

Lecture

8

ECE216B

Digital Filters

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Agenda

Example: radio systems [1]

- Band-select filters
- Adaptive equalization
- Decimation/interpolation

Filter types:

- Direct
- Recursive
- Multi-rate



Implementation

- Conventional
- Distributed arithmetic

[1] J. Proakis, Digital Communications, (3rd Ed), McGraw Hill, 2000.

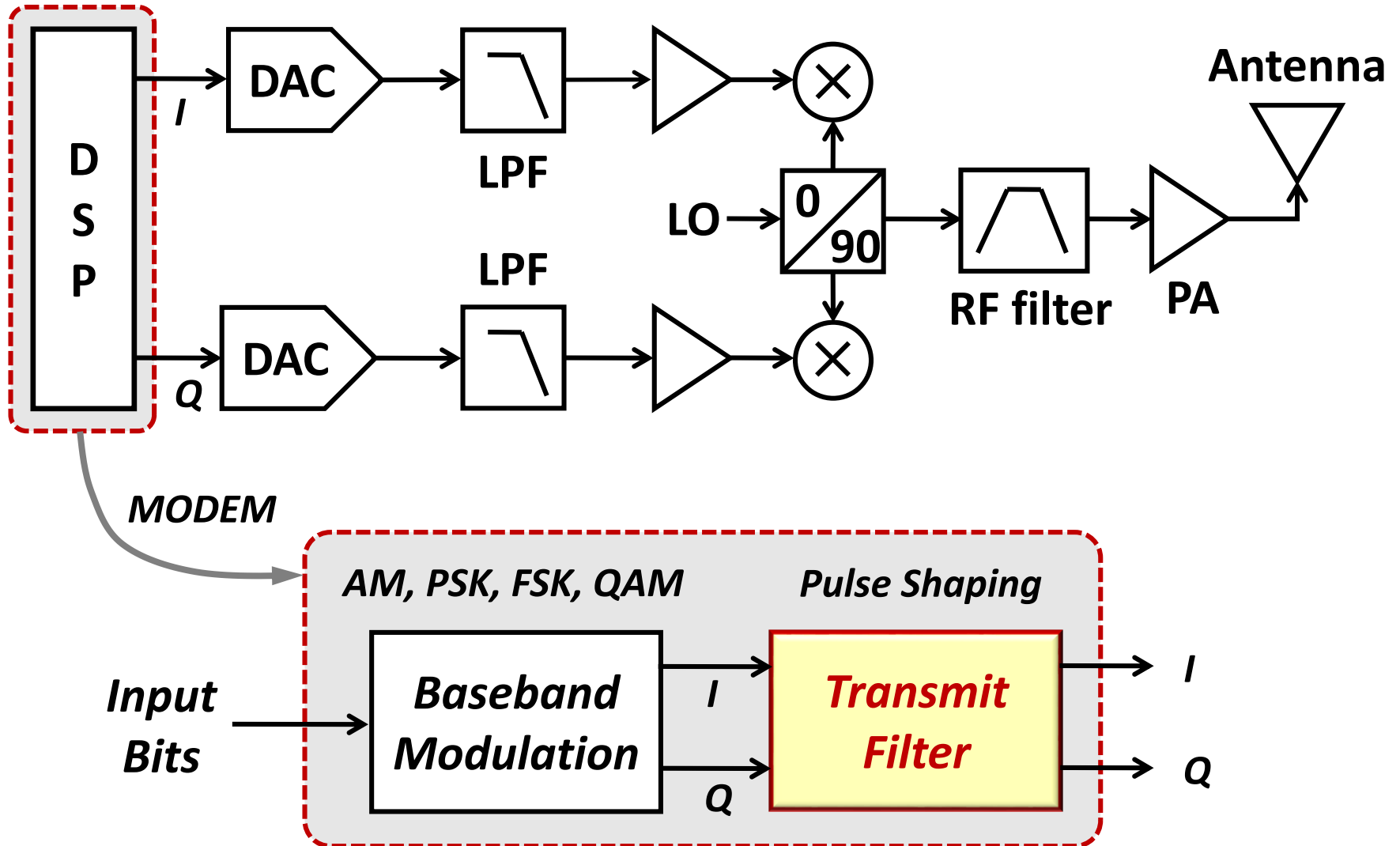
[2] A.V. Oppenheim and R.W. Schaffer, Discrete Time Signal Processing, (3rd Ed), Prentice Hall, 2009.

