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Digital FIR Filter Design

Digital Signal Processing: Lab 4

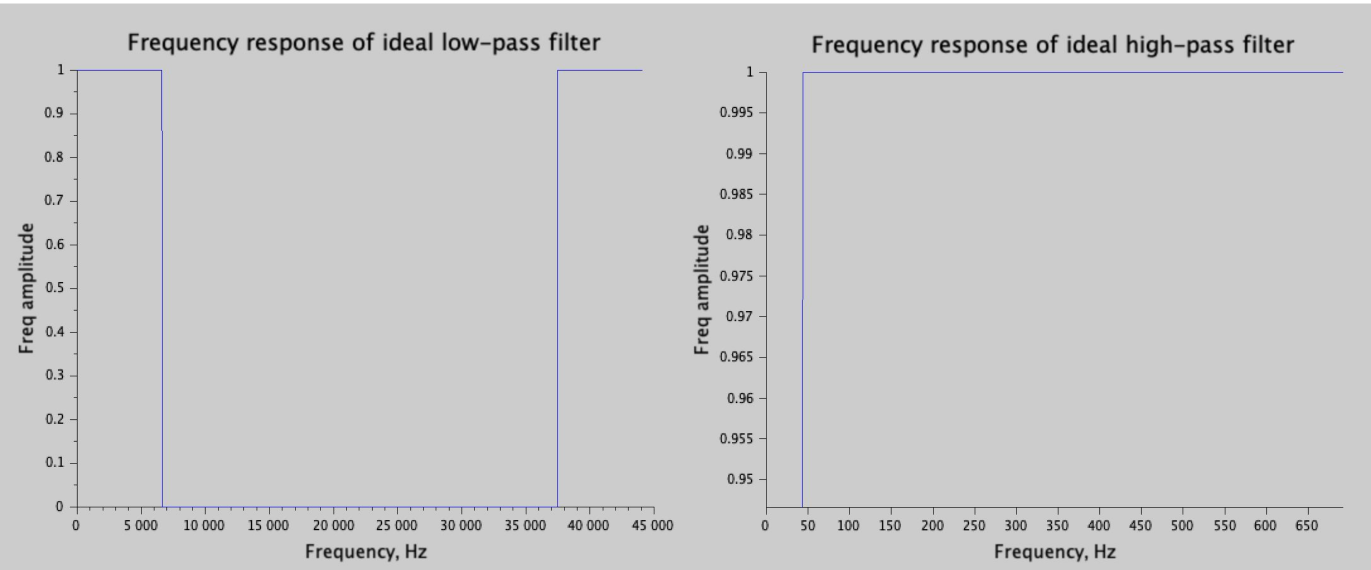
Task 1

Problem Description

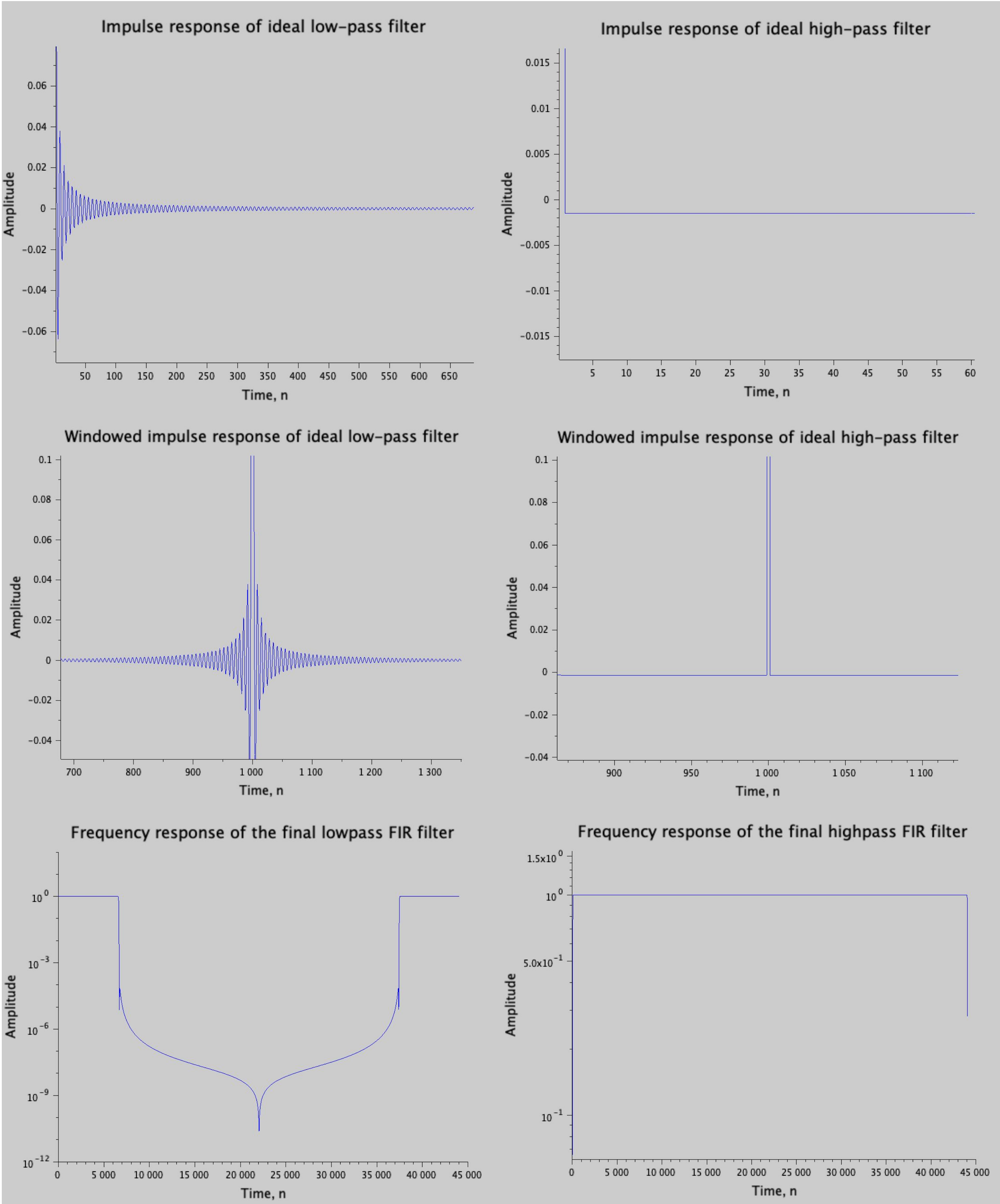
Design a FIR filter which is able to suppress high frequency (around 10KHz) and low frequency (10Hz and 20Hz) frequency noise components in order to apply it on the provided signal and obtain a cleaner one with clearly distinguishable human speech.

Solution

In order to achieve the goal we can apply **Low Pass** and **High Pass** filtering on the signal **sequentially**. This is close to application of Band Pass filter once.
Start with **ideal filter** models. As soon as we want to carefully suppress the low frequencies of the signal, having number of frequency discretization steps N=257 is simply not enough. If sampling frequency is 44100 Hz, one model step will be around 172 Hz. To solve this we need to set a bigger filter length, say N=2001, then one step will correspond to nearly 20Hz which is twice less than 40Hz, the values used further.
By setting cutoff frequencies to 40Hz and 6600HZ we obtain Frequency Responses:

