EI 015 Signals and Systems -2022 spring

Course Project

Goal:

- To help students to understand and master the fundamental theories of the course.
- To improve their ability to apply the theories in engineering.

Software Requirements:

Tools: MATLAB (suggested), or others like Python and C/C++

General Policy:

- Please submit on **CANVAS** a **.zip** version of your solutions to the course project, which should contain at least the following three items:
  1) a **project report** written in **English** and in **.pdf** format;
  2) resulting **audio clips**;
  3) and your **source code** to generate the above audio clips.
- **NO LATE** submission is allowed, otherwise the grade will be discounted.

Project 1 (5 points)

1) Select a piece of audio (music or so on), denote it as $f(t)$, and show its signal waveform.
2) Generate $f(-t), f(2t), f(t/2)$, and show their waveforms.
3) Calculate the Fourier transform of $f(t), f(2t), f(t/2)$, show their spectra, compare and analyze them.
4) Denote $F(j\omega)$ as the Fourier transform of $f(t)$. Show the waveform of the inverse Fourier transform of the magnitude spectrum $\mathcal{F}^{-1}\{|F(j\omega)|\}$ and the waveform of the inverse Fourier transform of the phase spectrum $\mathcal{F}^{-1}\{e^{j\measuredangle F(j\omega)}\}$. Compare them with the original signal $f(t)$.
5) Implement a low-pass filtering (using ideal low-pass filter, cut-off frequency decided by yourself) to $f(t)$ in the frequency domain, and show the waveform of the resulting signal.

Project 2 (5 Points, please choose either one of the following two options)

Remark: you can use the audio in "audio example.zip", or collect any other audios by yourself.

Option 1: Voice Eliminator

1) Design a software or a simulation program, named "Voice Eliminator", to complete the elimination of the voice of the singer in the song.
2) Analyze the basic design ideas and principles of the proposed Voice Eliminator, and implement it through a chosen programming language.
3) Further use appropriate methods and measures to improve the Voice Eliminator's performance from both the theoretical and the practical points of view.

Option 2: Speech Sampler

1) Collect the voice of someone's speech as a continuous-time signal, sample it several times to obtain several discrete signals at different sampling frequencies. (Recommended sampling frequencies are 44 kHz, 22 kHz, 11 kHz, 5.5 kHz and 2.75 Hz.)
2) Reconstruct the continuous-time signal again from each discretized signal sampled at different frequency. Analyze the influence of the sampling frequency on the reconstruction quality, and calculate the reconstruction error.
3) Analyze the basic design ideas and principles of the proposed Speech Sampler, and implement it through a chosen programming language.