Lab 7: Filters

Preamble

Other formats

This document is available in HTML format for online viewing and as PDF for printing.

Acknowledgements

This lab is based on Filter Design Using Matlab Demo by David Dorran.

There is a YouTube video that illustrates what we are going to be using.

Aims

This optional lab exercise demonstrates the design and simulation of digital filters. I is not assessed, but you may find it useful preparation for the project.

Setup

Before you start

If you haven't already, create a suitable folder structure on your file-store for your labs.

I suggest

```
OneDrive\workspace
    signals-and-systems-lab
        lab01
        lab02
        lab03
        lab04
        lab05
        lab06
        lab07
        :
```

Use folder `OneDrive\workspace\signals-and-systems-lab\lab07` for this lab.

Preparation

Download the example filter design script filters.m from this repository. Save it to your folder for `lab07`.

Open the script as a MATLAB Live Script and execute the embedded code step-by step and read and understand the commentary.

Lab Exercise

Lab Exercise 15: Interactive Filter Design

MATLAB provides a filter design tool with a graphical user interface called `fdatool`.

We want you to use this tool to design and test a low-pass, band-pass and high-pass Butterworth filter with sampling frequency equal to 44.1 kHz. The filter should implement the first, second and third stage in a three-stage graphic equalizer with a low pass filter with a cut-off frequency of 31.5 Hz, a pass-band filter for the middle filter (f_1 to f_2) of about one octave and centre-frequency f_c equal to 63 hz and a high-pass filter with pass-frequency of 125 Hz.

The aim of this exercise is to determine the order of the Butterworth filters to be used in your design and the Q factor needed (where $Q = f_c/(f_2 - f_1)$) for the pass-band filters required to implement the mid-range of your 10-stage graphic equalizer.

The centre pass-band filter should be designed so that f_1 and f_1 satisfy $f_c = \sqrt{(f_1 f_2)}$. Your goal is to find the Δf value for this filter that achieves a flat frequency response when it is combined with equal weight to the low-pass and high-pass filters.

What to hand in

Claim

Up to 3 marks each can be claimed for the design evidenced by a suitable Live Script and filter design file. You should use the `filterDesigner` from the DSP System Toolbox and save your designs to disk.

Up to 2 marks can be claimed if you have a Simulink model showing the filters set with a gain of 10, 0 and -10 dB respectively. You can start with the model used in the Project Descriptor (Three_Band_EQ_Model.slx).

You may find it useful to use the **Filter Realization Wizard** block (part of the Simulink Collection from the DSP System Toolbox) which combines the `fiterDesigner` with a block that can be used in simulation.

Submission

You should submit the following to the **Lab 07: Filters** Assignment on Canvas.

1. Complete the labwork self-assessment claim form and declaration.
2. Evidence of your filter design as a m-file or MLX file.
3. Simulink model of your three part filter with gain settings -10dB, 0dB and 10dB.
4. The audio file that you used for testing.

Deadline

The deadline for claims and submission is **4:00 pm, 21st April 2021**