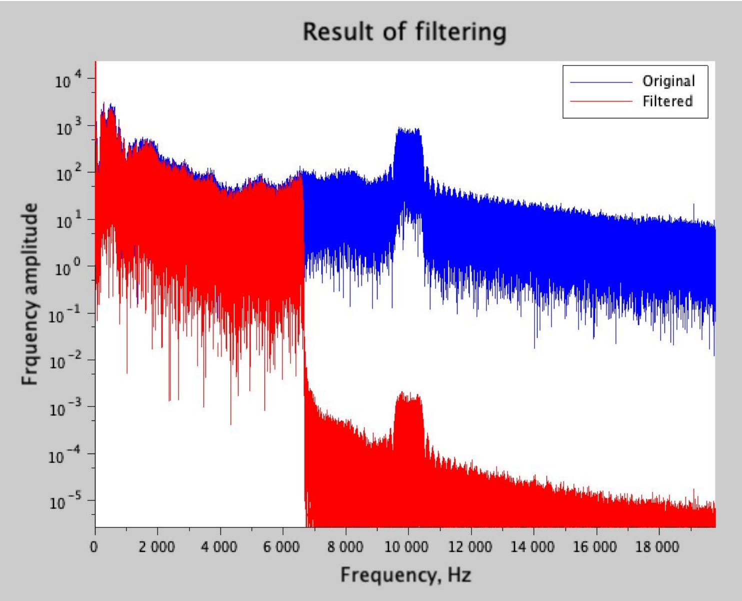


This filters are zero-phase and are not generally causal, we will introduce the linear phase by application of circular shift to Impulse Responses. Moreover, by looking to the impulse responses we can see that they never decay to zero, which also makes the filter impractical. To overcome this issue we apply Kaiser window function. After that, see the spectrums of the updated filters and notice the differences.



The next step is to sequentially convolve the original signal with Low Pass and High Pass filters and check the results.

Conclusion



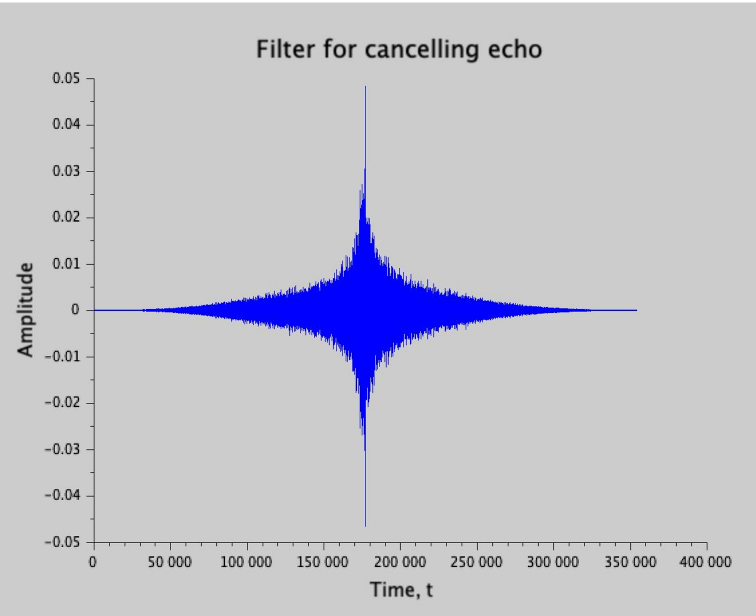
By looking (and hearing) to the results of filtering we can state we have achieved a much better quality of the audio with distinguishable human speech.
To conclude, we have reduced low and high frequency noises at 10Hz, 20Hz and 10KHz rates.

Task 2

Problem Description

Given signal and IRC, design an echo-cancellation filter **h** to obtain a signal close to original from results of convolution with IRC (echo applied). Verify the created filter by convolving it with the IRC.

Solution



First, according to the task, we obtain a filter by inverting the IRC in Frequency domain, that is

$$h = IDFT(\frac{1}{DFT(irc)})$$

This filter is still impractical like in the previous task. To overcome the issues and make it usable, apply circular shift and Kaiser window function. The obtained filter is depicted on the left.

