**HOMEWORK 3 REPORT**

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I can recognize a filter if it is a lowpass, bandpass or highpass. But finding the appropriate coefficient of my use requires some calculations and wide thinking, which as far as I understand was not obligatory for the scope of this homework and due to my tight schedule I preferred to use the functions that give me an appropriate filter with specific requirements I gave to the function.

The documentation for the function I selected can be found from [here](https://www.mathworks.com/help/signal/ref/butter.html) while forming the filters (background theoretical equivalent of the filter method **butter** is Butterworth filters).

My filter is a digital filter and I drew the magnitude response diagram of each filter with the Fs function without normalizing and squeezing it to +-pi range.

Butterworth filter is known to have a sharp cutoff along with flat passband. It is an IIR which makes the algorithm computationally more efficient compared to equivalent FRI filters.

The proof and mathematical formulas along with it’s history in more detail can be found [here](https://en.wikipedia.org/wiki/Butterworth_filter).

A quality filter has a sharp transition at cutoff frequencies. This is closely related to Taylor series mentality and the same logic applies to Butterworth too: the wider the filter is, the better the filter outcome is.

A more elementary yet explanatory link can be found [here](https://www.electronics-tutorials.ws/filter/filter_8.html) that talks about the formulation of Butterworth filters.

Finally, a more explanatory link can be found [here](https://www.youtube.com/watch?v=00hNt7uBpEI). Although the video is for analog signals and ours is digital, they are yet similar really much.

The coefficients per filter is chosen such that the frequencies I want to maintain from the signal is magnified (not fluctuating much) meanwhile excluded region is dimmed as much as possible.

**ANALYZING THE FILTERS:**

In below filter the peak is away from the direct current part but also away from the other margin too. It preserves a flat passing region with amplitude 1, which implies that the amplitude property per frequency is kind of preserved.

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Description automatically generated

Below, the filter is stuck to the origin area. It has a sharper transition. It also dims the parts outside cutoff frequencies totally. It has a short range and allows only frequencies really close to zero.

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Below filter passes all the frequencies after a certain cutoff frequency, again preserving the magnitudes (flat part corresponds to magnitude 1). The former information belongs to high pass filter thus this is a high pass filter.

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At each time step, a frequency-magnitude diagram can be thought of being multiplied with these filters elementwise basically, which explains the behaviour of the filters with visual correlation.

I tried two different methods and tried writing my own method and at the end the best results came from this Butterworth filter. (I tried fircegrip, designfilt before butter function from the function pool of matlab).

And below is the initial wave followed by filtered waves per instrument:

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A picture containing text, line, parallel, diagram

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