Third year communication

Lab 1: Amplitude Modulation

Using matlab program:

1- Choose any audio signal with 20sec length.

i) Read the audio signal in matlab and then plot the signal waveform in time domain, the signal amplitude and phase in the frequency domain.

2- Perform DSB-LC modulation

$$y(t) = (x(t) + A)\cos \omega t$$

ii) Choose reasonable values for A and ω to achieve DSB-LC modulation with your previous audio signal and explain how you choose them?

iii) Plot the modulated signal waveform in time domain and the modulated signal amplitude and phase in frequency domain.

iv) What do you think is a carrier's minimum Amplitude (A) to avoid over modulation? What is the problem with the AM signal when it is over-modulated?

v) Compare between the bandwidth of the audio signal and the modulated one by plotting both signal in the frequency domain.

3- Perform DSB-LC demodulation

i) Do synchronous demodulation to obtain $x(t)$, then plot the final signal in time and frequency domain as previous.

ii) Hear the demodulated signal and compare it with the original one.

Are the two signals the same? Explain why?

- **Important note: For any two copied projects the grade is zero.**
- **Deadline: 13/11/2022**
- **The lab is individual.**
- **Submission is one pdf file containing simulation results, comments and codes at the end of the file.**

General notes for matlab code

- You can read the audio signal in MATLAB using the command 'audioread' or 'wavread'. For later processing, you will need to obtain the sampling frequency (F_s) (which was used to digitize the recorded waves). The MATLAB function can return the sampling frequency.
- Higher carrier frequency in order of megahertz will increase the simulation complexity.
- If you want to modulate a signal, you shall generate the carrier: cos($\omega_c t$) which, in the discrete domain (in MATLAB environment), is generated as cos($\omega_c n \mathrm{T}_s$) where T_s is the sampling interval. This could be T_s =1/F_s , F_s the sampling frequency obtain from the 'wavread' command. $\mathrm{F}_c = \frac{\omega_c}{2\pi}$ must be less than $\frac{\mathrm{F}_s}{2}$ to achieve Nyquist criteria for sampling.

- You can plot the spectrum of the signal using the 'fft' and 'plot' commands. You will have to adjust the axis scale so the signal spectrum is plotted versus frequency. By doing so, you can estimate the bandwidth of the signal. You will use this bandwidth value when you design the filter. You may find the following MATLAB tutorial useful:

  http://www.ee.nmt.edu/~elosery/matlab/matlab.pdf

- For The design of LPF, You can browse MATLAB Help for the command 'fdesign.lowpass'.