Abstract

This report corresponds to the evaluation of the obtained results in the second laboratory of Applied Harmonic Analysis.

1 Removing noise

The aim of this laboratory was to eliminate any excess noise in an audio file. Initially, we read the audio file and played it using the *audioplayer* function in order to ensure that we were working with the correct audio file. Additionally, we have plotted the frequencies of the audio signal (Figure 1), allowing us to observe the evolution of the audio over time.

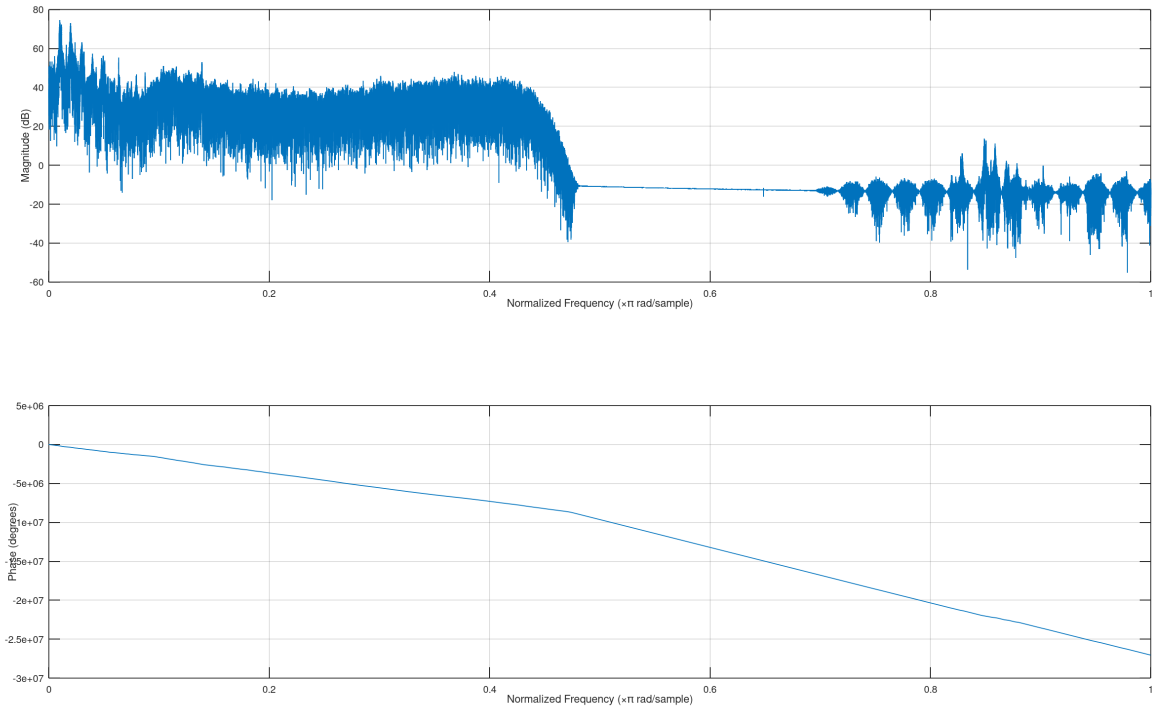

Figure 1: Frequencies of the original audio.

Afterwards, we will wait for the first audio to finish playing before adding any noise, and then we will use the same function as before *audioplayer*, to play the modified audio. Again, we have plotted the frequencies of the new audio signal, to detect where noise was added, see Figure 2.

Once we have detected the peak of frequencies near 0.2, we aim to eliminate it using the proposed filter. We are given the following transfer function:

$$H(z) = 1 - 2cos(\theta)z^{-1} + z^{-2}.$$

Since $H(z) = \sum_k h[k]z^{-k}$, we can easily obtain that $h[0] = h[2] = 1$, $h[1] = -2cos(\theta)$. Notice that, as stated in the problem, this filter will not pass pure sin waves at frequency $\omega = \theta$. Nevertheless, to catch exactly the value we are interested in, we need to normalize θ, i.e., divide it by the frequency at which it has been sampled, FS. Hence, we set θ to $35000/FS$, in order to remove the displayed peak. Passing this filter, we can observe how we have recovered the original audio, see Figure 3. Again, we wait for the noisy audio to finish playing before playing the filtered audio.

