ASSIGNED: Apr. 11, 2013. **READ:** Sects. 11.1-11.4 if you haven't already.
DUE DATE: UNGRADED. **TOPICS:** Multirate filtering, DSP noise filtering.

NOTE: This problem set will not be collected or graded.

[10] 1. *Using multirate filtering to alter the pitch of musical notes:*
You have a snippet of music at note E. You want to change note E to note A.
Note A has frequency 2/3 that of note E. Design a multirate system to do this.

[30] 2. *Drill on multirate filtering and aliasing:*
A 400 Hertz sinusoid is input to the DSP systems below at $1000\frac{\text{SAMPLE}}{\text{SECOND}}$.
The output of each one is one or more sinusoids. Compute their frequencies.

[10] a. 400 Hz$\rightarrow\boxed{\textbf{A/D}}\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow\boxed{\downarrow \mathbf{3}}\rightarrow\boxed{\textbf{D/A}}\rightarrow$?
[10] b. 400 Hz$\rightarrow\boxed{\textbf{A/D}}\rightarrow\boxed{\downarrow \mathbf{3}}\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow\boxed{\textbf{D/A}}\rightarrow$?
[10] c. 400 Hz$\rightarrow\boxed{\textbf{A/D}}\rightarrow\boxed{\downarrow \mathbf{4}}\rightarrow\boxed{\uparrow \mathbf{2}}\rightarrow\boxed{\textbf{D/A}}\rightarrow$?
Try checking your answers using Matlab (see my lecture notes for Matlab commands).

Download `p0.mat` from the web site. >>`load p0.mat` to get sampled signals `X1,X2,X3`.

[20] 3. `X1` is a periodic signal, with noise added, sampled at $1000\frac{\text{SAMPLE}}{\text{SECOND}}$.
[5] a. Plot `X1` as a continuous-time signal (use `plot`, not `stem`).
[5] b. Compute its spectrum using `FX=fft(X1)/length(X1)`. Plot its magnitude for $0 \le f \le 20$ Hz. Note f=(K–1)$\frac{1000}{1000}$=K–1, so plot `abs(FX(1:20))` vs. `[0:19]`.
[5] c. *Threshold* its spectrum using `FX(abs(FX)<0.9)=0`. Plot it for $0 \le f \le 20$ Hz.
[5] d. Plot `length(X1)*real(ifft(FX))` for the *thresholded* `FX`. The noise is gone, even though we didn't know *anything* about the signal except that it was periodic. That told us its spectrum was a line spectrum, so we could eliminate the noise by setting everything that wasn't a large line to 0. Thresholding was an easy way.

[20] 4. `X2` is a trumpet, with noise added, sampled at $44100\frac{\text{SAMPLE}}{\text{SECOND}}$ (standard CD rate).
Repeat #3. Use a threshold of 0.0015, and plot the spectrum for $0 \le f \le 4000$ Hz.
Listen to the noisy and filtered trumpets. The noise is eliminated again!

[20] 5. `X3` is 2 trumpets playing simultaneously notes A and B, sampled at $44100\frac{\text{SAMPLE}}{\text{SECOND}}$.
[5] a. Plot `X3(1000:1199)` as a continuous-time signal (use `plot`, not `stem`).
[5] b. Plot the spectrum for $0 \le f \le 4000$ Hz. Note the pairs of harmonics.
[5] c. Knowing *only* that note B has a higher frequency than note A, eliminate the trumpet playing note A by setting its harmonics to 0.
[5] d. *Listen* to the result in the time domain. Only one trumpet is left!

"The value of an idea lies in the using of it"–Thomas Edison. DSP applications.