[3] J.G. Proakis and D.K. Manolakis, Digital Signal Processing, (4th Ed), Prentice Hall, 2006.

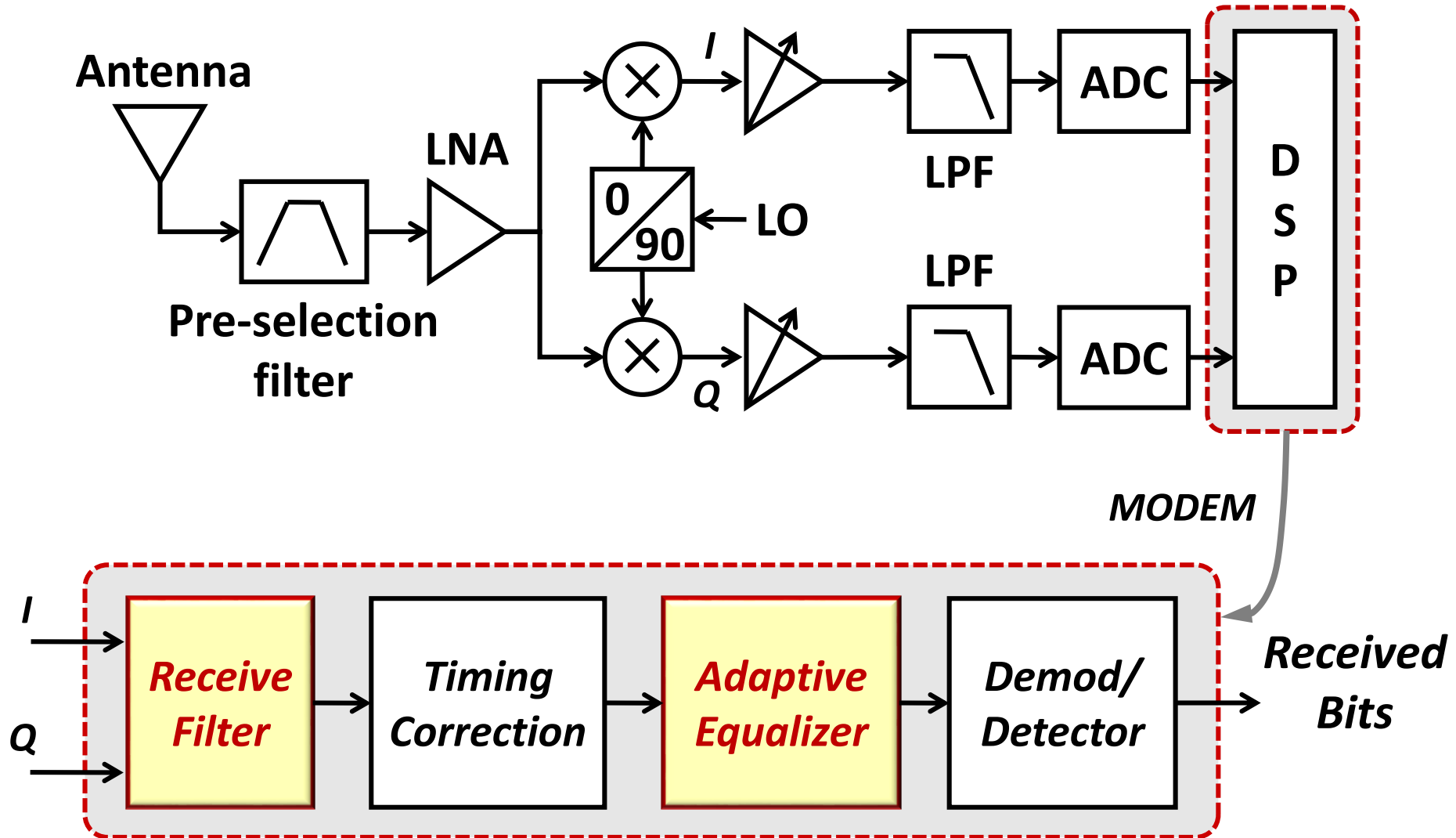
[4] K.K. Parhi, VLSI Digital Signal Processing Systems: Design and Implementation, Wiley, 1999.

