Assignment 1

Digital Signal Processing

Fourier Transform

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This assignment is about the Fourier Transform.

Form groups of three-four, work together and submit one report. Enter your team into the Wiki on moodle listing your names and matriculation numbers. Use the moodle forum to find team mates.

Record the voice of one member of your group as uncompressed WAV. Say one complete sentence. It can be any language. Make sure that you record at least at 44kHz or at a higher sampling rate and that the audio is not clipping. The full audio spectrum up to 20kHz needs to be available. Low quality MP3 downloads from websites are not allowed. Reports based on recordings at sampling rates below 44kHz or downloads will not be marked. If you do your research make sure to use publications describing full range studio/professional audio (not telephony or comms).

1. Load the audio samples into python.
   1. plot the audio signal in the time domain (linear axis: normalised amplitudes vs time) and
   2. in the frequency domain (logarithmic axis for both frequency and amplitude in dB) with proper axis labels.

2. Use a drawing program (Inkscape, Illustrator, drawio, ...) and mark
   1. the peaks in the spectrum which correspond to the fundamental frequencies of the vowels spoken.
   2. Mark up the frequency range which mainly contains the consonants up to the highest frequencies containing them.
   3. Mark up the whole speech spectrum containing the vowels, consonants and harmonics.

   Provide brief explanations. **20%**

3. Use the speech audio from the previous steps and determine the region of the highest harmonic voice frequencies in the spectrum and increase their frequency amplitudes with the help of the Fourier Transform with the goal to improve the voice quality. **30%**

4. Write a simple program which is able to detect at least two different vowels in your voice recording. Write a function which takes an audio file as an input and outputs the vowel as a string. Call the function voweldetector(wavfile) and call it twice (or more) in your program with two different wav files. **50%**.

The report should be brief, concentrating on the technical aspects and why you have done the different steps. Do not add generic theory about voice or Fourier Transform. Just describe the method and the result. Complete PYTHON code must be included in the appendix and submitted via moodle alongside the report. All figures inline in the report must be high quality graphics in vector format. Blurry jpeg figures or screenshots will not be marked. Submission must be PDF.

Upload your code, data/WAV files to moodle in form as a single zip file. Follow exactly the naming conventions for all files as specified on moodle. The scripts will be tested under Linux from the commandline (so not Spyder or Pycharm). Make sure your code is platform independent and does not contain absolute paths. Code that crashes will result in low marks. The same applies for code which won't display any plots. Your audio files must be original ones. No high level python signal processing / filtering commands are allowed except of the FFT and IFFT commands.

Deadline: 18th Oct, 3pm on moodle.