Homework 2

Please prefer readable tidy code using good pronouncable names over optimized code with bad names, needing additional explanatory comments.

Preparation

Please read aloud the following sentence and record it with your laptop, tablet or phone as `sentence.wav`.

> "There's a shortcut over that hill to my house. Some local people say the hill is haunted. No one likes to pass through those fields after dark."

Please also record the vowels [ɑː] "uh" as in "dark", [oʊ] "o" as in "over", [aʊ] "ou" as in "house", as well as the single sounds [ʃ] "sh" , [f] "f", and [kʰ] "k" as separate files.

Exercise 1 Representation in the time domain (3 Points)

Please start Audacity and open the recorded file `sentence.wav`.

- Which sounds are voiced, which are unvoiced? How can you distinguish them visually?
- Determine the sounds "uh", "o", "ou", "sh", "f", and "k" in `sentence.wav` and compare them with the voice records of your individually recorded sounds. Why do they look different? How is this effect called?

Exercise 2 LTI systems (8 Points)

a) Which of the following are LTI-systems and which are not? Please explain your answer. **(2 points)**

- Two people talking to each other using the **air** as a medium for their speech signals.
- A **loud speaker.**
- A **dam** and the water it holds.
- **A high pass filter.**
- The human **vocal tract.**
- The **position of the sun** seen from a fixed point on the earth.

b) **Coding Exercise.** The lecture script specifies the output of an LTI system to be computed by convolving the input with its impulse response. Now it's your turn to try it out. Please implement the function `compute_echo_signal()` in `specom_homework_2.py` which returns the signal with added echo using following steps. **(4 points)**

```
compute_echo_signal(signal: numpy.ndarray, sample_rate: float, echo_delay_seconds: float) -> numpy.ndarray
```

i. Convert the delay from milliseconds into an interval size of samples.

ii. Create an array as impulse response which starts with 1 at time 0, is 0.5 after `echo_delay_seconds` and is 0.2 at twice the `echo_delay_seconds`. In between, it should be zero.

$$\text{impulse\_response}(t) = \begin{cases} 1 & \text{if } t{=}0 \\ 0.5 & \text{if } t{=} \text{ echo\_delay\_seconds} \\ 0.2 & \text{if } t{=}2\cdot \text{ echo\_delay\_seconds} \\ 0 & \text{else} \end{cases}$$

iii. This exercise treats an echo like an overlapping repetition of the input with decreasing loudness. Therefore, after 100 ms, the original signal repeats with 0.5 times the amplitude, and after 200 ms it repeats with 0.2 times the amplitude.

iv. Convolve the impulse response with your input signal. Tip: you may use `numpy.convolve` for this job.

v. Apply normalization to the result which scales the signal values to the range −1 and 1 while keeping the proportion between the signal samples. You don't need to center the signal.

Finally, test your function with audio signal from the wav-file `13ZZ637A.wav` and a delay of 100 ms. Play your echo sound with `sounddevice.play(…)`.

Additional Questions (2 points)

1) Is the previous normalization process after the convolution an LTI system? Why or why not?
2) We are normalizing the result to avoid sample values above 1 or below -1. What is the reason that the result of a convolution can be above 1 or below −1 if both convolved signals are already bounded between −1 and 1?

Exercise 3 Convolution and Fourier transform (4 Points)

a) What is the result of the convolution of the two signals V1 and V2? Please solve this task graphically by drawing a plot of the result.

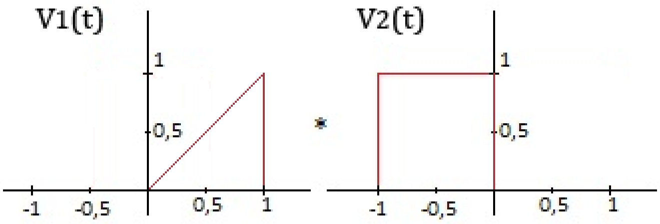

Figure 1: Two signal definitions for the task

b) Fourier transform

- What does the Fourier transform achieve?
- What is the difference between the Fourier transform and the Fourier series?
- If $s(t) = a \cdot \sqcap\left(\frac{t}{T}\right)$ is a **centered** rectangle signal with a time duration $T > 0$ and height a, then $\mathcal{F}[s(t)] = S(j2\pi f) = aT \cdot \text{sinc}(fT)$ is the corresponding Fourier spectrum.
  Note: $\text{sinc}(f) = \frac{\sin(\pi f)}{\pi f} = \text{si}\left(\frac{\omega}{2}\right)$.

  What is the corresponding Fourier transform of the time-shifted rectangle signal $s(t - t_1)$?