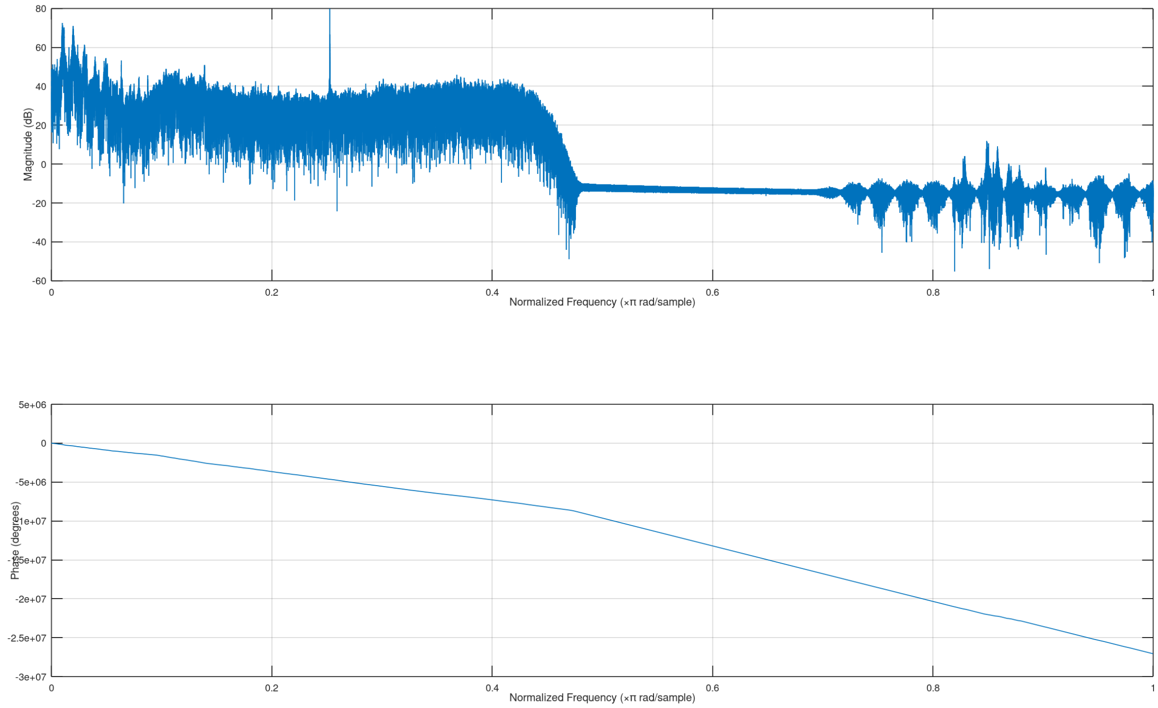

Figure 2: Frequencies of the noisy audio.

