EECE 403/525 HW #1

Due 1/30/2023 @ 11:59PM

See also the provided following files posted on Bb:

- **tukey.m & drop.m (the latter is used by the former)**
- **HW_1_Signal.wav**
- **Big_Hall.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. This problem explores window choice, DFT length, and the STFT.
   Use MATLAB's audioread command to read in the audio signal provided in the file HW_1_Signal.wav as follows:

   ```
   >> [x,Fs]=audioread('HW_1_Signal.wav');   % signal is put into x and sampling rate into Fs
   ```

   a. Compute and plot the DFT of the full signal versus <u>positive</u> frequency in Hz
      i. Use a rectangular window
      ii. Zero-pad to a total of 2^16 points
   b. Compute and plot the STFT of the signal
      i. But…. Use dB instead of just abs as shown in the notes!
      ii. Explore different choices of
         - block length
         - window type
         - overlap
      iii. appropriately choose zero-padding in each case based on block size
   c. From the results in part (b), how would you explain in words the nature of this signal? How would you write an expression for it in the time domain?

d. Repeat part (a) using a hamming window and compare the results to part (a). Based on your insights from part (c) does this result make sense?

2. Suppose you have two N-pt blocks of real-valued signal data $x_1[n]$ and $x_2[n]$ and you wish to compute the DFT of each of them: $X_1[k]$ and $X_2[k]$. One method is to simply compute the FFT of each block. An alternative is to first form $x[n] = x_1[n] + jx_2[n]$ and compute the DFT of that signal to get $X[k]$ and then form the two desired DFTs using

$$X_1[k] = \frac{1}{2}\left[X[k] + X^*[N-k]\right]$$

$$X_2[k] = \frac{1}{j2}\left[X[k] - X^*[N-k]\right]$$

Assume that the standard radix-2 FFT algorithm is used in each method and compare the computational complexity of the two methods.

3. Consider the DT filter with transfer function given by

$$H(z) = \frac{\alpha + z^{-1}}{1 + \alpha z^{-1}}$$

a. <u>Analytically</u> find the magnitude of this filter.
   i. Numerically verify your analytical result by using the matlab commands “freqz” and “abs” to compute and plot the magnitude in dB vs positive values of Ω (normalized by π) for a few representative values of α.
b. <u>Analytically</u> find the phase of this filter and plot it vs positive values of Ω (normalized by π) for a few representative values of α.
   i. Use the matlab commands “freqz” and “angle” to compute the phase and graphically compare the result to your analytical result.
c. <u>Analytically</u> find the phase delay of this filter and plot it for a few representative values of α.
   i. Use the matlab command “phasedelay” to compute the phase delay and graphically compare the result to your analytical result.
d. <u>Analytically</u> (see bulleted comment below for some advice) find the group delay of this filter and plot it for a few representative values of α.
   i. Use the MATLAB command “grpdelay” to compute the group delay and graphically compare the result to your analytical result.
      - You may use wolframalpha.com to do the derivative for the analytical work. Open the web page and type “derivative” (without quotes) into the entry box and click on one of the examples that pops up… that should give you enough guidance to do it. Note that wolfram alpha often misses some trig identities so you should look to see if any obvious ones have been missed!
e. Use MATLAB to run a test to see the effect of the group and phase delay of this filter. Use the case of $\alpha = -0.8$ and apply a signal of the form

$$x[n] = e[n]\exp\left(j\Omega_o n\right)$$

where the envelope $e[n] \geq 0$ smoothly ramps up to 1 then stays at 1 for some time and then smoothly ramps down to 0. The use of a complex sinusoid here makes it easy to extract the envelope $e[n]$ from $x[n]$: $|x[n]| = |e[n]|\,|\exp(j\Omega_o n)| = e[n]$. Use the following MATLAB command to compute the signal $x[n]$:

```
>> x=[zeros(1,100) tukey(1000,0.8)  zeros(1,100)].*exp(j*(pi*0.05)*(0:1199));
```

where "tukey" is a specific function that is provided.

Then use MATLAB to filter this signal using the above filter with $\alpha = -0.8$ to create the signal $y[n]$.

You can compute the envelope of the input and output using abs(x) and abs(y). Plot abs(y) and abs(x) and verify that the envelope is delayed by approximately the correct group delay for this case.

Plot real(x) and real(y) and verify that the carrier is delayed by approximately the correct phase delay for this case.

4. This problem explores the use of an exponentially swept sinusoid to measure the impulse response and frequency response of a room.

   a. Write a MATLAB function that does the following
      i. Input variables are f1 (start frequency in Hz), f2 (stop frequency in Hz), T (signal duration in seconds) and Fs (sampling rate in Hz)
      ii. Output variables are x (sweep signal to put into system to measure) and x_inv (the "inverse filter" signal to be convolved with the measured output of the system
   b. Compute and plot the spectrogram of the two signals created using your function with f1 = 10 Hz, f2 = 23 kHz, T = 2 seconds, and Fs = 48 kHz.
      i. To make the frequency axis logarithmic you can run the following command after plotting the spectrogram:
         - set(gca,'YScale','log')
   c. Write a MATLAB script that does the following to test the use of your sweep signals for measuring a room response. Use the parameters from part (b).
      i. Read in the signal Big_Hall.wav using the audioread command
         - He result will have two columns – indicating that this is a stereo impulse response. Use the first column only and discard the second one.
      ii. Convolve your sweep signal with the RIR provided in Big_Hall.wav to simulate the measurement of the response of a room to your sweep signal
      iii. Apply your "inverse filter" to the measured response
      iv. Show that the result provides a good estimate of the given RIR