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

Lattice form for an FIR filter of order 3:

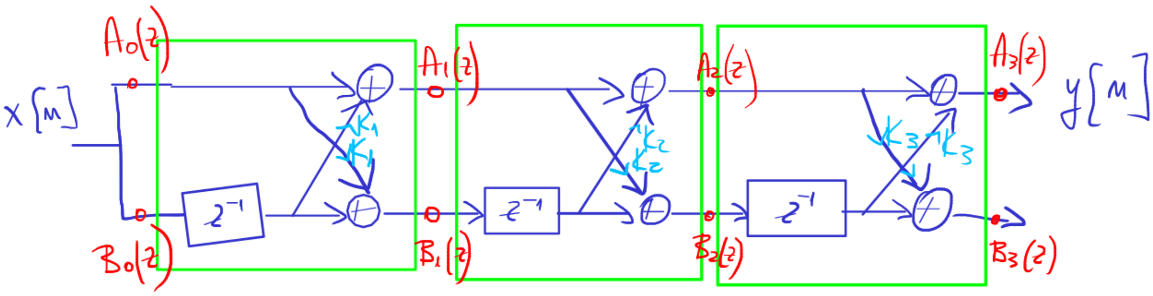


Figure 1: Lattice form, order 3

Equations:

$$\begin{aligned}
A_0(z) &= B_0(z) = 1 \\
A_m(z) &= A_{m-1}(z) + K_m \cdot z^{-1} \cdot B_{m-1}(z) \\
A_{m-1}(z) &= \frac{A_m(z) - K_m \cdot B_m(z)}{1 - K_m^2} \\
B_m(z) &= z^{-m} B_m(z^{-1}) = \text{ like } A_m(z)\text{, with coefficients reversed}
\end{aligned}$$

These equations allow to convert $H(z)$ to the reflection coefficients needed by the lattice implementation, or to find $H(z)$ from a given lattice implementation.

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. In the Matlab environment, use the `fdatool` tool to design one of the following filters:

   a. A low-pass IIR filter of order 4, elliptic type, with cutoff frequency of 5kHz at a sampling frequency of 44.1kHz;
   b. A high-pass IIR filter of order 4, elliptic type, with cutoff frequency of 2kHz at a sampling frequency of 44.1kHz;
   c. A band-pass IIR filter of order 4, elliptic type, with passband between 1kHz and 3kHz at a sampling frequency of 44.1kHz.

4. In the Simulink environment, implement the above filters in *lattice* form. Apply at the input an audio signal and play the output signal, as well as the original, for comparison. How does the filtered signal sound like, compared to the original?

Notes:

- Set the following parameters for the SImulink model, to enable a discrete simulation with fixed (auto) step:
  - Type: *Fixed-step*
  - Solver: *discrete (no continuous states)*
- You will need the blocks *Unit Delay*, *Sum* and *Gain*
- At the input put a *From Multimedia File* block, and at the output put a *To Audio Device* block
- At the output, before the *To Audio Device* block, put a *Manual Switch* block in order to be able to switch easily between the original signal and the filtered one
- For the *From Multimedia File* block, select an audio file (de ex. Kalimba.mp3 from My Documents) and update the following settingsȘ