Filter Design for Digital Radios

Radio Transmitter



Radio Receiver



Design Procedure

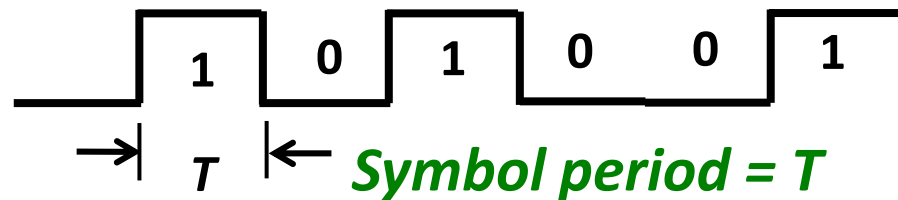
Assume: Modulation, T_s (symbol period), bandwidth

- **Algorithm design**
 - **Transmit / receive filters**
 - Modulator
 - Demodulator
 - Detector
 - Timing correction
 - **Adaptive equalizer**
- **Implementation architecture**
- **Wordlength optimization**
- **Hardware mapping**

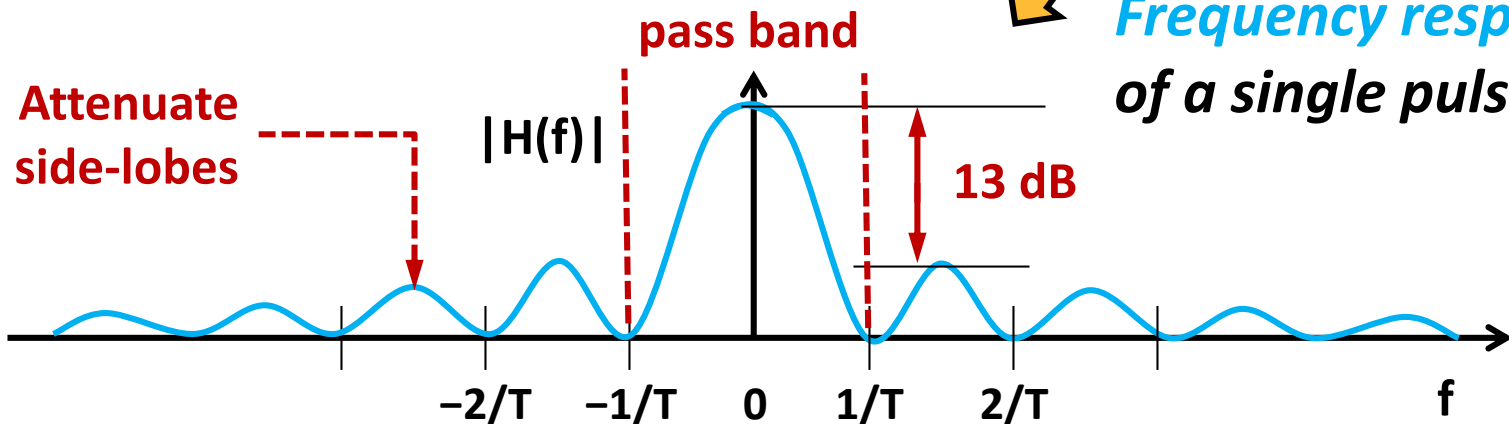
Signal Bandwidth Limitation

- **Modulation generates pulse train of zeros and ones**
 - Its frequency response is not band-limited
 - Filter before Tx to restrict bandwidth of symbols