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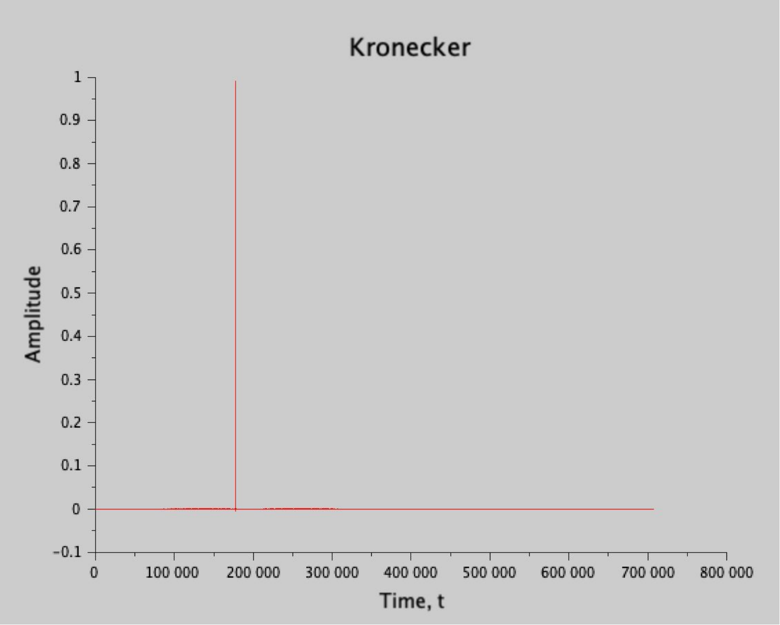
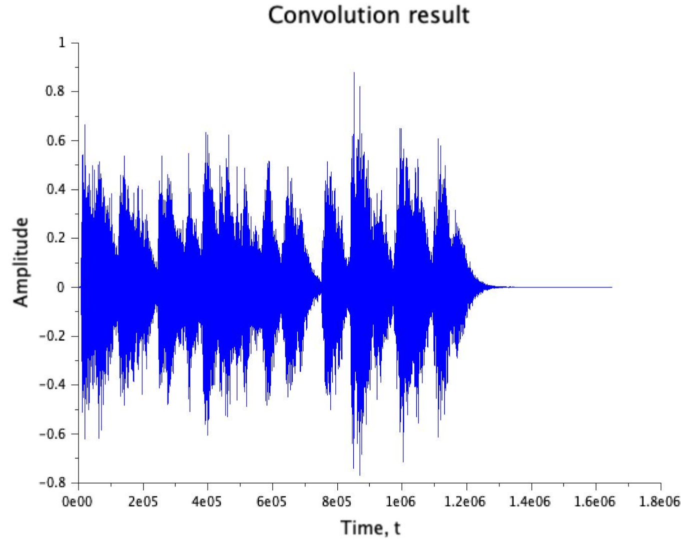
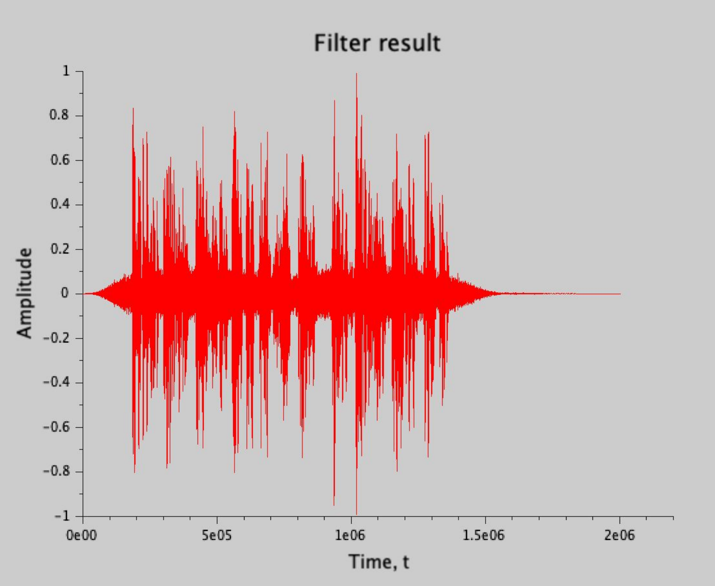


We chose Kaiser window function as it is considered nearly optimal in the trade-off between the main-lobe and side-lobe widths. To verify the filter **h** we convolve it with the original **IRC** and see that the obtained result is very close (almost) Kronecker delta function (shifted).

Conclusion





To conclude, let us see (and hear) the results we have achieved. Applying the filter we have designed we were able to almost cancel the echoing effect from the signal. We could not get the right original signal, because the filter (to make it practical) was modified and some information was lost. But we have tried to reduce this “side effect” by choosing the parameters of the filter (for example, window function).

Difficulties: I had operating system troubles installing and using the tools. -------------- *Thank you!*