
Information about assignments and in particular on Moodle submission modalities can be found in DSP-Tutorial_2024S_CourseInfo.pdf. Submission deadline is Fr., **July 5, 2024, 08:00**.

Exercise 1 *Signal Distortion and Group Delay (20%)*

Generate three periods of the signal

$$x[n] = \sum_{i=1}^4 \frac{1}{2i-1} \sin(2\pi 0.005(2i-1)n),$$

and load (MATLAB command `load`) the provided mat file `Filter_coefficients.mat`. This file contains the filter coefficients of an FIR filter (`b1` and `a1`) and the filter coefficients of an IIR filter (`b2` and `a2`). Both filters are designed to fulfil the same design criteria (filter specification). Filter with both the FIR and the IIR filter the signal $x[n]$ and observe the signal distortion. Show and discuss the results in your report.

Exercise 2 *z-Transform (25%)*

Consider the difference equation of a recursive LTI system

$$y[n] = x[n] - \frac{1}{15} y[n-1] + \frac{2}{5} y[n-2].$$

- Sketch the block diagram of the LTI system corresponding to the given difference equation.
- Determine the filter type (FIR or IIR) and explain your choice in the report.
- Compute the transfer function $H(z) = \frac{Y(z)}{X(z)}$.
- Calculate the poles and zeros of $H(z)$ and include a sketch of the pole-zero map in your report. Also include the region of convergence (ROC) in the pole-zero map.
- Is the system stable? Explain your answer.

Exercise 3 *Recursive Filter (25%)*

Let the poles and zeros of a recursive filter be given:

- Zeros: $N_1 = -1, N_2 = j, N_3 = -j$
- Poles: $P_1 = 0, P_2 = 0.75 + j0.25, P_3 = 0.75 - j0.25$

- Does this filter have real coefficients? Justify your answer.
- Draw a sketch of the pole-zero map.

- (c) State the transfer function, first with polynomials of z^{-i} , $i = 0, 1, 2, 3, \dots$, and afterwards with polynomials of z^{+i} .
- (d) Draw the block diagram of a direct-form-I implementation of the filter and specify the coefficient values in the block diagram.
- (e) Plot the magnitude and phase response of the filter in MATLAB.
- (f) Plot the impulse response of the filter for $0 \leq n \leq 50$ in MATLAB.

Exercise 4 Lowpass Filter Design (30%)

An analogue signal is sampled with a sampling frequency $f_s = 20$ kHz and filtered subsequently. The digital lowpass filter should exhibit the following specification:

- Passband cutoff frequency: $f_{\text{pass}} = 3.4$ kHz
- Stopband cutoff frequency: $f_{\text{stop}} = 4$ kHz
- Allowed ripple in the passband: ± 5 %
- Minimum stopband attenuation: 45 dB

- (a) Specify the normalized radian frequencies for the passband Ω_{pass} and the stopband Ω_{stop} , the passband tolerance δ_1 and the stopband tolerance δ_2 . *Hint:* Be sure to use the decadic logarithm $\log_{10}(\)$ for conversion to decibels.

- (b) What is the ideal impulse response $h_{\text{ideal}}[n]$ for the ideal frequency response

$$H_{\text{ideal}}(\Omega) = \begin{cases} 1 & \text{for } 0 \leq |\Omega| \leq \Omega_0 \\ 0 & \text{for } \Omega_0 \leq |\Omega| \leq \pi \end{cases},$$

$$\text{with } \Omega_0 = \frac{\Omega_{\text{pass}} + \Omega_{\text{stop}}}{2} ?$$

- (c) Which two measures are necessary to deduce a realizable FIR system of order N from the ideal impulse response $h_{\text{ideal}}[n]$?
- (d) How can the decrease to the specified filter order N be interpreted, and which effect on the frequency response of the realizable filter does that have?
- (e) Design an FIR filter of order $N = 20$ with a rectangular window with a corner radian frequency Ω_0 . Plot its frequency response and the tolerance scheme in one plot. To this end, complete the provided file `dsp_5_4.m`.
- (f) Is the tolerance scheme being violated? Can the tolerance scheme be fulfilled by increasing the filter order to $N = 90$?
- (g) Now, use a hamming window instead of the rectangular window (for $N = 90$) and assess the result.
- (h) Use the MATLAB `filterDesigner` to design an elliptic IIR filter fulfilling the above described requirements. What is the order of this filter? What is the disadvantage of this filter?