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Lesson 9: Digital Signal Processing Applications
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Design a linear interpolator that would double the sampling frequency. The code should
be parametrizable with a flexible word size, but you can assume that it will always
double the sampling frequency by inserting one sample between adjacent samples received.
The inserted interpolated word should represent the average of the two received samples.
The output should be operating at the faster clock, and the input should update at the
slower clock frequency. This should be clear in the test bench. The module should
have a reset.
Combine your code for the interpolator, the testbench, and simulation results
in one pdf file.
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module Interpolator (data_in, clk, reset, data_out);

parameter word_size = 4;
input  [word_size-1 : 0] data_in;
output [word_size-1 : 0] data_out;
input  reset, clk;

reg      [word_size : 0] data_reg;      // enough bits to calculate the sum
reg      slow_clk;
reg      [word_size-1 : 0] buffer [0:2]; // to guarantee sample alignment

assign data_out = data_reg [word_size-1 :0];

always @ (posedge clk)
    if (reset) begin
        slow_clk    <= 0;
        buffer[0]   <= 0;
        buffer[1]   <= 0;
        buffer[2]   <= 0; end
    else begin
        slow_clk    <= ~slow_clk;
        buffer [0]  <= data_in;           // input reg to guarantee sample alignment
        buffer [1]  <= buffer[0];        // even if the input is off sync with clock
        buffer [2]  <= buffer[1];
        data_reg    <= slow_clk ? {1'b0,buffer[1]} : (( buffer[0] + buffer[2] )>>1) ;
    end

endmodule

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