# Implementing digital FIR filters in the lattice form

Lab 5, SDP

### **Objective**

The students should become familiar with *lattice*-type realization structure used for implementing FIR filters.

#### Theoretical notions

#### **Exercises**

- 1. Find the FIR filter coefficients in direct form, if the reflection coefficients of the lattice FIR structure are:  $K_1 = \frac{1}{2}$ ,  $K_2 = 0.6$ ,  $K_3 = -0.7$ ,  $K_4 = \frac{1}{3}$ .
- 2. Find the reflection coefficients of the lattice structure for a FIR filter with system function:

 $H(z) = 1 + \frac{2}{5}z^{-1} + \frac{7}{20}z^{-2} + \frac{1}{2}z^{-3}$ 

- 3. Using the Octave software, use the fir1() function to design one of the following FIR filters:
  - a. A low-pass FIR filter of order 4, with cutoff frequency of 5kHz at a sampling frequency of 44.1kHz;
  - b. A high-pass FIR filter of order 4, with cutoff frequency of 2kHz at a sampling frequency of 44.1kHz;
  - c. A band-pass FIR filter of order 4, with passband between 1kHz and 3kHz at a sampling frequency of 44.1kHz.

Read the documentation of the fir1() function to find out how to use it.

- 4. Create an Octave function to tf2latc() to compute the coefficients of the lattice form of a FIR filter, starting from the coefficients of the Transfer Function. Call it like this: K = tf2lact(coef)
- 5. Create an Octave script to filter an input signal x with a FIR filter in lattice form, for which the reflection coefficients K are known:

```
y = filter_latc(x, K)
```

- 6. Use the function above to load and low-pass the audio signal Kalimba.mp3.
  - a) Load the file using audioread()
  - b) Use tf2latc() to convert the filter to lattice form
  - c) Filter the signal with filter\_latc()

## **Final questions**

1. TBD