ECE446F Sensory Communication
Digital Audio Signal Processing Problems

These problems require you to have access to MATLAB which is available for free download for all U of T students. It is also available on many of the ECF and ECE undergraduate computer lab PC's (although you will need sound, so it may be difficult to operate remotely). To record sounds, you can use Voice Recorder (Windows OS), Quicktime Player (MacOS), Audacity (free download) or by other means (with a cellphone, for example) and load that file into MATLAB using the `audioread` *command.*

1 Background Preparation

Following the course notes, learn how to generate a pure tone at a given frequency by making a time variable, followed by a sinusoidal function at a fixed frequency. You will need to choose the sampling rate and the length of signal, which will affect your time variable. Find out how to listen to this signal. Learn also how to use the Fast Fourier transform to find the frequency content of your signal. Understand how the length of the vector in the frequency space depends on the length of your time domain signal. If you have, say, a signal of 1000 Hz of length one second vs two seconds, notice how your interpretation of the frequency domain representation changes. Learn also how to use the `audioread` command to load audio signals into MATLAB and extract the sampling rate, as well as the limitation that PCM encoded audio signals must lie within ± 1. There is a corresponding command `audiowrite` to save files. Searching for help through Google is a good place to learn about the syntax for the various MATLAB commands.

2 Format of Deliverable

To get full marks for your assignment deliverables, you must do the following:

- Create **one** document (in PDF format) containing your answers to the questions (if a question is asked of you) and including any image files that are asked of you.
- Create a folder with subdirectories labelled '1', '2', '3', etc. In each subdirectory, place the sound files you were asked to generate *for the associated question*. You can save your file using `audiowrite`. When uploading this folder, compress it with zip before uploading.

3 Problems

1. **(2 marks)** Manipulation of signals in the frequency domain. You will generate a signal by directly manipulating the frequency domain. With sampling rate of `Fs` = `44100`, create a sound of 3 seconds in length. To do this, you will insert two complex phasors each with magnitude 0.5, 90 degrees out of phase with each other, at frequencies 440 Hz and 660 Hz. Do this in the frequency domain. Ensure that your signal is real before taking the `ifft` to get the time domain signal. Save your signal.

2. **(1 mark)** Create a 10 second sinusoidal signal at frequency 1000 Hz in the time domain with sampling rate 44.1 kHz. Call this variable `x1`. Next, create an identical signal in `x2`, but set the values from 1-9 seconds equal to zero. Thus, `x2` will consist of a 1000 Hz sinusoidal tone playing twice, once from 0-1 s and the other at 9-10 s. In your report, describe the differences between `abs(fft(x1))` and `abs(fft(x2))`, and explain this difference in terms of the time-bandwidth product relationship. Your answer should be brief and no longer than 50 words.

3. **(3 marks)** The following system known as the Dual Tone Multiple Frequencies (DTMF) is used to generate touch tone sounds:

		Upper Band		
		1209 Hz	1336 Hz	1477 Hz
Lower Band	697 Hz	1	2	3
	770 Hz	4	5	6
	852 Hz	7	8	9
	941 Hz	*	0	#

   Generate the tones for '1', '2' and '4' for a one second duration using Fs=10 kHz. Save your signals.

4. **(2 marks)** Following Q3. You must now design the telephone exchange algorithm to decode the DTMF tones. How would you do it? Using only the `fft` function, develop a simple algorithm to tell whether '1' or '2' or '4' was pressed given an unlabelled one second audio signal. Include the Matlab algorithm code in your report. (Keep your code simple. This is not a hard task.)

5. **(2 marks)** Study the slides from lecture DA 1. Generate a continuous dial tone for 5 seconds with Fs=10 kHz. Generate a busy signal for 10 seconds with Fs=10 kHz. Save your signals.

