the "ready for data" FALSE-TRUE edge and sets "data available"/select FALSE

**The minimum accepted for full credit:**

1. Device n tells computer "ready"
2. Computer selects tri-state n (or device n)
3. Computer asserts data on data bus
4. Computer tells device n "data available"
5. Device n reads data

**Problem 2**

1. Set all D/A bits to 0
2. Set i = N (the MSB)
3. Set bit i to 1
4. If D/A output > A/D input, set bit i to 0
5. i = i – 1
6. If i > 0, go back to step 3

**Problem 3**

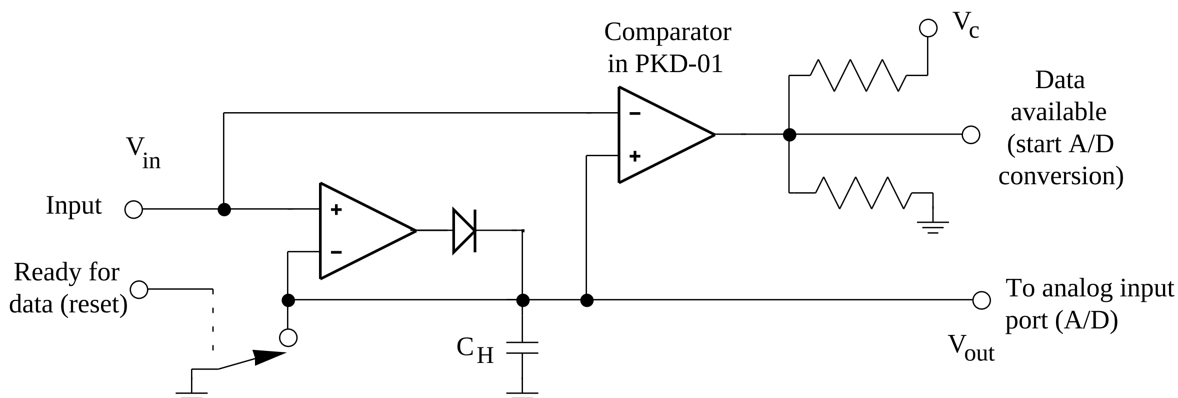
- **Use a microcomputer with analog input and a clock to trigger periodic sampling**
- **Use an anti-aliasing filter to eliminate white noise above 1/2 the sampling frequency**
- **Sample M values over a time window S.**
- **Use a Hanning window to reduce the spectral leakage of the white noise background into the regions of the spikes.**
- **Perform the FFT.**
- **Due to the random, non-periodic noise, it is not possible to see the periodic signal in the time domain- but the FFT concentrates the periodic signal at discrete frequencies- the harmonics.**
- **The first harmonic should always be non-zero. If it appears at Fourier index $n_1$, then the period $P = S/n_1$, where S is the sampling window width. Depending on the harmonic content of the periodic signal, there may be other non-zero harmonics. The kth harmonic will appear at $n_k = k\, n_1$.**
- **If no spikes are visible, then S is shorter than the period P, and it will be necessary to increase S and try again.**

**[10 points off for trying to solve the problem entirely in the time domain]**

**[3 points off for sampling, taking the FFT, subtracting the average noise, zeroing the values between the harmonic spikes, and then taking the inverse FFT to measure the period in the time domain. This approach requires pattern recognition and is not as accurate or as easy as finding the index of the first harmonic in the FFT.]**

**Problem 4a**

**The idea was to use negative feedback to track a rising pulse on $C_H$. When the pulse reaches its peak and starts dropping, the diode effectively disconnects $C_H$, and the comparator produces an edge that starts the A/D conversion of the peak value held on $C_H$. When conversion is complete, the capacitor is reset.**




**[1 point off for not showing resistors] [3 points off for unlabeled lines, extra lines]**
**[4 points off for missing lines]**

