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Course 31606: Signals and Linear Systems in Discrete Time

31606 E18

Cloudy...

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Filtering back and forth

Submission deadline: Thursday 25.10.2018 before the start of the lecture

Assignment

These exercises are part of the final evaluation, and must be handed in as hard-copy and as .pdf named `31606_e18_assignment_<number>_<your_group_number>.pdf` (via DTU Learn). Please work on the assignment in groups of (at most) three, and produce a short report about your results written in English. Please address and answer all questions in the report. If you are asked to plot something, include the plot in the report. **For each problem, outline the problem in your own words, your approach, what you did and why you did it.** Make sure the figures are readable (see the general comments handed out before hands-on 1), and use the figure caption to describe the figure. Provide all the code in a .zip file and upload it together with your assignment. Make sure the code is **runnable, well commented and follows the general guidelines** uploaded on DTU LEARN. Organize the code in a folder structure as shown below:

```
31606_e18_grp_<your_group_number>
└─ assignment_<assignment_number>
    ├─ <multiple files if you feel like it>
    ├─ <add a cooking recipe if you have a good one>
    └─ ...
```

For statistical (non-commercial) purposes, please indicate the number of hours spend for each of the group members on the report.

1 Making more of less

The following example comnes from a book on DSP. However, there is some error in the book: The filter specs in the book don't match the solution. To improve the book for future generations, let's do a solution that does meet the filter criteria specified. Analytical calculations are helpful up to a certain point - feel free to do them and to include them. But you might as well make use of the tools built-in into Matlab to do the numerics.

Here you will be designing a lowpass filter that meets certain specifications. The approach is similar for different filter types and what you get out is something you can apply and that has properties depending on how you generated it - that's why it is neccessary to understand what is behind different routines.

1.1 A specific example

Design a digital lowpass filter using bilinear transform with prewarping meeting the following specifications:

- A gain of unity at $\omega = 0$

Why is the bandpass not just 1 number?

- A gain of no less than -2 dB ($G_p = 0.785$) over the passband $0 \leq \omega \leq 10$
- A gain no greater than -11 dB ($G_s = 0.2818$) over the stopband $\omega \geq 15$
- The highest frequency to be processed $\omega_h = 35$ rad/s

  `useful commands: butter, buttord`

Show the amplitude and phase response of the resulting filter together with the filter specifications in the title of the plot. Write down the transfer function of the resulting filter and specify the filter type.

1.2 The issue of phase shifts and order

Now we have a filter meeting the required specifications. Now assume your filter coefficients are implemented, compiled and the source codes are lost. You have a filter of the order coming out of its design and there is no way to change the properties of the filter - it is already in production and now you are to use the product you have in your hand.

I assume you find out how much it phase shifts adn thn write an all-passfilter that phase shifts the opposite.

- When passing a pure tone signal through the filter, it will be scaled and shifted in phase. Assume you are asked to process a signal through your filter - while the magnitude of the transfer function is less important, you definitely need zero phase shift. Use the filter designed above to process a sinusoidal signal such that the resulting phase shift is zero. Plot the original and the processed signals on top of each other.
- Now assume the opposite: You need to have a signal processed where the filter order needs to be about twice/three times what your designed filter has, but the phase is less important. What could you do in order to have a filter with approximately twice or three times the order? Use the filter you just designed to solve this problem. Plot the corresponding frequency transfer functions on top of each other.

Could you just pass it through several times?

2 Some stuff from the previous hands-on

The remaining part of the assignment is based on the hands-on. The focus in this part is to document that you understood how the exercises have been solved, why they were solved in the way you did it and what the theoretical background behind them is. You may use snipplets of your code to explain how it works and how you solved the task. The code needs to be complete, a solution without explanation is not a solution!

2.1 Hands-on 5

Please provide a solution and the corresponding explanation for 3.