# Effects of overflow and underflow in digital filtering

Lab 10, SDP

### **Objective**

Students should observe the effects of internal format overflow and underflow events on the output of a digital filter.

#### Theoretical notions

#### **Exercises**

1. Consider the following system:

$$H(z) = \frac{1 - \frac{1}{2}z^{-1}}{\left(1 - \frac{1}{4}z^{-1}\right)\left(1 + \frac{1}{4}z^{-1}\right)}$$

- a. Draw the direct form II implementation and one of the series implementations
- b. Assume a fixed-point implementation on b bits. Each product is quantized by rounding to b bits. Find the variance of the rounding noise due to the internal multiplications, at the output of each implementation from a.
- 2. In Matlab, use the fdatool tool to design a high-pass IIR filter of order 4, Butterworth type, with cutoff frequency equal to 1.5kHz for a sampling frequency of 44.1kHz. Convert the filter to direct form II and export it to Simulink (check the option Build model using basic elements).
- 3. In the Simulink model, make two copies of the filter (Copy/Paste). At the input of both filters put a sample audio signal (e.g. Kalimba.mp3 or mtlb). For the second filter, the input signal shall be converted to fixed-point 1S2Î9F data type. Compute and plot the difference between the two outputs. Export the difference signal to Matlab's Workspace and compute its average value and the dispersion.

- 4. Make a new copy of the second filter, and for this copy un-check in the properties of the *Sum* blocks the option *Saturate on integer overflow*. What is its effect? Compute and plot the difference compared to the first filter. Which of the filters 2 or 3 produces smaller errors?
- 5. Use the function normescal() to compute different scaling norms for the given filter. Scale the input signal with each one of these norms, and rescale back the output. What is the effect of this procedure?

## **Final questions**

1. TBD