

Signal Processing Block:

Configurable LPF/HPF for Digital Audio Signal

Integration in the Audio Synthesizer Project

Digital Filter type FIR

Structure

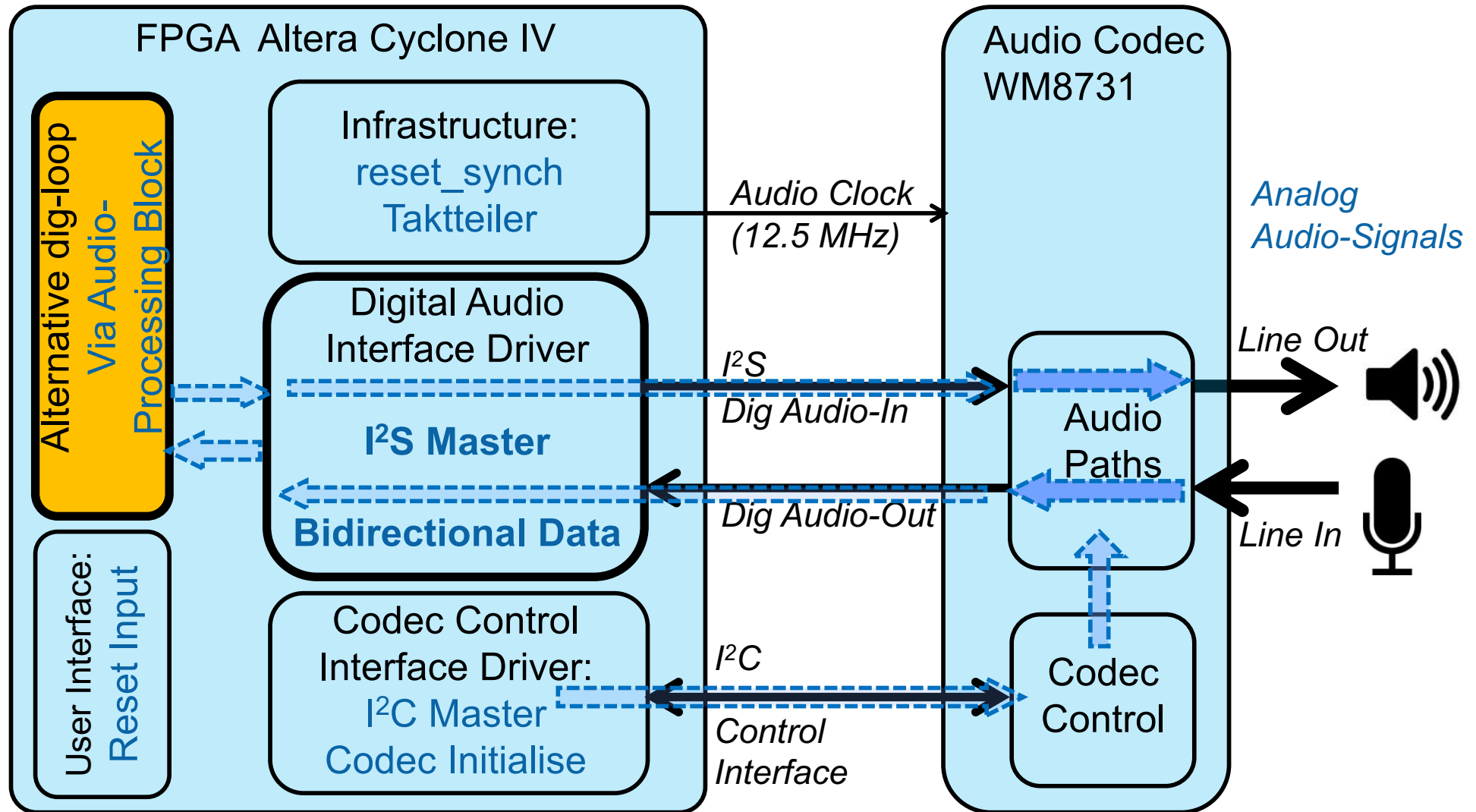
Design & Quantisation (w/ Matlab)

HW Implementation (w/ FPGA)