*Binary sequence
generated after
modulation*



*Frequency response
of a single pulse*

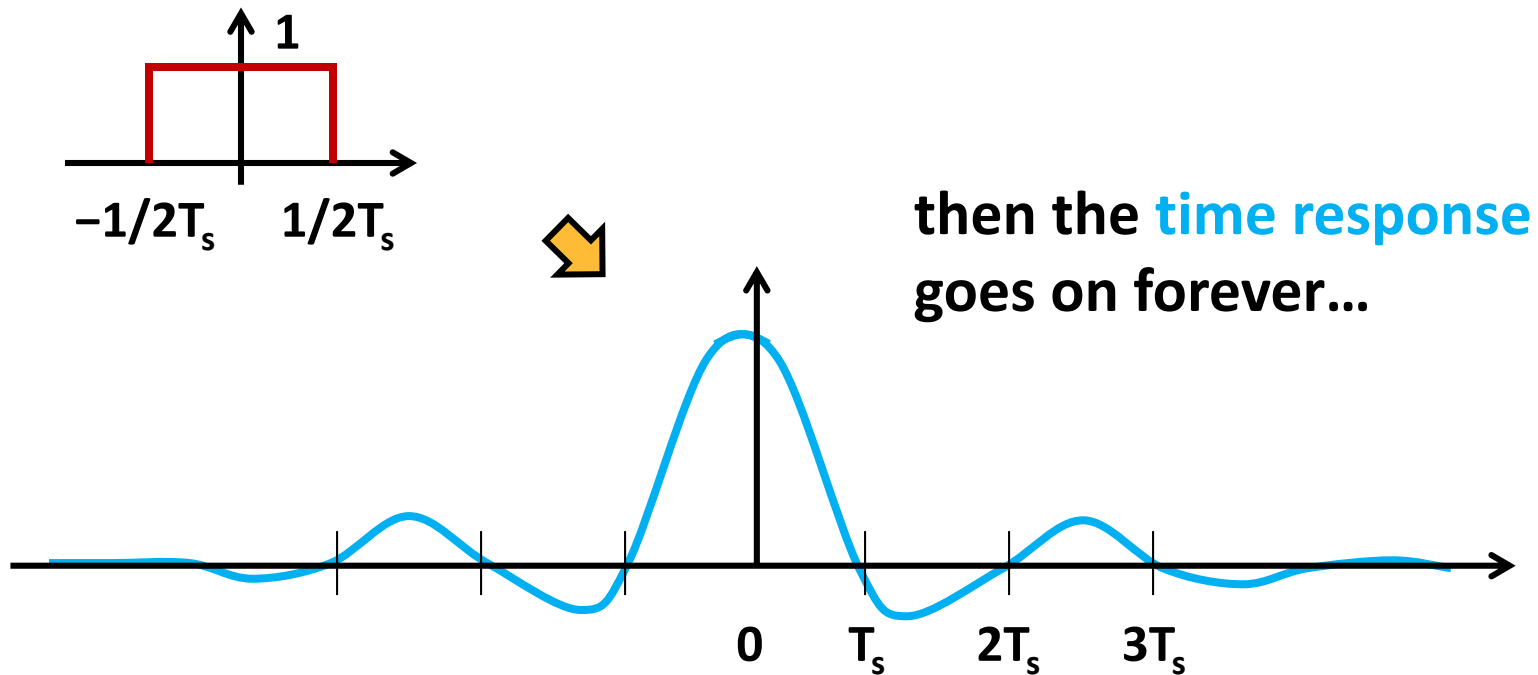


Ideal Filter

Sample rate = f_s

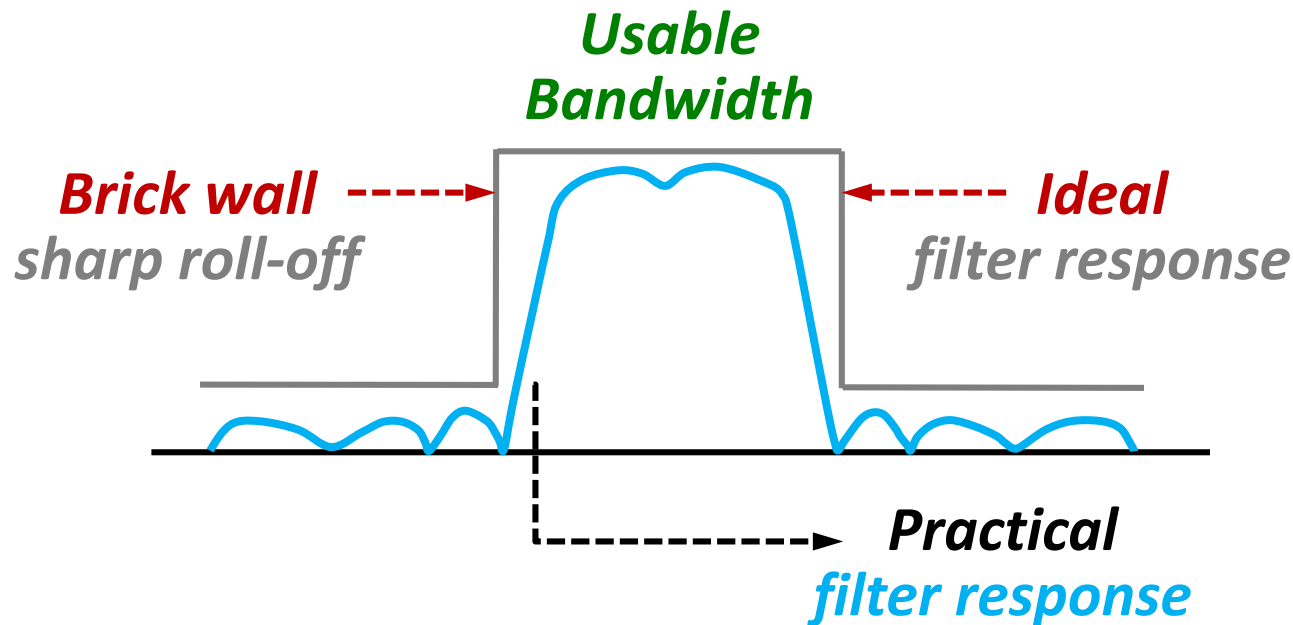
Baseband BW = $f_s/2$ (pass band BW = f_s)

- If we band-limit to the minimum possible amount $1/2T_s$,



