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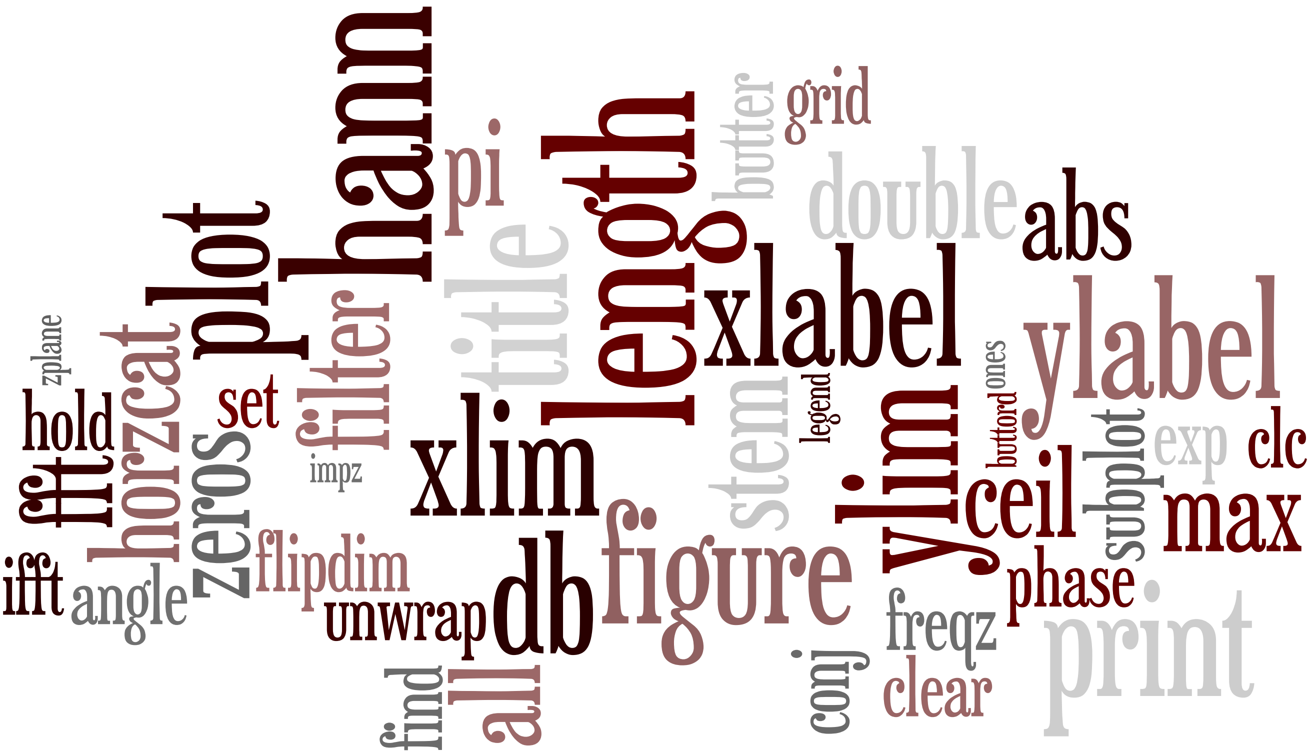
Course 31606: Signals and Linear Systems in Discrete Time

Cloudy...



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Bass/treble - advanced

Submission deadline: Thursday 15.11.2018 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 From bass/treble to an equalizer

In the previous hands-on you programmed a digital version of a bass/treble knob. This allowed some amplification/attenuation of the low/high frequencies, just as on an old amplifier. You might think that this is not enough, so let's get a bit more advanced and construct a digital version of an equalizer!

1.1 Five knobs to turn

Implement a function that constructs a FIR filter mimicking five different bands (subbands) of an equalizer, each covering an equal bandwidth on a linear frequency scale. The function takes the gains (which can be negative or positive) for each of the subbands and returns the filter coefficients of the resulting (overall) filter. Choose the order of the filter yourself and explain your choice.

- Plot the frequency response of the filter with gains that decrease in steps of 5 dB from the lowest to the highest subbands
- Zero-pad your impulse response to twice/ten times the length and plot again the frequency response. Explain what you see.
- Pass a white noise signal through your filter and plot the spectra of the original and the filtered signal on top of each other.
- Process your favourite piece of music through the filter and listen to it! It's fantastic, isn't it?

2 The grandparents and DSP

It was suggested by one of the feedback persons to have some "real" questions in the assignments. So here they come. Please write your answer in technical terms and in a concise way. Assume you would want to explain the issue to a colleague whos DSP knowledge might be a bit rusty:

1. Assume your grandmother is bored during christmas eve and is generating a sweep in MATLAB (sampling frequency of 5000 Hz), starting at 50 Hz and reaching up to 5000 Hz. Now she plays back the sweep in via your stereo system in your living room. Despite being impressed by your DSP-grandmother, your parents are a bit puzzled by the result. Explain a) what you expect to hear, b) why your parents are puzzled and c) what the reason behind the observed phenomenon is.

2. Now your grandfather woke up and also wants to play with DSP. But since the computer is occupied by your grandmother, he takes pen and paper and wants to impress people with his math skills. He shows that the stock market curve in the newspaper is smoothed by the application of averaging 3 subsequent points on the graph. He forgot what the time interval between two sampling points are, but figured out a way to have a generalized description. He found that some fluctuations would be totally suppressed by this averaging technique and explains this using his drawing of the frequency response of this 3-point filter. He also managed to compute the analytical form of the frequency response and further showed that this filter has linear phase and can even have zeros phase under some conditions. Derive the frequency response of the filter used by your grandfather and show the linear/zero phase

of that filter. Calculate which frequency would be totally suppressed if the update time between two points on the stock market would be one second. Explain how this frequency response can be expressed without knowing the exact sampling frequency.

3 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!

3.1 Hands-on 8

Please provide a solution to hands-on 8, number 1.