Simulation & Test

Phase-2 Milestone-2

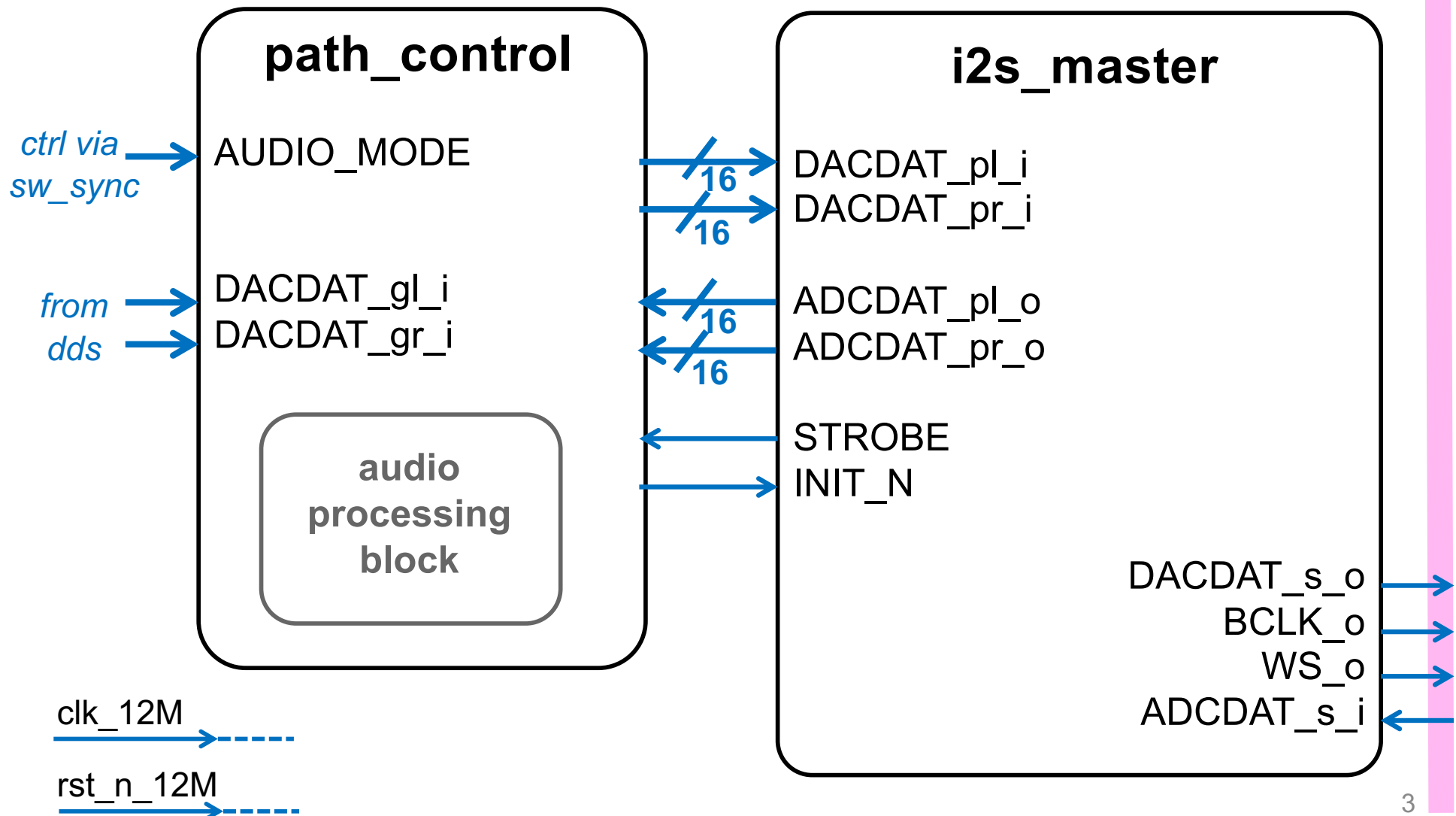
Digital Audio-Loop Test



Objective: modify the digital audio signal with a filter

Digital Audio Interface Driver Blocks

FPGA



Digital Filter type FIR

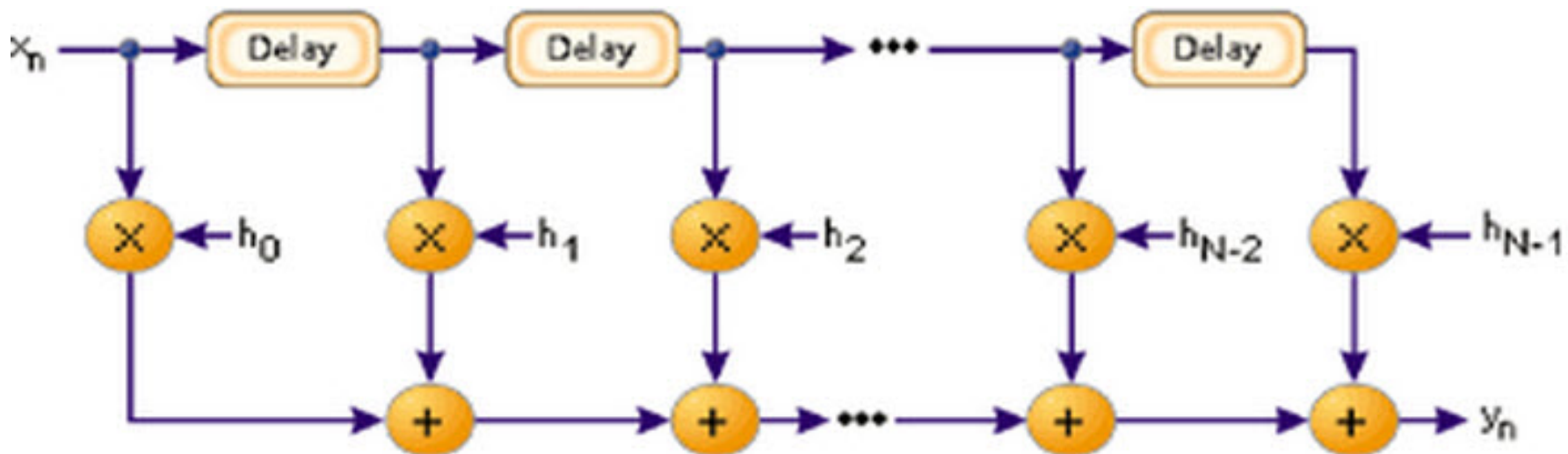
FIR (finite impulse response)

Principle:

Several input samples are multiplied with coefficients and added up.

Implementation Structure:

Need a delay line, multipliers and adders.



Digital Filter type FIR

FIR (finite impulse response)

Digital Filter Design:

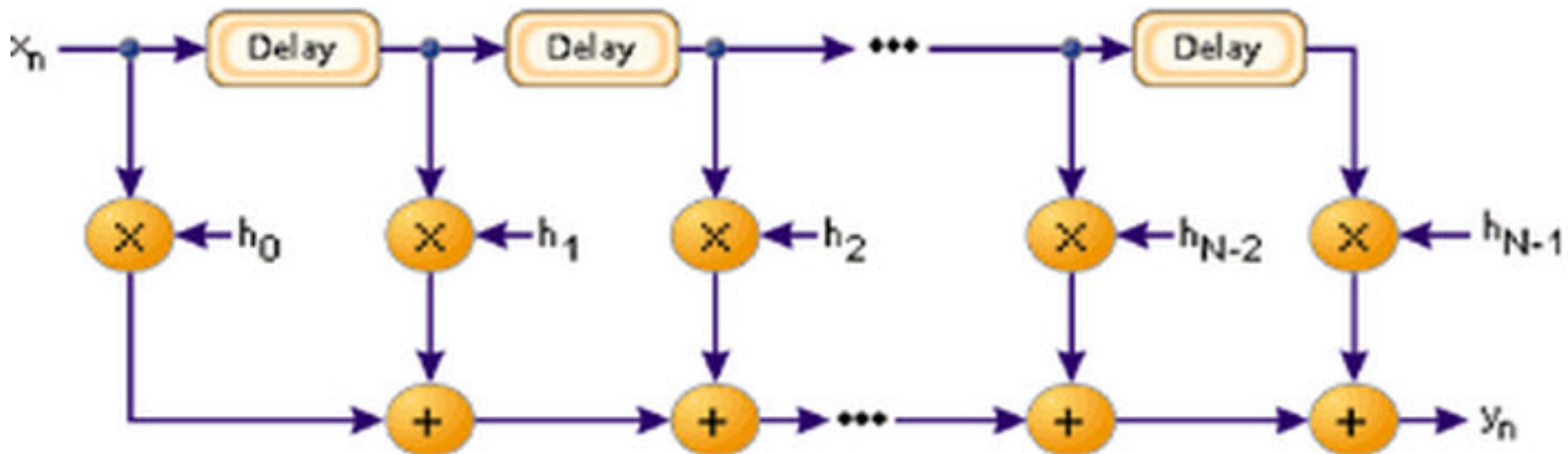
How many coefficients? How many bits pro coefficient (resolution)?

