Communications Project

The objective of this mini-project is to simulate AM (DSB-LC) and FM, and compare their performance under additive noise using Matlab.

You may either use a Matlab sample audio file (e.g. “SpeechDFT-16-8-mono-5secs.wav”) or use a short audio file of your choice. You can read the file and obtain its sampling rate using ‘audioread’ function.

Amplitude Modulation

1. Perform DSB-LC modulation on the audio file. You may use a carrier frequency of 100 kHz. Use a modulation index of 0.9. Use the AM equations discussed in lectures.
2. Add random noise to the modulated signal. You may use ‘awgn’ function.
3. Demodulate the signal. You may use ‘envelope’ function. A blocking capacitor can be simulated by subtracting the mean from the envelope detector output.
4. Listen to the demodulated signal using ‘sound’ function.
5. Repeat the above for different values of signal-to-noise ratio (SNR), say from 0 to 20dB.

Frequency Modulation

1. Perform FM modulation on the audio file. You may use a carrier frequency of 100 kHz. Use a deviation ratio $\beta = 5$. You may use ‘cumsum’ function to obtain the instantaneous phase deviation.
2. Add random noise to the modulated signal.
3. Demodulate the signal. You may use ‘fmdemod’ function.
4. Listen to the demodulated signal.
5. Repeat the above for different values of signal-to-noise ratio (SNR), say from 0 to 20dB.

Discussion:

1. Explain your code briefly.
2. Discuss how you selected the sampling rate of the carrier and modulated signal.
3. Discuss how you set the modulation index to 0.9 in AM.
4. Discuss how you set the deviation ratio to 5 in FM (WBFM).
5. What do you notice on the demodulated audio as SNR increases in both cases?
6. What do you notice on the demodulated audio for small values of β, say 0.1 (NBFM).
7. Plot a graph comparing the mean squared error (MSE), which is the average of the square of the error signal between the original audio and the demodulated audio for different values of SNR for both types of modulation. Comment on your results.

Hint:
Modulated signals are generally of very high frequencies, which typically require higher sampling rate than the baseband audio file, according to the Nyquist sampling theorem. So make sure that you select an appropriate sampling rate for the carrier. Accordingly, you may need to increase the audio file sampling rate, which may be accomplished using 'resample' or 'interp1'.

Group size: 1 to 2 students

1. PDF Project report (**not compressed**)
   - First page listing the names of the group members, sections, BN's.
   - Project discussion as mentioned above.
   - Full Matlab code.
2. One zipped file containing the Matlab code.

   Email subject: CMP3 – CommSystems project

Expect an empty reply confirmation email from the TA in 24 hours. If you don't receive any email in 24 hours, then please resend your project along with a screenshot from your email server that you sent the first email before the due date.

Discipline: Cheating will result in a grade of zero for both groups.

Project submission deadline: 19/12/ **2019, 11:59 pm Cairo time.**