**Problem 4b**

1. **Pulse arrives**
2. **$C_H$ charges**
3. **When pulse passes peak value, $V_{out}$ becomes greater than $V_{in}$, triggering the comparator to produce a "data available" pulse.**
4. **The leading edge of this pulse starts A/D conversion**
5. **After conversion, computer sends a pulse on the "ready for data" line to the PKD-01 reset, which discharges $C_H$.**

**[5 points off for using a large threshold voltage and a delay]**

**Problem 4c**

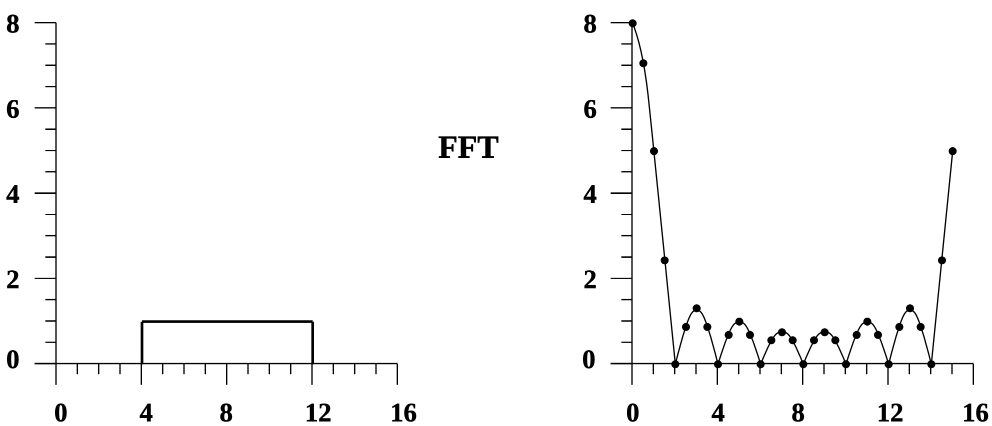
| | |
|---|---|
| **PKD-01 settling time** | **45 µs** |
| **Comparator trigger time** | **150 ns** |
| **Switching time** | **100 ns** |

A/D Conversion time 50 µs
**TOTAL** 95.25 µs (≈ 10,000 pulses per s)
**[2 points off for omitting the settling time]**

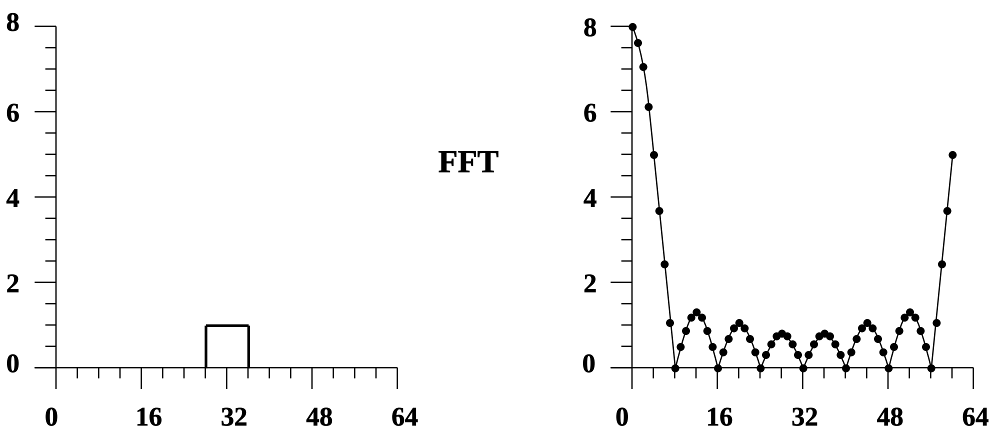
**Problem 4d**
**1 LSB = (20 V)/4095 ≈ 5 mV**
**Droop = (0.1 mV/ms) (50 µs) = 5 µs = 0.001 LSB**
**[1 point off for factor of 1000 error (confusing µs with ms)]**
**[2 points off for not giving an answer in LSB] [2 points off for using the wrong time in the calculation]**