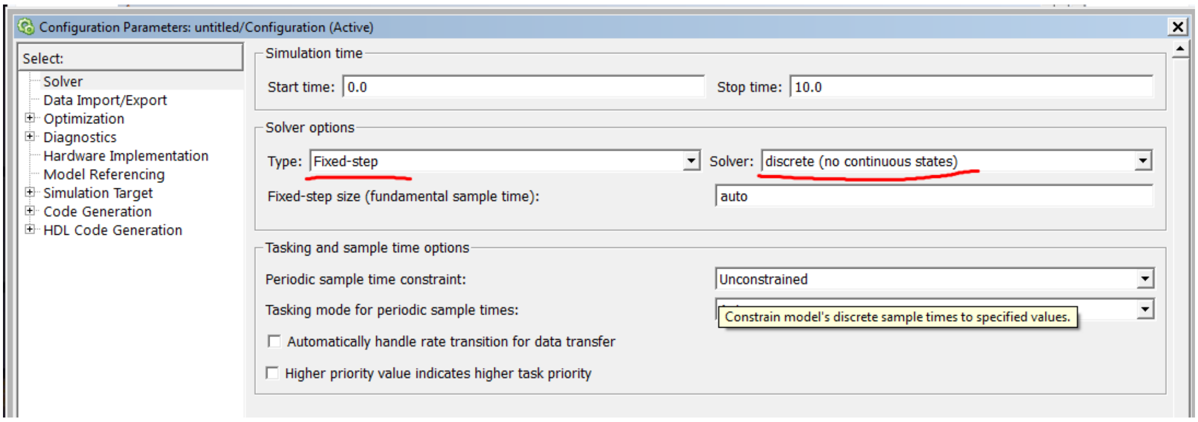
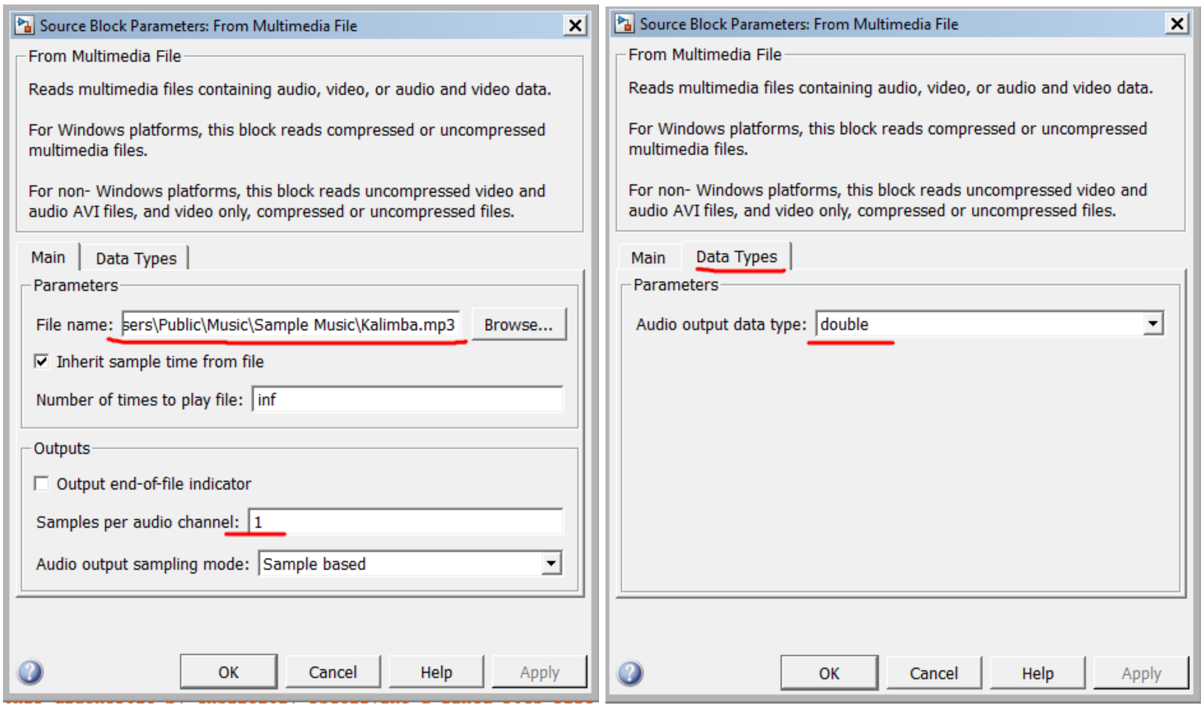


Figure 2: Model settings for discrete models

- choose *Sample-based*
- *Samples per audio channel* = 1
- "DataTypes/Audio output data type" = *double*



Final questions

1. TBD