Basics of sampling and signal processing

***** You are free to complete this assignment in whatever programming language you prefer *****

Download the following files from Canvas:

1. empty_fs1kHz.csv
2. LP_fs1000Hz_fc40Hz.csv

Each file contains a single dimension array of values.

fs = sampling rate (frequency of samples).

fc = cut off frequency (frequency at cut = -3dB point of the filter)

The empty_1kHz.csv is a synthesized signal representing a typical recording of an EMPTY pin on your microcontroller. The only source connected to this EMPTY pin is the air around the device. Thus there are two main components here: white noise, and the 60Hz component of the AC mains power. If you are interested, I have posted the MATLAB script used to generate the EMPTY signal and the filters (which are essentially signals themselves).

First, we need to explore our signal. These are always useful initial steps.

1. What is the size of the EMPTY_fs1kHz signal?
    a. The data are sampled at 1000 times per second (1kHz). What is the <u>duration</u> of the signal, based on the SIZE and the SAMPLING RATE?
2. Plot the full EMPTY_1kHz signal.
    a. Make a plot where the x-axis unit is TIME (seconds) and y-axis is the Voltage of the EMPTY_1kHz signal.
3. Plot only 0.25s of the EMPTY_1kHz signal. Again, x-axis is TIME, y-axis is Voltage.

At this point you have visualized the signal in the TIME DOMAIN. Now we will inspect the signal in the FREQUENCY DOMAIN before we make any manipulation of the signal.

4. Generate a spectral analysis of the full signal.
    a. You can use any built-in function or package that you are comfortable with. In Python, scipy has great tools for this task. Ask us if you do not know where to find a spectral analysis package.

b. There will be two figures for this spectral analysis (see the Bode plots in the slides for an example). Each plot has an x-axis of FREQUENCY. One y-axis will be MAGNITUDE. The second figure y-axis will be PHASE.

At this point, you have inspected the signal in the most common ways, TIME DOMAIN, zoomed in TIME DOMAIN, and FREQUENCY DOMAIN.

Our first manipulation will be to reduce the sampling rate of our EMPTY signal.

5. Reduce sampling rate to 100Hz, repeat the following plots:
    a. TIME DOMAIN zoomed in
    b. FREQUENCY DOMAIN magnitude only.
6. Reduce sampling rate to 50Hz, repeat the plots again:
    a. TIME DOMAIN zoomed in
    b. FREQUENCY DOMAIN magnitude only.

Now we will be manipulating the original 1000Hz signal using our FILTER:

7. Go back to your 1000Hz signal. Convolve the Low-Pass filter with the ORIGINAL empty_fs1kHz signal. For the resulting, filtered signal (still a 1000Hz sampling rate)
    a. What is the duration of this result? Make sure it makes sense before proceeding with the plots:
    b. TIME DOMAIN zoomed in
    c. FREQUENCY DOMAIN magnitude only.
8. Now take the result of the convolution in #7 and reduce the sampling rate again to 100Hz. First create the plots again for this new signal:
    a. TIME DOMAIN zoomed in
    b. FREQUENCY DOMAIN magnitude only.
    c. Give a brief written answer: Why is the Frequency Domain magnitude so different for #**5B** and #**7C**? In comparing your plots from 5B, 6B, and 7C, what do you notice is happening to the 60Hz ‘noise’ component in this signal?