**Problem 5a**
**Let M = total number of samples, S = sampling window width (in seconds), and m = width of the rectangle wave.**
**The DFT defines $H_0$ as the sum over the time values. In these three cases , the height of the rectangle = 1, so $H_0$ = m.**
**As given at the end of the exam, the Fourier transform of a rectangle of width $T_0$ and amplitude 1 is $H(f) = T_0 \sin(\pi T_0 f)/(\pi T_0 f)$, which is zero whenever $f = k/T_0$ for k = any positive integer.**
**In the discrete domain, H(f) becomes $H_n$, the width $T_0$ = mS/M and the frequency f = n/S. The condition for $H_n$ = 0 becomes n/S = k/(mS/M) or n = kM/m.**
**So the DFT has a d.c. value $H_0$ = m and is zero whenever n = kM/m, where k = any positive integer.**

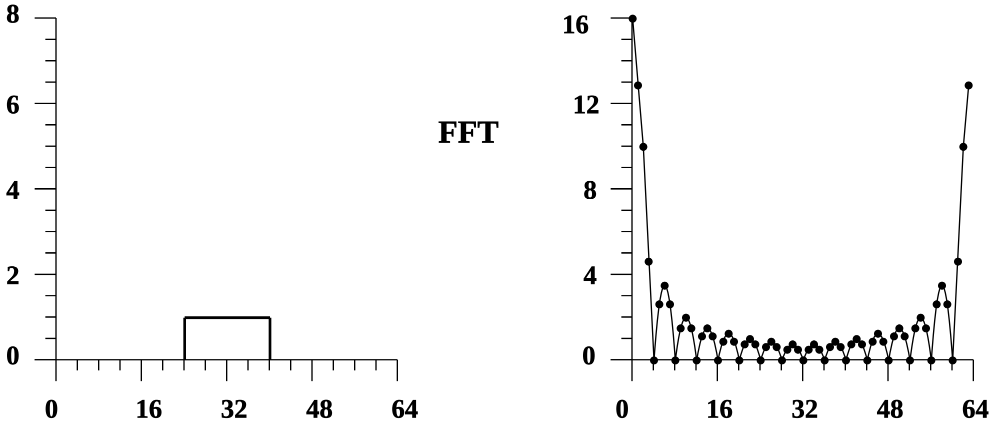
**The first transform had M = 16, m = 8. So $H_0$ = 8 and $H_n$ =0 for n = 2k.**




**The second transform had M = 64, m = 8. So $H_0$ = 8 and $H_n$ = 0 for n = 8k. The purpose was to show that for a rectangle of width 8, increasing the sampling window from 16 to 64 increased the frequency resolution by a factor of 4.**

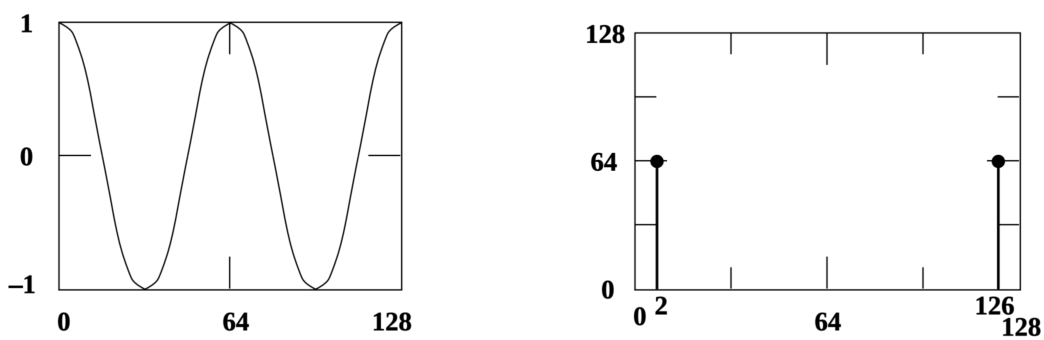


**The third transform had M = 64, m = 16. So $H_0$ = 16 and $H_n$ = 0 for n = 4k. The purpose was to show that for the same M = 64, increasing the rectangle width from 8 to 16 decreased the width of the transform by a factor of 2.**

