Digital Signal Processing

Exercise 6

The exercise is concerned with the Discrete Fourier Transform (DFT) and its application. The exercise should be solved using Matlab. You should submit a zip file with your Matlab code.

You can submit the exercise in pairs.

You should also explain why your code works, as comments in the header of the function (so it can be seen using the help command in Matlab).

The file myUname.m should be updated to include your user name and ID.

1. In this question you should write a function for identifying tone phone numbers. In tone phone (see figure 1), each number is the sum of two pure sines - one for its column and one for its row. The frequencies (in HZ) for the rows are:

f_r = [697 770 852 941]

and the frequencies for the columns (in Hz) are:

f_c = [1209 1336 1477]

1	2	3
4	5	6
7	8	9
10	0	11

Figure 1: Tone phone. The * and # were switched with 10 and 11 in order to avoid dealing with strings in Matlab.

For example the signal that represents the number 8 is combined of two frequencies -
- The frequency that represents the second column
- The frequency that represents the third line

The sampled signal would look like this:

$$x_8[n] = sin(2\pi \frac{852}{F_S} n) + sin(2\pi \frac{1336}{F_s})$$

where F_s is the sampling rate.

(a) Download the file phone.zip from the course web site. Extract the files phone_x.wav where $x \in \{i\}_{i=0}^{11}$. Read the sound files using the Matlab function ***wavread***. Make sure you understand the API of the function (could be done using the command: help wavread).

(b) Explain how to use the DFT for identifying the number that a signal represents (You can use the function ***fft***, don't forget to perform ***abs*** and ***fftshift*** before you view the DFT).

(c) In real life signals, the line we use can be noisy, which can change our signal. In the files phone_x_noise.wav where $x \in \{i\}_{i=0}^{11}$ the same tones are played, but noise was added. The noise that was added is called "white noise" which is basically random noise at all frequencies. You can open the file noise_only.wav to look how this noise and its DFT looks.

(d) Write a Matlab function:

$$[number] = identifyPhoneTone(\, x \, , \, f_s \,)$$

The function gets the sampled noisy signal vector and the sampling rate of the signal, and output the number that was played.

2. In this question you will write a function that identifies what sequence was entered.

(a) Download the file phone_seq.zip from the course web site. Extract the files seq_x.wav where $x \in \{l\}_{l=1}^{10}$.

(b) We will define the following protocol: each tone will last for 0.1 [sec]. Between each tone there will be a period of no tone for at least 0.05 [sec].

-In the files seq_x.wav $x \in \{l\}_{l=1}^{4}$ you can find examples with no noise. This means that at the time of no sequence the signal is zero. In the rest of the files noise was added.

(c) Write a Matlab function:

$$[seq] = identifyPhoneSeq(filename)$$

The file receives a file name and returns the tone sequence that was in the file. In the file example.m you can find the correct results for all the examples.

3.

(a) What are the properties that a sampled signal x[n] should obtain, so that we could use the DFT transform on it ?

(b) Given a signal x(t), we'll want to analyze it using CTFT. Given that $\forall \omega \in [4.99, 5.01]$ $X^F(\omega) = 2$.
Can you bound the time segment in which the signal is affected by the frequencies $[4.99, 5.01]$?

* You don't need to submit the following question- Think of what we gain of using DFT instead of DTFT in 2C, and on what is the problem of doing so (We'll talk about that on the upcoming weeks)