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# Realization of digital linear systems

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**Abstract**
For the design of digital filters, the system function H(z) or the impulse response h(n) must be specified. Then the digital filter structure can be implemented or synthesized in hardware/software form by its difference equation obtained directly from H(z) or h(n).Each difference equation or computational algorithm can be implemented by using a digital computer or special purpose digital hardware or special programmable integrated circuit.
In order to implement the specified difference equation of the system, the required basic operations are addition, delay and multiplication by a constant. A chosen structure determines a computational algorithm. Generally, different structures give different results. The most common methods for realizing digital linear systems are direct, cascade and parallel forms, and state variable realizations.

**Keywords:** Unit delay, adder, multiplier

## 1. Introduction
### 1.1 Building Blocks of Digital Filters
Any digital system that is linear, time invariant, rational, and causal can be realized using three basic types of element:

1. A unit delay: The purpose of this element is to hold its input for a unit of time (Physically equal to the sampling interval T) before it is delivered to the output. Mathematically, it performs the operation

$$Y[n] = x[n-1] \qquad \ldots(1)$$

Unit delay can be implemented in hardware by a data register, which moves its input to the output when clocked. In software, it is implemented by a storage variable, which changes its value when instructed by the program.

2. An adder: The purpose of this element is to add two or more signals appearing at the input at a specific time. Mathematically, it performs the operation

$$Y[n] = x1[n]+x2[n]+x3[n]+\ldots \qquad \ldots(2)$$

3. A multiplier: The purpose of this element is to multiply a signal (a varying quantity) by a constant number.
Mathematically,

$$Y[n] = a\, x[n] \qquad \ldots(3)$$

We do not use a special graphical symbol for it, but simply put the constant factor above (or beside) the signal line. A physical multiplier (in hardware or software) can multiply two signals equally easily, but such an operation is not needed in LTI filters [1, 2].
A given transfer function may be realized in many ways. Consider how a simple expression such as could be evaluated – one could also compute the equivalent. In the same way, all realizations may be seen as "factorizations" of the same transfer function, but different realizations will have different numerical properties. Specifically, some realizations are more efficient in terms of the number of operations or storage elements required for their implementation, and others provide advantages such as improved numerical stability and reduced round-off error. Some structures are better for fixed point arithmetic and others may be better for floating point arithmetic [3-5].

## 2. Direct Form Realisation of Iir Systems

IIR filter transfer function can be expressed as:

$$H(z) = \frac{B(z)}{A(z)} = \frac{\sum_{k=0}^{N} b[k] \cdot z^{-k}}{1 + \sum_{k=1}^{N} a[k] \cdot z^{-k}}$$

Where:
N - Filter order;
$b_k$ - Coefficient of non-recursive part of IIR filter;
$a_k$ - Coefficient of recursive part (feedback) of IIR filter.

The coefficients $b_k$ and $a_k$ are of interest for IIR filter realization (both hardware and software). Figure 1 1illustrates the block diagram of IIR filter [6-8].

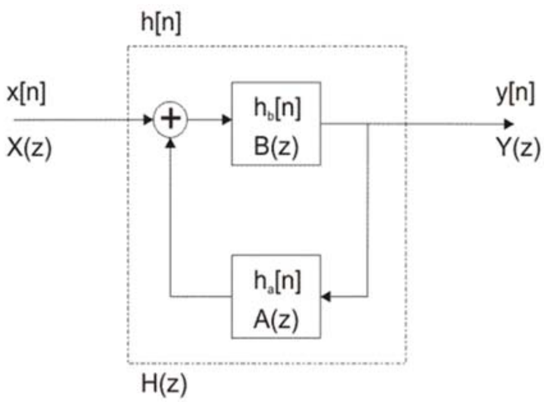
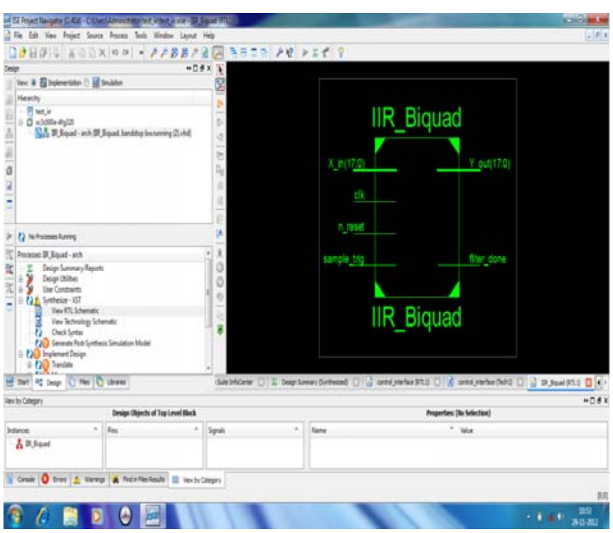


