Lecture "Digital Signal Processing"

Prof. Dr. D. Klakow, summer term 2019

Tutorial 3

Submission deadline: 20.05.2019, 10:15

Submission Instructions:
You have one week to solve the tutorials.
The code should be well structured and documented. Do not use any Matlab-Toolbox or Python external libraries if not mentioned that you could use it.

- You are allowed to hand in your solutions in groups of two students.
- The theoretical part should be submitted before the lecture.
- For the practical tasks please submit files via the email address
  Tutorial 1: dsp.tutorial1@gmail.com Tutorial 2: dsp.tutorial2@gmail.com
- The subject of the letter should be [**DSP TUTORIAL X**]. **X** is the tutorial/assignment number.
- Rename and pack the main directory:
  Ex03_matriculationnumber1_matriculationnumber2.zip.

The directory that you pack and submit should contain the following files:

- code files and supporting files (library, image and sound etc.);
- file "answers.pdf" which contains answers to the questions appearing in the exercise sheet;
- file "README" that contains an information on all team members:
  name
  matriculation number
  email address.
- Note: If you use Jupyter Notebook, you don't have to submit "answers.pdf".You can write your theoretical answer in the markdown area

1 (6P)Exercise

In this exercise you will implement the *linear predictive coding* (LPC) algorithm. In the lecture, the objective function of LPC was solved and concluded with a linear systems of equations.

$$\Phi a = \varphi \tag{1}$$

Note: Do not use LPC built in function

1.1 (2P) Subtask

Implement two functions which computes Φ and φ and return them. You must follow the derivation that was described in the lecture to implement these functions. Both of the function takes two arguments, signal vector and order(P) of LPC model. Name the functions $cap\_phi(signal, order)$ and $sml\_phi(signal, order)$ respectively.

1.2 (1 P) Subtask

Write a function that returns coefficients(a) of LPC. The input arguments of the function are signal vector and order of LPC. Inside this function you will use the previously created two functions, and your favorite linear systems of equations solver(Use any built in method from Matlab of Numpy) to find the coefficients. Finally return that coefficients vector. Name the function $solve\_coef(signal, order)$

1.3 (1P) Subtask

Record audios of "a","m","l" and read them as vector(use sampling rate 48000, single channel and duration 1 sec). only store 40000 sample from the middle of the signal and store them as wav file. Play the wav files to check the recordings. Calculate their LPC the coefficients(use order 10). Note: Include the wav files in submission for checking. Use "conda install -c conda-forge python-sounddevice" to install sound recording library in your DSP conda environment.

1.4 (1P) Subtask

Write a function that take a portion of signal and and predict the next value of that signal. and return that value(Use portion of signal and coefficient vector as input argument).

1.5 (1 P) Subtask

Use the saved signal and calculated coefficients to find the error signal. You can write this step as a function. And apply this on the three recorded audio.

2 (4P)Exercise

In this exercise you will re-construct your original signal from a portion of original signal, the coefficients and the error signal.

2.1 (1 P) Subtask

Implement the following equation as a function and name it *reconstruct_n()*.

$$\hat{x}(n) = \sum_{k=1}^{P} a_k x(n-k) + e(n) \tag{2}$$

2.2 (3 P) Subtask

Use function to reconstruct the whole signal from the beginning portion of the signal (take first 10 samples of the original signal), coefficients, and error signal. Play the reconstructed signals Plot the original signal and the reconstructed signal in the same plot. For three audio there should be three plot. For all three audio files also plot the error signal as a function of time.