CSE 3048 FINAL QUESTION 4

Here is the 4th question of your final exam.

Due date: 16.06.2023, @14:00

Submission address: kubra.uludag@marmara.edu.tr

Submission format: name_surname.pdf

In https://www.dropbox.com/scl/fi/l1ydoxrt5u6ssmw5b5ufx/List.xls?dl=0&rlkey=bqyv8179r7kgpe01ho21h36f6,

you will find a file containing the list of all students taking this lecture.

In this list, find your name. In front of your name, there will be the number assigned to you.

In the dropbox link https://www.dropbox.com/scl/fo/wflkjq6mwhy8ld9g9ow1g/h?dl=0&rlkey=hctxny5e0yohx0re1y6a9nsc7

You will see many small composite modulated signal files, numbered from 1 to 221. Download the file whose number is the number assigned to you. You will work on this file, which is generated by the third program listed in

http://www.marmaralectures.com/experiments-on-sound-recording-modulation-and-demodulation/

The file contains a composite modulated signal, which contains two three-second recordings of speech.

Your task is:

1) Find out which frequencies are used to modulate the individual components of the modulated signal.

2) Bandpass filter and demodulate the composite signal and find out what is said in each of them.

You have to hand in a report in .pdf format which contain two parts;

1) At the top of your report, you have to list the contents and modulation frequencies forming your composite signal, like:

modulation frequency 1: 4KHz   Message 1: Sweden

modulation frequency 2: 11KHz   Message 2: One Two Three

2) This should be followed by an explanation section, in which you have to briefly explain how you demodulated your signals. These explanations should be approximately the same ones given in

http://www.marmaralectures.com/experiments-on-sound-recording-modulation-and-demodulation/

but you have to use the figures derived from the composite signal you have downloaded.

Important issues:

1) In this project the sample rate (fs) is 96 kHz instead of 44.1 kHz.
2) When solving this problem, you will be needed to look at the same signal in time and frequency domains repeatedly. Let us denote time and frequency representations of the same signal as $x(t)$ and $X(\omega)$. Then the passage between this two representations is done by using fft and ifft functions of numpy: $X(\omega)$=fft($x(t)$), $x(t)$ =ifft($X(\omega)$). BUT, due to some cosmetic reasons, you have to use these two functions as $X(\omega)$=fftshift(fft($x(t)$)), $x(t)$=ifft(fftshift($X(\omega)$)). This will put the 0Hz frequency at the center of your plot.
3) You cannot simply plot a frequency domain signal as it has real and imaginary parts. Hence, when you print $X(\omega)$, you should print it like plt.plot(np.abs($X(\omega)$)).
4) Your time domain signals must be real, ie, they wont have an imaginary part. But, when you return back from frequency to time, ie $x(t)$ =ifft(fftshift($X(\omega)$)), numpy will consider your $x(t)$ signal as a complex quantity, and will add an imaginary part to it. If you have done everything correctly, this imaginary part will be zero and will not affect any of the operations. But, when you want to print $x(t)$, it will cause error messages, as complex signals are not printable. Therefore, you should print such signals as plt.plot(np.real($x(t)$)).