Can be calculated & simulated in Matlab, and it is topic for later semesters!

Implementation Details:

Delay line in FPGA => memory block

Multiplication block => not many available, solution: higher frequency and reuse

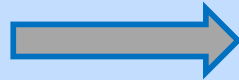


Digital Filter type FIR

HW Implementation

FPGA Resources

what is new



Mem Blocks

for delay line

Multipliers

also called DSP-blocks

VHDL Description

what is new

Mem Blocks

use a separated block with behavioural description of RAM blocks which is supported by synthesizer
For Cyclone devices:
M4K Blocks, described in handbook as embedded memory

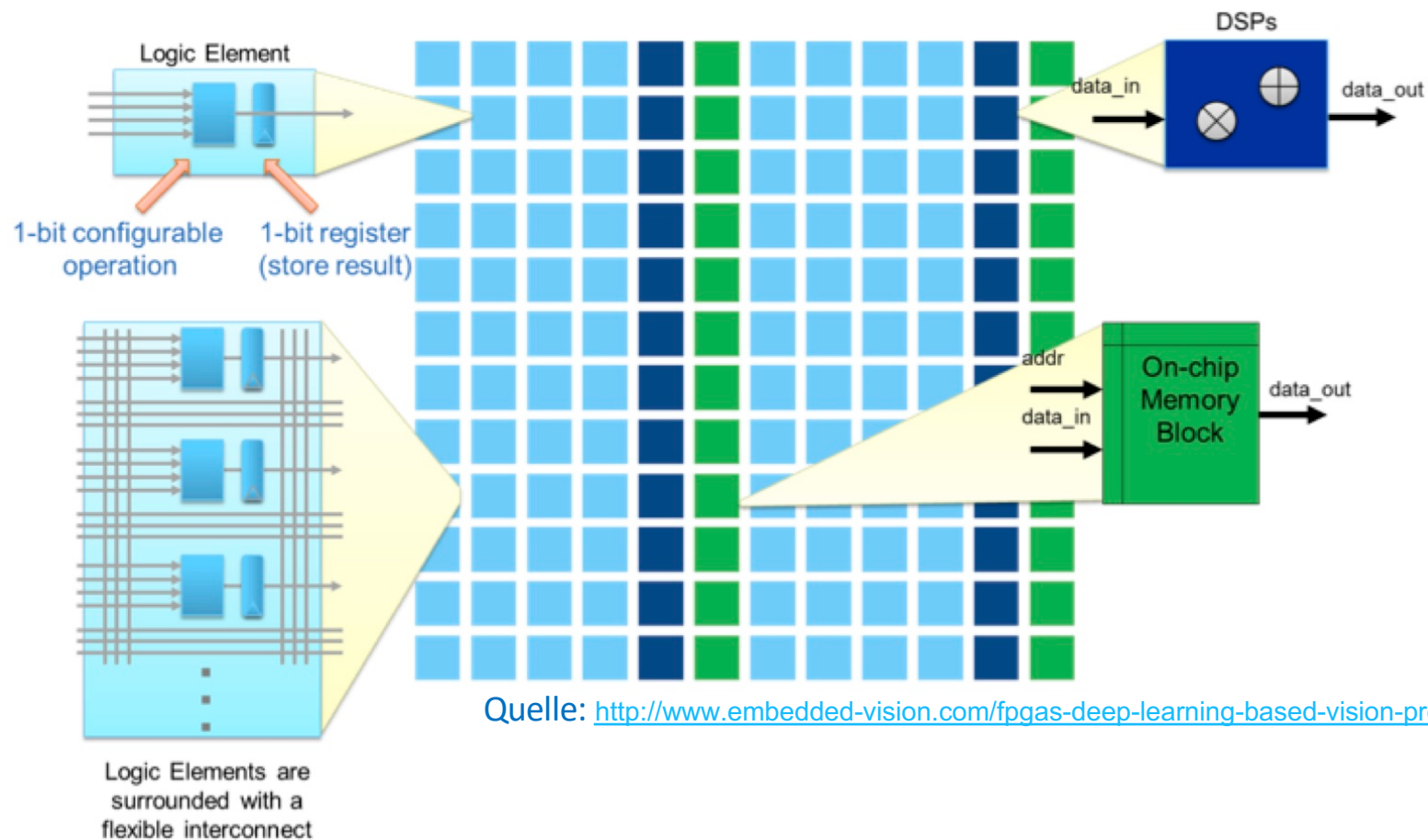
Multipliers

use signals with defined resolution
consider increase of width on output

Field Programmable Gate Array (FPGA) Architecture (2/3)

Plus Macrocells, like :

RAM blocks ; Multipliers and Adders (DSP blocks) ; PLLs (clock generation);
Clock networks (clock distribution) and Input/Output blocks



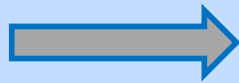
Quelle: <http://www.embedded-vision.com/fpgas-deep-learning-based-vision-processing>

Digital Filter type FIR

HW Implementation

VHDL Syntax

what is new



Arrays

For Look-Up-Table (LUT storing filter coefficients)

[extract from audio_filter_pkg.vhd]

```
constant N_LUT:                natural := 255;    -- length of LUT = 255
constant N_RESOL_COEFF:        natural := 16;      -- Attention: 1 bit reserved for sign

-- range : [-2^15; +(2^15)-1] = [-32768 ; +32767]
subtype t_fir_range is integer range -(2**(N_RESOL_COEFF-1)) to
                                     (2**(N_RESOL_COEFF-1))-1;

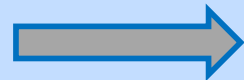
type    t_lut_fir is array (0 to N_LUT-1) of t_fir_range;

constant LUT_FIR_LPF_200Hz : t_lut_fir :=(
    -14,-15,-19,-25,-34,-45,-58,-73,-90,-109,-129,-149,-170,-189,-208,
    ...    );
```