6. **(4 marks)** Find an empty bottle (could be water, beer or wine bottle) and record its associated Helmholtz resonance frequency by blowing across the top of the bottle. Analyze the frequency spectrum of the sound file you recorded using `fft` and compare this to the predictions from Helmholtz's equation given in the course notes. Repeat with the bottle half filled with water. Put your numerical analysis into the report (including any plots of the power spectrum) and save both recorded audio signals. *Also include a picture of the bottle in your report.*

7. **(3 marks)** The human voice and any other musical instrument can both "play" the same note, yet sound entirely distinct from each other. This is the concept of timbre. You will

generate a 440 Hz (middle A) reference tone, listen to that tone, and then record yourself singing this tone. Next, you will generate and record the same tone using a piano or guitar or violin. If you do not have access to a musical instrument, you can download a pre-recorded violin 440 Hz tone from the course repository (`a440.m4a`). Analyze the spectrum of both the voice as well as the instrument. Notice the location of the peaks and the relative magnitudes of the peak. What do you see in common, what is different between the voice and the musical instrument? Your answer, to be put in the report, does not need to be long ($<$ 50 words). Save both sound files (your voice and the instrument sound, if you have an instrument).

8. **(3 marks)** Using an 16 kHz sampling frequency, create a constant amplitude chirp of one second in length that starts at 500 Hz and increases linearly to 7.5 kHz. Repeat with an 8 kHz sampling frequency. Listen to both signals. Look up the concept of *aliasing* and answer the following question ($<$ 50 words): Why does the frequency of the chirp appear to decrease for the second half of the signal? Include your answer in the report and save both signals.

9. **(2 marks)** Manually create a spectrogram. Using the chirp signal from Q8, divide the signals into non-overlapping segments of 256 samples each. For each segment, take a 256 point FFT and store the positive frequency part of the spectrum into a vector of length 128 points. Do this for all segments, and put all of the resulting 128 point vectors into one large matrix of size 128 x (number of segments). Use the `imagesc` function to plot the values inside this matrix. Apply the spectrogram to both chirp signals from Q8. Save both images.

10. **(3 marks)** Download the file `doppler.m4a` from the ECE446 file repository. This sound was generated by a Learjet during takeoff. Analyze the signal using the following the following command

```
spectrogram(x,256*20,256*15,256*20,Fs,'yaxis')
```

where `x` is the audio signal and `Fs` the sampling rate of the sound. For an object moving with constant velocity, the apparent frequency $f(t)$ of a pure tone source as measured by a stationary observer is given by the expression

$$f(t) = f_0 \frac{c}{c + v \sin\left[\arctan\left(\frac{vt}{d}\right)\right]} \tag{1}$$

where f_0 is the frequency of the source when it is stationary, v the velocity of the source, d the closest distance by which the source passes the microphone at $t = 0$ and c the speed of sound (Lubyako et al, "Understanding the Doppler effect by analysing spectrograms of the sound of a passing vehicle," *Physics Education*, vol. 52, pp. 1-8, 2017).

Focus on the most prominent frequency shift located between 5-9 kHz. Find the values of f_0, d and v which gives the best fit to this shift as observed in the spectrogram. Include these values in your report. Also include a plot of the spectrogram you generated, as well as a plot of the equation using the parameters you found. How does the speed of your calculation compare with the takeoff velocity? Note that the takeoff velocity of a Learjet is approximately 140-150 knots (1 knot = 0.514 m/s). What major assumptions are made when using this equation to compare with the recorded sound file? There is at least one. Include this answer with your report.

11. **(3 marks)** Generate white, pink and brown noise using the algorithm provided in the class notes. Use a 10 kHz sampling rate and generate 5 second length signals. Save your signals. Plot both the time domain signal as well as the power spectrum (the power spectrum is the square of the amplitude). For the power spectrum, use a log-log plot. Include the graphs in your report. Notice that white noise varies rapidly over time; whereas, pink and brown noises vary much slower. Knowing what you know about the power spectrum, explain why the time variations of noise depends on its 'colour'.

12. **(1 mark)** White noise can be generated even more simply using the random number generator. The command `x = rand(1,n)` generates numbers distributed between $[0, 1]$. What transformation to `x` must you apply in order to ensure that it is a proper PCM encoded acoustic signal at maximum amplitude? Include your answer in the report.