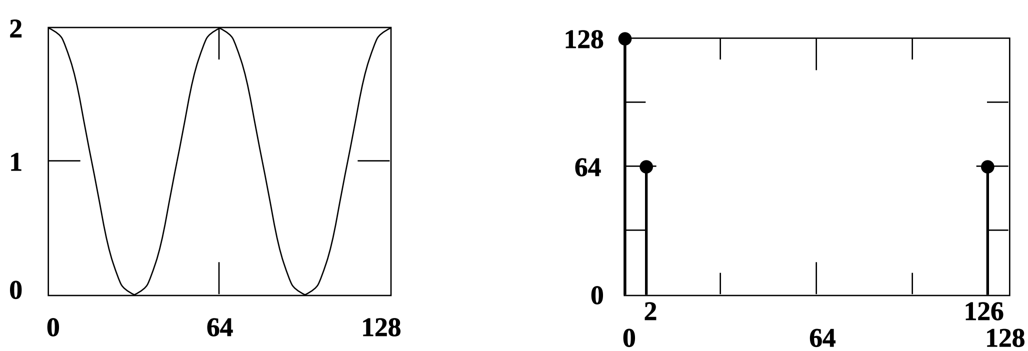



**[3 points off for no sinc function]**
**[3 points off if the second transform did not have more resolution than the first transform]**
**[3 points off if the third transform did not have narrower lobes than the second transform]**

