EEE 424 Coding Assignment 2 Spring 2024-25

Due-date: 16th May, 2025, 23:59

Q.1) [70 pts] Design a linear-phase bandpass filter of length $L = 10$. The filter must have a narrow passband such that for audio recordings sampled at 20 `kHz`, it must only pass frequencies within the specified **portion** of the human voice range. The portion's range is 300 `Hz` to 3 `kHz`. Write down its impulse response and $\mathcal{Z}$-Transform. Also plot the pole-zero diagram, impulse response, phase and magnitude response. Explain how close your design comes to the given specifications. Is it possible to do better than your design under the constraints imposed by the problem?

Q.2) [30 pts] Make a 10 second audio recording of your own speech. Depending on your hardware this recording should have been sampled at 44.1 `kHz` or 48 `kHz`. If it has been sampled at 44.1 `kHz` then change its sampling rate first to 48 `kHz` and then change it to 20 `kHz`. If it was sampled at 48 `kHz` then change its sampling rate to 20 `kHz` directly. To make these sampling rate changes utilize the down-sampling and up-sampling operations but do **NOT** use MATLAB's built-in '`resample`' function. To enable this process the non-integer factor has to be approximated by a rational number. Explain **CLEARLY** how you carried out the rate changing operation.

Now, apply the filter you designed in the previous question to the 20 `kHz` sampled signal. Comment on the output similar to the way you have done in Q1. Is the output coherent with your findings from the previous question?

Upload three audio files:

(a) The original voice recording sampled at the original sampling rate of 44.1 `kHz` or 48 `kHz`.

(b) The rate changed voice recording whose sampling rate should be 20 `kHz`.

(c) The filter's output also with a sampling rate of 20 `kHz`.

Also write down in your report what the size should be in *number of samples* for each file listed above (in single channel).

Note: To get an accurate 10 second recording simply record for slightly more than 10 seconds and then once you load it into MATLAB, trim it to the exact number of samples that a 10 second recording should have depending on the sampling rate that it has. The built-in '`audioread`' function should be preferred. Moreover, your original audio recording will also likely be dual channel (verify by checking its size). Convert it to single channel using the following command:

```
'monoAudio = mean(Audio, 2);'
```

After this change everything should be in single channel and no further modifications to the dimensions shall be necessary.