Practical Transmit / Receive Filters

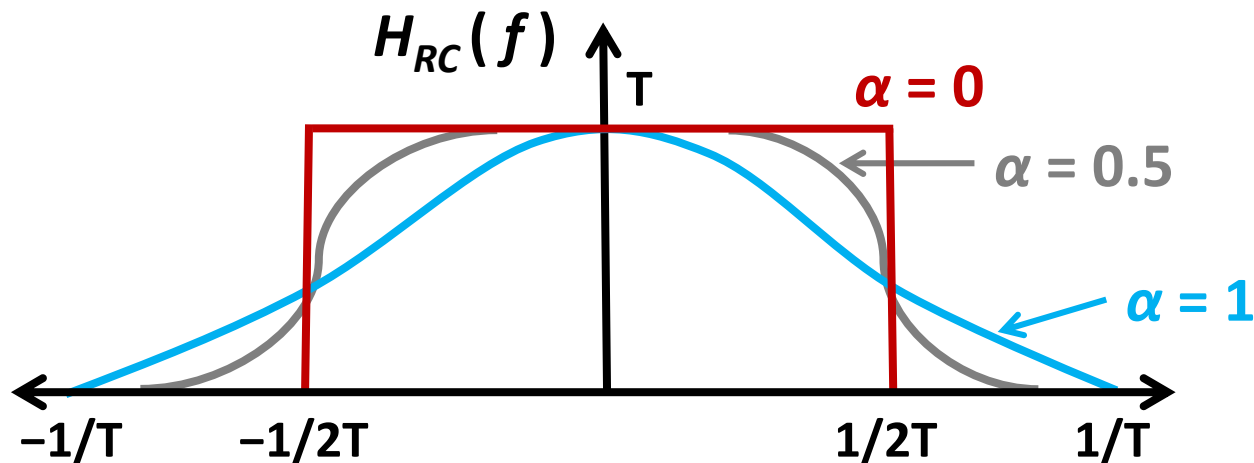
- **Tx:** restrict the Tx signal BW to a specified value
- **Rx:** extract the signal from a specified BW of interest



Ideal filter has infinitely long impulse response.
Raised-cosine filter is a practical realization.