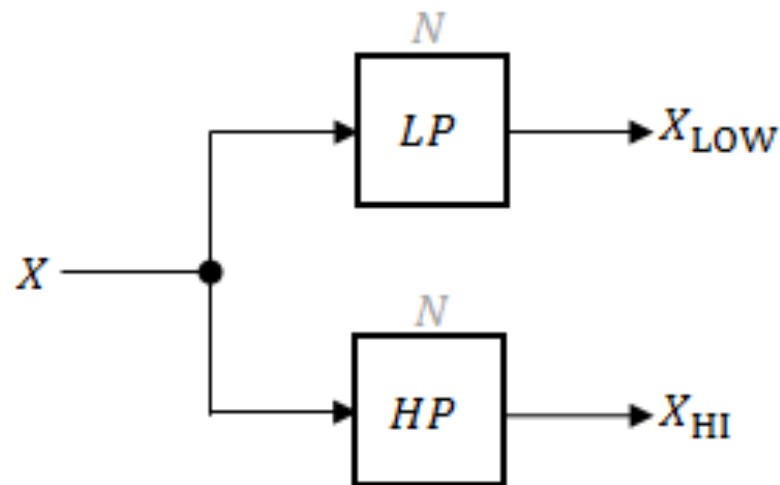

Digital Filter type FIR

HW Implementation

Low Pass Filter



Complementary High Pass Filter



Based on the idea above

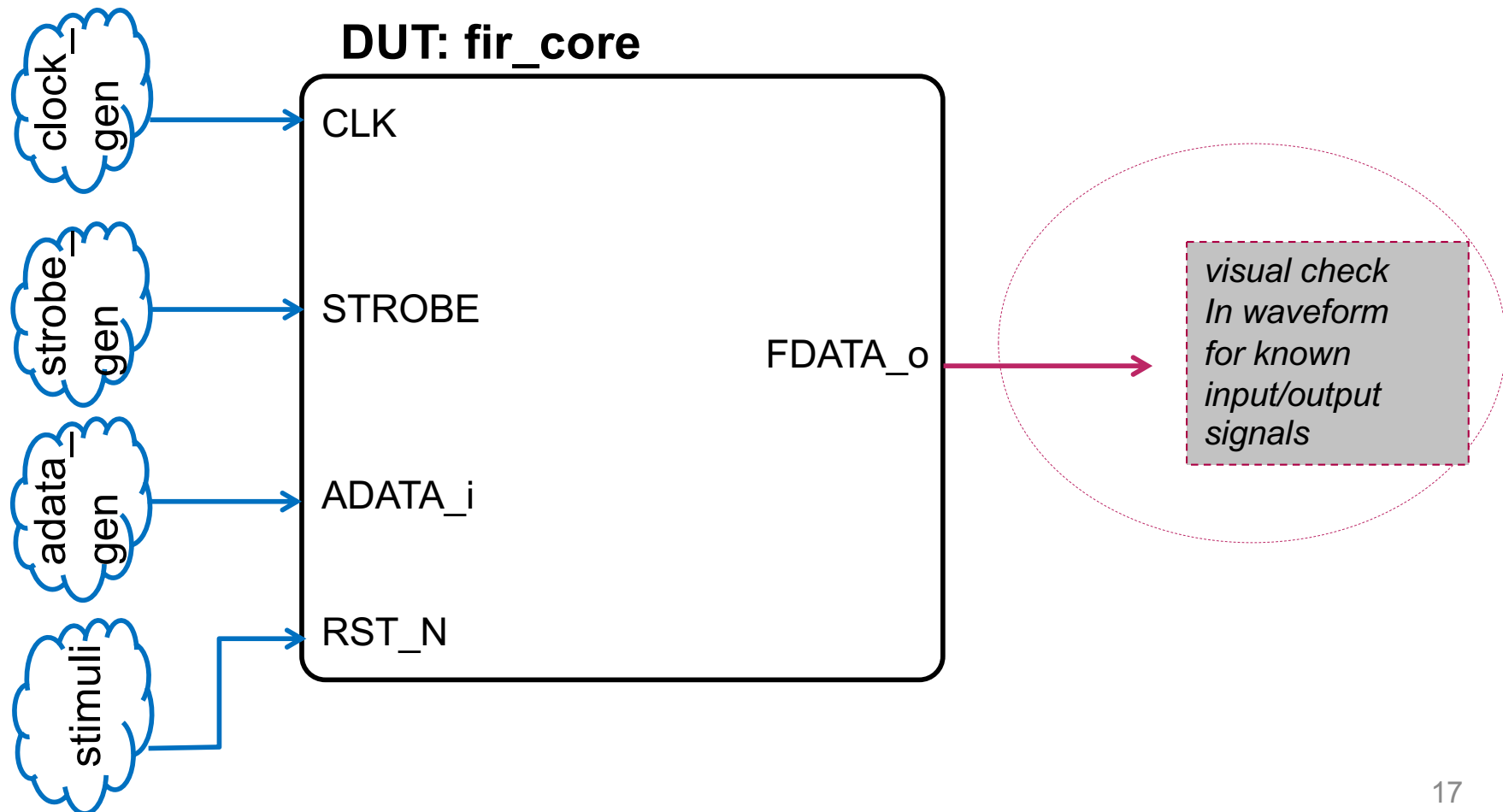
(that the input signal can be split into its low-pass and high-pass frequency contents)

How would you calculate the output of the HPF based on the LPF?

Testbench for FIR-Core

Block Level TB around LPF

testbench



Testbench for FIR-Core

Block Level Simulation: Impulse & Square-Pulse Responses