**Problem 5b**
**average (d.c.) is zero, exactly 2 cycles per 128 samples.**

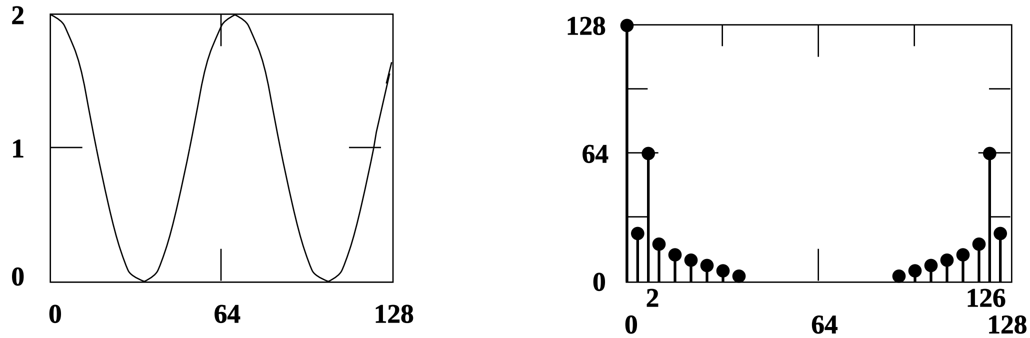


**As above, but average shifted from 0 to 1**



**[2 points off per missing line, 1 point off for location, 1 point off for magnitude]**

**Problem 5c**
**Same as second graph above, but with spectral leakage**

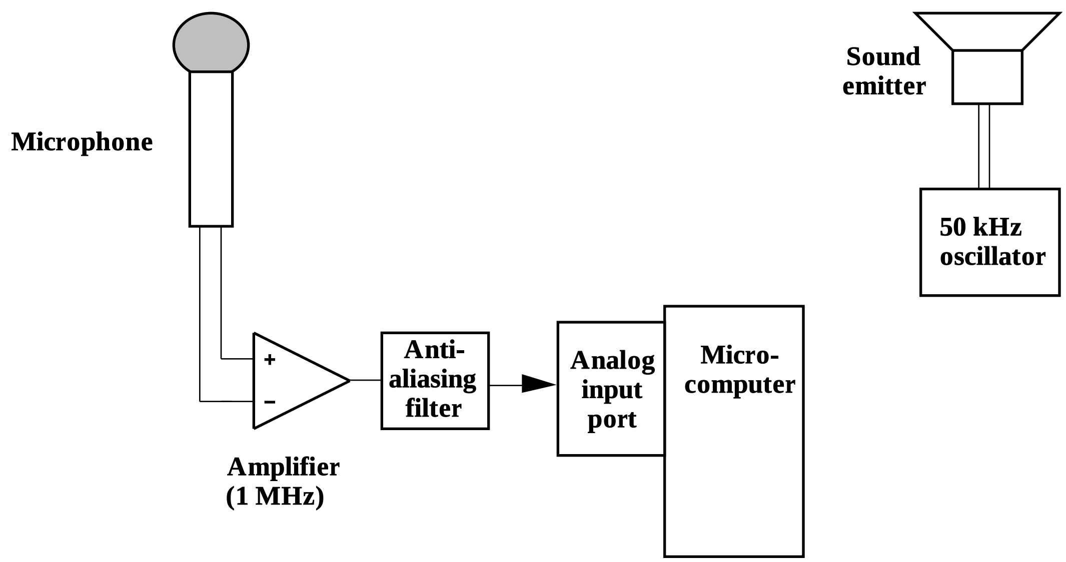



**[2 points off if d.c. not consistent with second graph above]**

**Problem 5d**

**Since 5 square waves were sampled, we expect non-zero magnitudes at n = 5 and 123 (first harmonic at 15 Hz), n = 15 and 113 and (third harmonic at 45 Hz), n =25 and 103 (fifth harmonic at 75 Hz), etc.**
**The additional non-zero magnitudes at n = 20 and 108 corresponds to a pure 60 Hz harmonic, which must be due to power line interference in the lab.**
**[3 points off if odd harmonics not identified]**
**[4 points off if 60 Hz electromagnetic interference not identified]**

**Problem 6a**
- **Use an anti-aliasing filter to block white noise from the 1 MHz amplifier that would otherwise obscure the weak signal.**
- **Use a Hanning window to reduce spectral leakage in the return echo, which could be at any frequency**




**[3 points off for using a 2 MHz low pass filter- much higher than needed]**
**[2 points off each for missing microphone or sound emitter]**

**Problem 6b**

- The highest frequency to be expected for an automobile moving toward the sound sensor will be on the order of 60 kHz. A practical anti-aliasing filter that passes frequencies from d.c. to 60 kHz will only be able to block frequencies above 80 kHz.
- Therefore it will be necessary to sample at or above 160 kHz.

[3 points off for $f_s$ < 100 kHz; 2 points off for $f_s$ = 100 kHz; 1 point off for $f_s$ = 120 kHz]