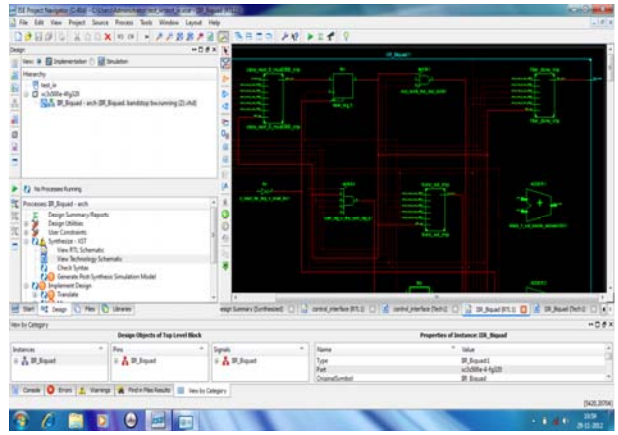

**Fig 1:** Block diagram of IIR filter

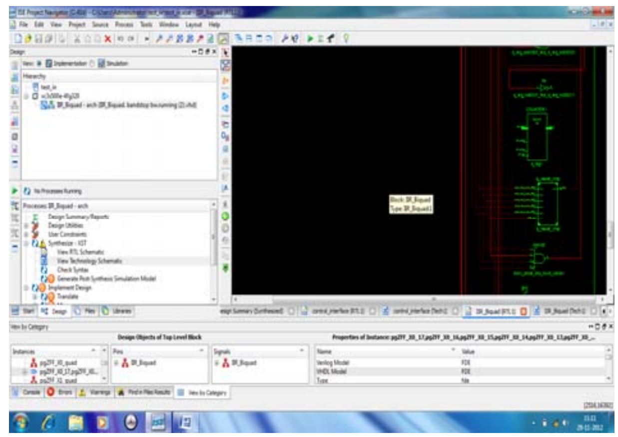
There are several types of IIR filter realization. This chapter covers direct form 1, direct form 2, cascade realizations and parallel realization. All of them are very convenient and most commonly used for both hardware and software IIR filter realization [9-13]

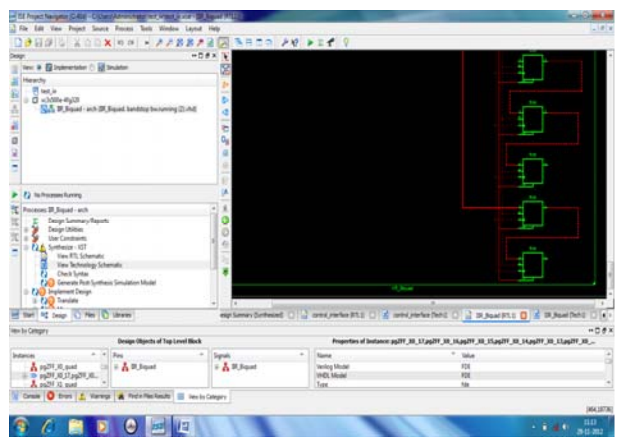
## 3. Simulation, Implementation Details and Results Obtained

RTL VIEW







## 4. Conclusion

In this paper, the design of IIR filters was considered. Several results from theory were verified in the design. The bilinear transformation was studied in some depth through its application to the design of two filters. The characteristics of a number of important approximations – Butterworth, Chebyshev, and Elliptic – were affirmed from the results obtained. The design of the low-pass filter was particularly insightful in comparing the relative merits and demerits of FIR and IIR filters in general as well as the individual IIR filter approximations. The significant observations made in the design process were:

- IIR filters result in a lower order than the corresponding (designed to meet the same Specification) FIR filter
- IIR filters exhibit non-linear phase. The bilinear

Transformation results in a frequency warping of the higher frequencies.

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