EECE 403/525 HW #2

Due 2/13/2023 @ 11:59PM

See also the provided following files posted on Bb:

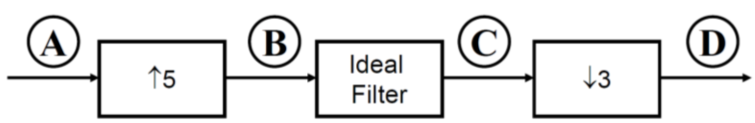
- guitar1.wav

Notes:

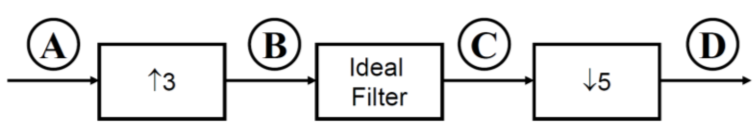
- **You <u>must</u> submit your HW via submission to Brightspace (just click on the assignment and you'll be given an interface for uploading your HW)**
- **You <u>must</u> submit your HW in either PDF or doc/docx format!**
    - **Most scanners will give you the option of the PDF format**
        - **You can use a phone app to create a single PDF of your HW**
        - **The app called "Adobe Scan" works quite well**
    - **If you are using your phone to take pictures of each page…**
        - **Insert each photo into a word document (one per page, suitably filling the page!)**
        - **Then submit the word document or convert it to PDF**
- **Word gives an option to save as PDF**
    - **Submission of individual images will NOT be accepted!**
- **You <u>must</u>** put the problems into the submitted file in the order they are assigned below.

1. Assume that the DTFT of signal *x*[*n*] fills the entire frequency range of –π to π (assuming a triangular shape is a good idea here).

    a. Draw the spectra after each step in the following figure



    b. Draw the spectra after each step in the following figure

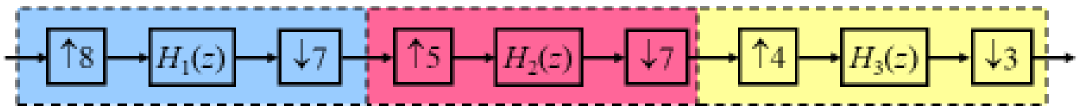


2. Re-do the multi-stage decimation problem on slides 8 – 12 of Note Set I-DSP-7 to compare the case where the order of the stages is reversed (e.g., re-do for the case where $M_1 = 4$ and $M_2 = 3$).

3. This problem explores the use of MATLAB's "resample" command to convert an audio file from a sampling rate of 44.1 kHz (CD rate) to 48 kHz (DAT rate).

    a. Determine how to modify the multi-stage approach shown on slide 15 of Note Set I-DSP-7 to enable conversion from sampling at 44.1 kHz to sampling at 48 kHz
        - Note that there are 6 permutations of the 3 up factors and 6 permutations of the 3 down factors, for a total of 36 combinations
        - Some of those combinations will result in intermediate sampling rates that are lower than 44.1 kHz and are therefore invalid
        - Of the valid combinations, we should strive to make the choice so as to minimize the largest sampling rate that occurs anywhere in the system
        - So, you could do the analysis to find the best choice of ups and downs with Excel, but I encourage you to instead write a MATLAB script that will check all the cases for you and extract the best selection
    b. Implement the resampling scheme selected in part (a) as follows:
        - Read-in the provided audio file guitar1.wav, which was recorded at a sampling rate of 44.1 kHz. Call that signal vector x_cd.
        - Use MATLAB's resample command to implement each of the stages you determined in part (b). (See information below about the resample command). Call the resulting signal vector x_dat
        - You can use the sound command to listen to the x_cd (using 44.1 kHz sampling rate) and to listen to the new signal x_dat (using 48 kHz sampling rate) and they should sound very similar if not identical.
        - After creating the appropriate time vectors t_cd and t_dat, do the following plot and comment on the result

            plot(t_cd,x_cd,'b-o',t_dat,x_dat,'r--x'); xlim([1 1.001])
        - Compute the DFT of the first 0.4 seconds of x_cd and the DFT of the first 0.4 seconds of x_dat
            a. Zero pad to a total of 2^{16} samples
            b. Plot each result in dB vs positive frequency in Hz on a single set of axes (use blue for the CD signal's DFT and red dashed for the DAT signal's DFT)
            c. Comment on the similarities and differences between the two results (zooming can be helpful here)
    c. Repeat part (b) but instead use the following scheme



What is wrong here?

Information on the resample Command for Problem #3

`y = resample(x,p,q)` resamples the input sequence, `x`, at `p/q` times the original sample rate.

`resample` applies an FIR lowpass filter to `x` to remove images and prevent aliasing. It compensates for the delay introduced by the filter.

The fact that it compensates for the delay of the filter allows one to easily compare the original and the resampled signals!

4. Consider the 3^{rd} order DT system shown on slide #12 of Note Set I-DSP-8.
   a. Derive the state space model result shown on slide #13
   b. Write a MATLAB function that takes as input the vectors **a** and **b** that hold the coefficients of a difference equation (any order), the input signal x, and the initial state vector and then uses the state model to recursively compute the state and the output, both of which are provided as outputs of the function.
   c. Test your function in part (b) by using it to compute the impulse response of the following system and check the result by comparing its result with that found using the MATLAB filter command

$$y[n]-0.9y[n-1]+0.81y[n-2]-0.792y[n-3]=x[n]$$