**Problem 6c**

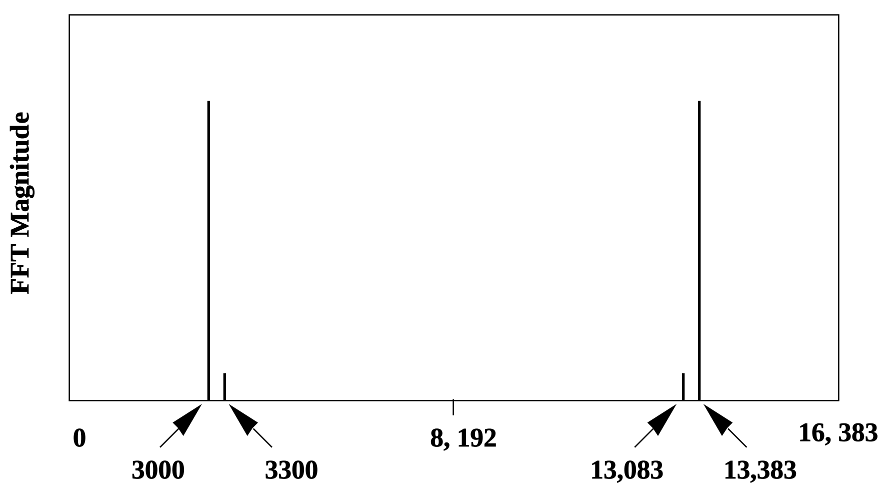
- A change in velocity of 0.1 m/s means a change in frequency $\Delta f$ = (50 kHz)(0.1 m/s)/(300 m/s) = 16.7 Hz. If this is to be the frequency interval between the Fourier transform coefficients, the sampling window must be at least S = 1/16.7 Hz = 0.06 0 s.
- Sampling at the minimum frequency of 160 kHz for the minimum time 0.06 s means taking 9,600 samples. This is not a power of 2, so we can either increase the sampling frequency (and reduce the requirements on the high frequency fall-off of the anti-aliasing filter) or increase the time interval S (and improve the frequency resolution of the FFT) to take 16,384 samples., which is the next power of two above 9,600.
- Taking only 8,192 samples during 0.060 s means a sampling frequency of 136 kHz, and the anti-aliasing filter must pass all frequencies up to 60 kHz and effectively reject all frequencies above 68 kHz. A Butterworth filter with G = 0.9 at 60 kHz and G = 0.01 at 68 kHz would require 43 poles, which is not practical.

  (Note: the formula for the Doppler shift given in the final was simplified to make the problem easier. The proper formula for the Doppler shift is $f_0 = f(1 + v/v_s)/(1 - v/v_s)$, where $f_0$ is the observed frequency, $f_s$ is the frequency at the source, v is the velocity of the object, and $v_s$ is the speed of sound)

[2 points off for 8,192 samples; 3 points off for less]

**Problem 6d**

- For v = 30 m/s, $\Delta f$ = (50 kHz)(30 m/s)/(300 m/s) = 5 kHz.
- In addition to a large echo at 50 kHz from stationary objects, there will be a smaller Doppler shifted echo at 55 kHz from the object moving toward the sound sensor.
- In the FFT plot, $F_1$ is the magnitude at 16.6 Hz, $F_{3000}$ is the magnitude at 50 kHz, $F_{3300}$ is the magnitude at 55 kHz. For a 16,384 point FFT, $F_{13,384} = F_{3000}$ and $F_{13,084} = F_{3300}$.




[2 points off if no echo from stationary objects]

**Problem 7a**

- **The Fourier transform of the sum of two functions is the sum of their Fourier transforms**
- **The Fourier transform of the product of two function is the convolution of their Fourier transforms**

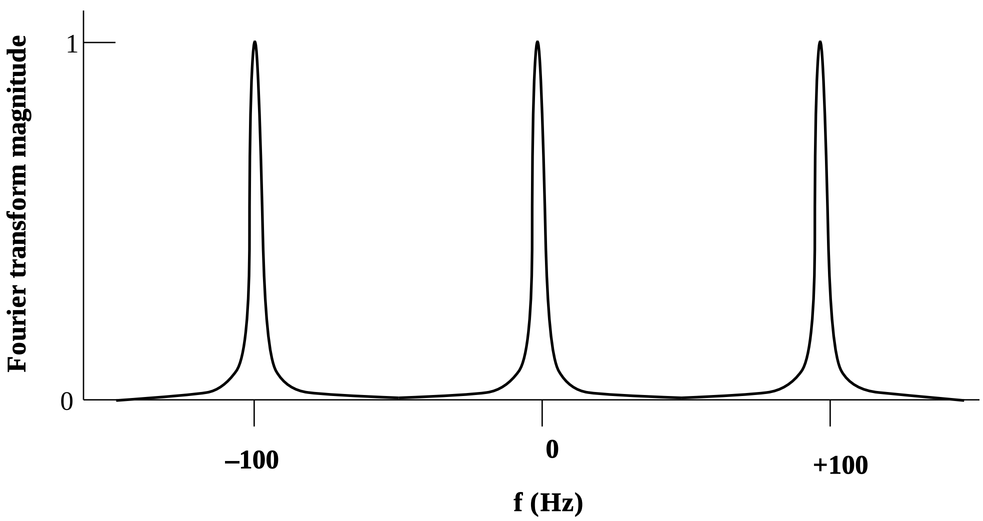
$$c(t) = F\left[e^{-t/\tau}\right] + F\left[2e^{-t/\tau}\cos(2\pi f_0 t)\right]$$