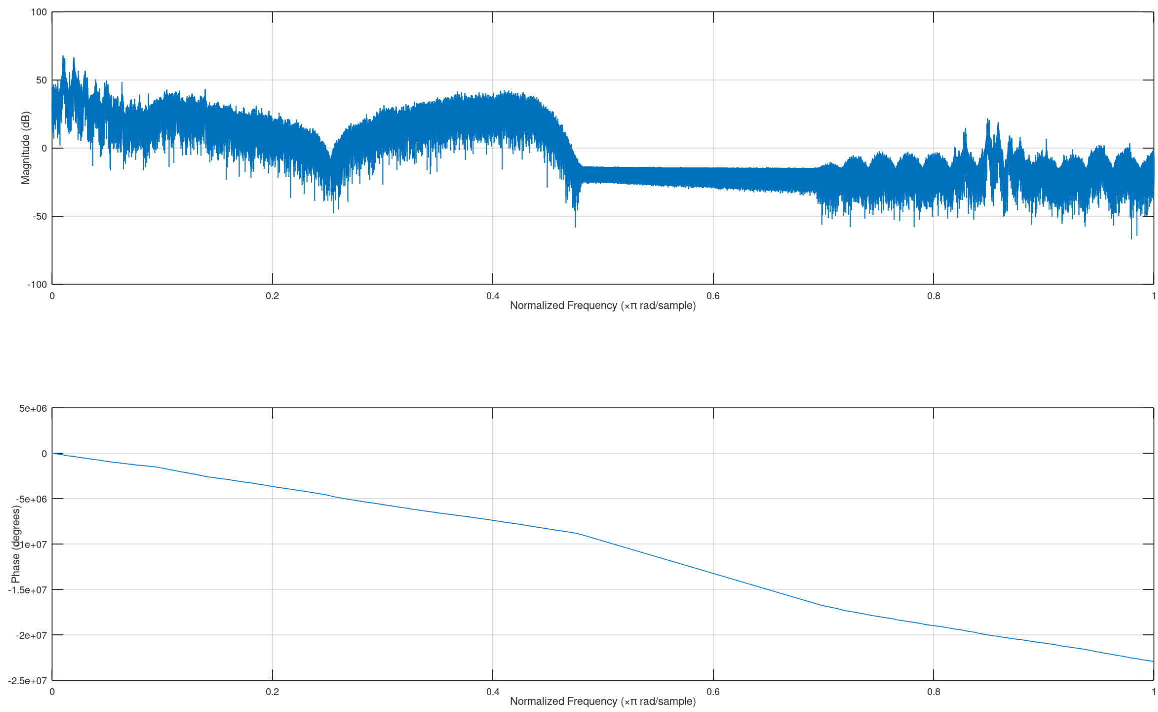

Figure 3: Frequencies of the filtered audio.

Finally, we are asked to determine the recursion formula for the filter used and plot the magnitude of the filter's frequency response $|H(e^{i\omega})|$ as a function of ω on the interval $-\pi < \omega < \pi$ where $\theta = \pi/6$. Let us start with the recursion formula. Notice that we have already computed it, it is implicitly in the code! Using the same notation as the statement, we have that:

$$a[0]y[n] = b[0]x[n] + b[1]x[n-1] + b[2]x[n-2]$$

where x is the original signal, $a[0] = 1$, $b[i] = h[i]$ for $i = 0, 1, 2$.

To finish, see Figure 4 to visualize the magnitude of the filter's frequency response for $\theta = \pi/6$.

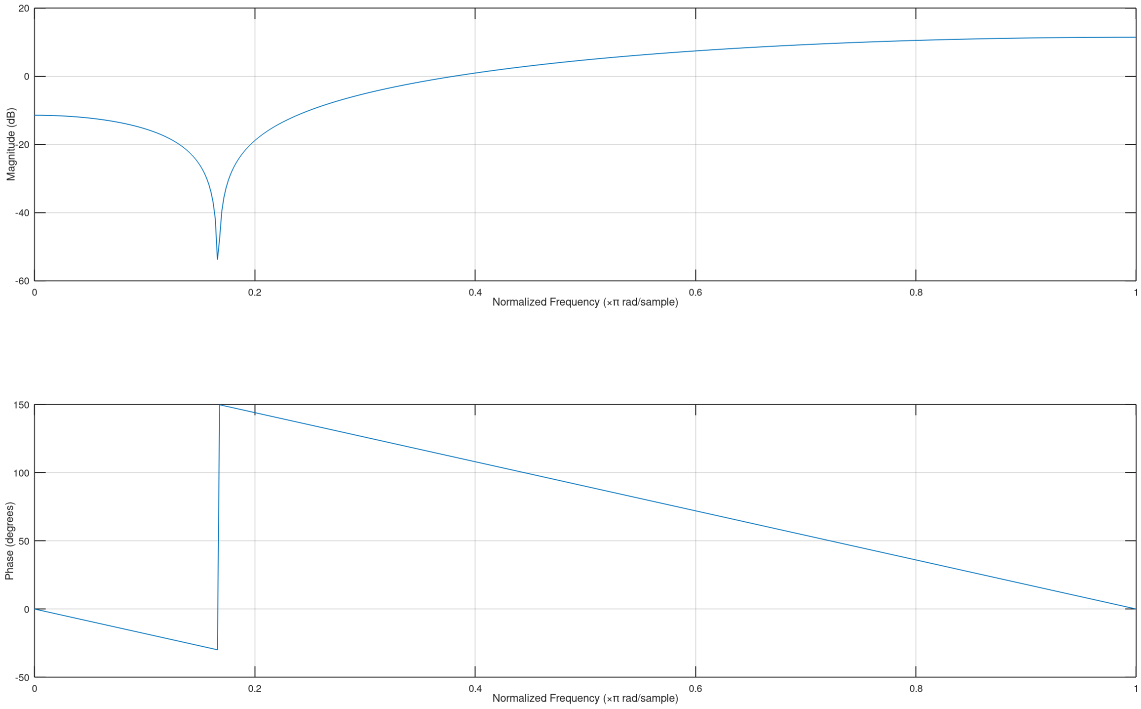

Figure 4: Magnitude of the filter's frequency response.