CMPE 352

Homework 2

Due Date : 12.04.2019 17.00

1-) In the first part of the project, you will combine two sounds given which are **street.wav** and **mike.wav** using MATLAB. Then you will extract speech from noise by using filter. You have to convert the mixed sound to frequency domain. After that, find human voice spectra (frequency band). You will filter outside the human speech band. Then convert the sound to time domain and listen.

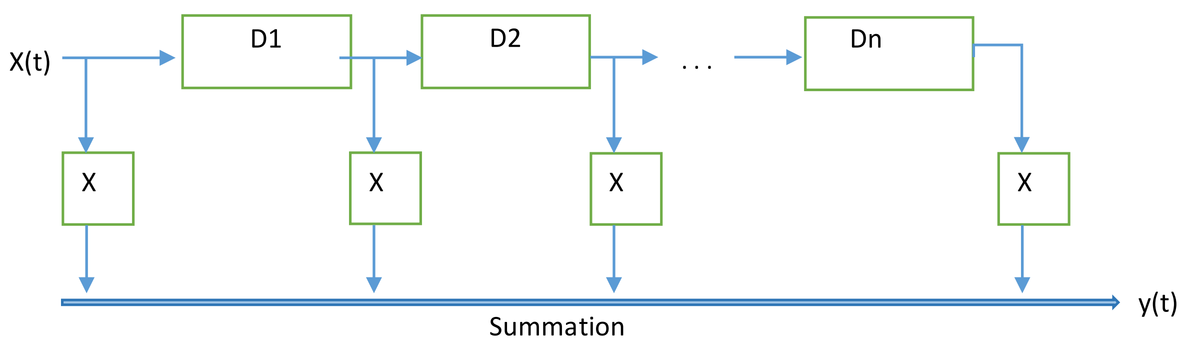
Both sounds' durations are the same and 9 sec . By this way, it is easier to compute matrices of sounds. Your script should be named **clearNoise.m**

Output: You will output four figures (Each includes several figures as below)

1- Frequency Domain Representation of mike.wav, street.wav, mike+street.wav
2- Time Domain Representation of mike.wav, street.wav, mike+street.wav
3- Frequency Domain Representation of mike.wav, filtered (mike+street).wav
4- Time Domain Representation of mike.wav, filtered (mike+street).wav

You will calculate the SNR value (see end of the document for computation detail) of recovered signal and original mike.wav.

2-) In this part, you will design N tap filter. N-tap filter is used for diminishing the effects of delayed versions of the sound. You are asked to write a program in MATLAB that combines mike.wav and mike.wav with delay K miliseconds. K is by default 100 miliseconds. You will use N-Tap filter below:



Delays (Ds) will be multiples of K. Second parameter of the function is n. You will change n from 1-50. In your report, explain the effect of n and K.

Output: You will output two figures .

1- Use constant K, change n from 1 to 50 and plot SNR of mike.wav and recovered signal.
2- Use constant n, change K between 100,200,300,400 miliseconds and plot SNR of mike.wav and recovered signal.

How to Calculate SNR of two audio files:

Signal-to-Noise Ratio (SNR) is used as an objective measure for the metric of imperceptibility. Signal to Noise Ratio (SNR) is a difference metric that is used to calculate the similarity between the original audio signal and the recovered audio signal. The SNR computation is carried out according to equation (1), where In is the original audio signal, and En corresponds to the recovered audio signal.

$$SNR(dB) = 10\log\frac{\sum_n {I_n}^2}{\sum_n (E_n - I_n)^2} \qquad (1)$$