$$= \frac{1}{\sqrt{1+4\pi^2\tau^2 f^2}} + \left[\frac{2}{\sqrt{1+4\pi^2\tau^2 f^2}}\right]\left[\frac{\delta(f-f_0)+\delta(f+f_0)}{2}\right]$$

$$= \frac{1}{\sqrt{1+4\pi^2\tau^2 f^2}} + \frac{1}{\sqrt{1+4\pi^2\tau^2 (f-f_0)^2}} + \frac{1}{\sqrt{1+4\pi^2\tau^2 (f+f_0)^2}}$$

**Problem 7b**

- **The Fourier transform consists of three peaks, at –100 Hz, 0 Hz, and + 100 Hz.**
- **Each peak has the same profile as a one-pole low-pass filter, with a corner frequency that is $f_c = 1/(2\pi\tau)$ = 0.159 Hz away from the center of the peak. So the peaks are narrow compared to their separation.**
- **In between the peaks (say at 50 Hz), the Fourier magnitude is 0.0032 + 0.0032 + 0.0011 = 0.0075**




**Problem 7c**

- **The Fourier convolution theorem says that if a(t) is the convolution of b(t) and c(t), then $F$(a) is equal to the product $F$(b) x $F$(c).**
- **So the input b(t) that when convolved with the impulse response c(t) gives a square wave a(t) is computed as:**

$$b(t) = F^{-1}\left[F(a)/F(c)\right]$$

- **$F$(c) was computed in part 7a above and $F$(a) is well known.**
- **The above example is simple enough to be done analytically, using integral Fourier transforms, but it could also be done using the FFT.**

**145M Final Exam:**

| Problem | 1 | 2 | 3 | 4 | 5 | 6 | 7 | Total |
|---|---|---|---|---|---|---|---|---|
| Average | 26.1 | 12.3 | 12.1 | 25.7 | 21.7 | 24.7 | 23.1 | 145.7 |
| rms | 3.9 | 3.9 | 3.3 | 9.3 | 9.2 | 4.6 | 4.6 | 32.1 |
| Maximum | 30 | 15 | 15 | 40 | 40 | 30 | 30 | 200 |

**145M Numerical Grades:**

| | Lab total | Lab Partic. | Midterm #1 | Midterm #2 | Final | Total |
|---|---|---|---|---|---|---|
| Average | 458.6 | 91.5 | 84.5 | 86.3 | 145.7 | 866.6 |
| rms | 30.9 | 8.5 | 7.1 | 10.6 | 32.1 | 102.4 |
| Maximum | 500 | 100 | 100 | 100 | 200 | 1000 |

**The average scores of the two lab report graders differed by only 0.03 points. No equity correction was necessary.**

**145M Letter Grade Distribution (passing students only)**

| Letter Grade | Course Totals (1000 max) |
|---|---|
| A+ | 952 |
| A | 920, 944, 946*, 948.5*, 950.5* |
| A– | 897.5, 899 |
| B+ | 870, 872.5, 873, 877 |
| B | 838, 849, 855, 855, 858 |
| B– | none |
| C+ | 794, 804, 814 |
| C | none |
| C– | none |
| D+ | none |
| D | none |
| D– | 667 |

*** graduate students**