INDIVIDUAL ASSIGNMENT 1

Signals and linear systems in discrete time
22051

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In this assignment, I will see the "Peak at 1000Hz" as a peak at 500Hz, so that it is possible to recreate with my own voice

To start off, it is important to have a handy tool like MATLAB to analyze, visualize, and manipulate audio recordings. From the given plot, there are key information that will be used to recreate it:

- Plot axes - magnitude (in dB) and frequency (in Hz). This shows that this is a plot of the recording in the Fourier domain.
- Peak at 500Hz - this could mean that the frequency with the greatest magnitude in the recording is 500Hz.
- 2Hz between samples - indicates the spectral resolution. The spectral resolution can be used to determine the duration of the original sample.
- "standard" equipment - could mean that the recording is created with a "standard" sampling rate of 44100Hz (sampling rate: samples per second).

To recreate the audio sample a microphone, recording software, and (possibly) a quiet environment is required.

Before recording, it is nice to know for how long you should at least sing for, which is calculated by dividing 1 with the spectral resolution:

$$\frac{1}{2Hz} = 0.5s \tag{1}$$

When you have finished the voice recording at 500Hz, to be absolutely precise, you should only choose the amount of samples needed. This can be done by dividing the sampling frequency with the spectral resolution:

$$\frac{44100Hz}{2Hz} = 22050 \text{ samples} \tag{2}$$

In the plot, the magnitude of the recording is normalized (by dividing all the values with the maximum magnitude), which is going to make it peak at approximately 1 (makes comparison easier).

To see what this looks like in the time domain, you could plot the recording in the time domain, with a clean 500Hz sine wave to see how they correlate!

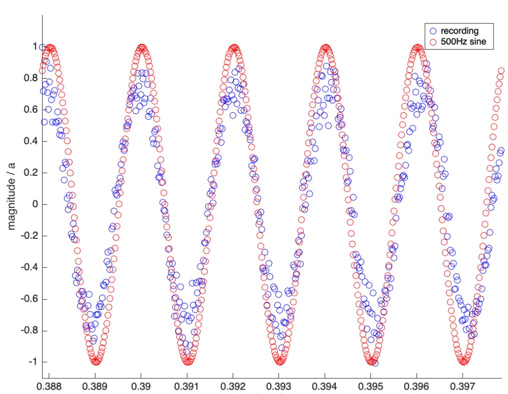

Figure 1: Time domain plot, comparing the recorded signal and a "clean" 1kHz sine wave over 5 periods

They look pretty similar! The most noticeable differences is the amplitude of the recording not being constant and that it is not a "clean" wave. This is possibly caused by clutter (microphone clarity, noise, etc.) and the human voice not being able to replicate a single frequency.

Now, both signals will be transferred to the Fourier domain - using the Fourier transform. Before plotting, the magnitude is calculated to dB, as decibels best describe how the human ear perceives sound.

The plotted Fourier domain shows the magnitude of the respective frequencies present from the two signals.

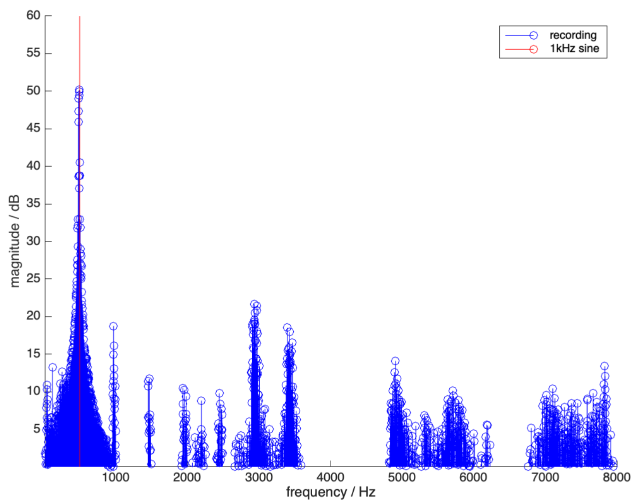

Figure 2: Fourier domain plot, the human vocal (hearing) range is between 20 & 8000Hz (0 and 130dB)

The plot clearly shows a peak at 500Hz - with other small peaks around the main peak. The different peaks could be the different frequencies that my voice have produced, when singing at 500Hz. Some peaks could be background noise/equipment clutter. This could be close to the original recording, but there is no way to be completely sure without comparing with the actual recording.

If you wanted to reproduce the signal through a speaker (i.e. headphones), it is important that the given speaker can replicate the discrete sound signal at the recorded sampling frequency (44100Hz).

I hope this gives you an approximate insight in the paper you gave me, but if there is anything else you can always ask!