Signals and Systems – 2022 Spring

Course Project

Due: May 27th, 2022, at 8:00 pm
Assigned: April 29th, 2022

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/Chinese** and in **.pdf** format;
  2) resulting **audio clips**;
  3) and your **source code** to generate the above audio clips.
- **No less than 30%** notes in the source code
- **NO LATE** submission is allowed, otherwise the grade will be discounted.

Project 1 (5 points)

1) Select a clip of audio (music or so on), denote it as $x[n]$. Use an interpolation method (e.g., zero order hold, first order hold, etc.) to construct the continuous time counterpart $x(t)$, and show its signal waveform.
2) Generate $x(-t)$, $x(2t)$ and $x(t/2)$, respectively, and show their waveforms.
3) Calculate the Fourier transform of $x(-t)$, $x(2t)$ and $x(t/2)$, respectively. Show their spectra, compare and analyze them.
4) Denote $X(j\omega)$ as the Fourier transform of $x(t)$. Show the waveform of the inverse Fourier transform of the magnitude spectrum $\mathcal{F}^{-1}\{|X(j\omega)|\}$ and the waveform of the inverse Fourier transform of the phase spectrum $\mathcal{F}^{-1}\{e^{j\measuredangle X(j\omega)}\}$. Compare them with the original signal $x(t)$.
5) Implement a low-pass filtering (using ideal low-pass filter, cut-off frequency decided by yourself) to $x(t)$ in the frequency domain, and show the waveform and spectrum of the resulting signal.

Project 2 (5 Points, please choose one from the following three options)

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 kHz.)
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.

Option 3: Other Topic

Feel free to choose any other topics of interest. If that is the case, please contact with the teacher to get confirmed first.