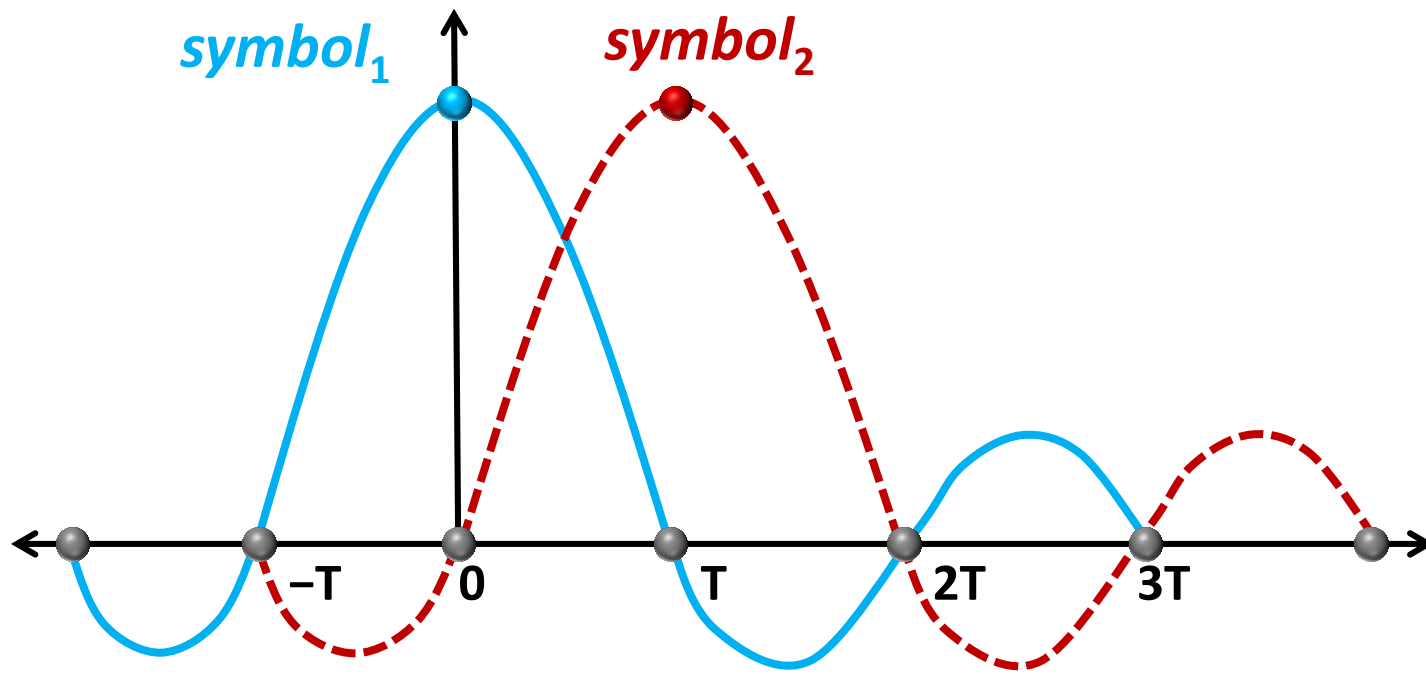
Raised-Cosine Filter: Frequency Response

$$H_{RC}(f) = \begin{cases} 1 \Leftrightarrow |f| \leq \frac{1-\alpha}{2T_s} \\ \frac{T_s}{2} \left[1 + \cos \frac{\pi T_s}{\alpha} \left(|f| - \frac{1-\alpha}{2T_s} \right) \right] \Leftrightarrow \frac{1-\alpha}{2T_s} < |f_s| < \frac{1+\alpha}{2T_s} \\ 0 \Leftrightarrow |f| > \frac{1+\alpha}{2T_s} \end{cases}$$



Raised-Cosine Filter: Time Response

- Time-domain pulse shaping with raised-cosine filters
- No contribution of adjacent symbols at the $k \cdot T_s$ instants
 - No inter-symbol interference with this pulse shaping



Square-Root **Raised-Cosine Filter**

**Split the filter
between Tx & Rx**

$$H_{RC}(f) = H_{Tx}(f) \cdot H_{Rx}(f)$$

$$H_{Tx}(f) = H_{Rx}(f) = \sqrt{H_{RC}(f)}$$

- **Choose sample rate for the filter:**
 - f_{sample} equal to D/A frequency on the Tx side and A/D frequency on the Rx side
 - If $f_{D/A} = f_{A/D} = 4 \cdot (1/T_{\text{symbol}})$,
- ⇒
$$h_{Tx}(n) = \sum \sqrt{H_{RC}(4m / NT_s)} \cdot e^{j(2\pi mn)/N}$$

for $-(N-1)/2 \leq n \leq (N-1)/2$

Impulse response is finite: implement as FIR filter

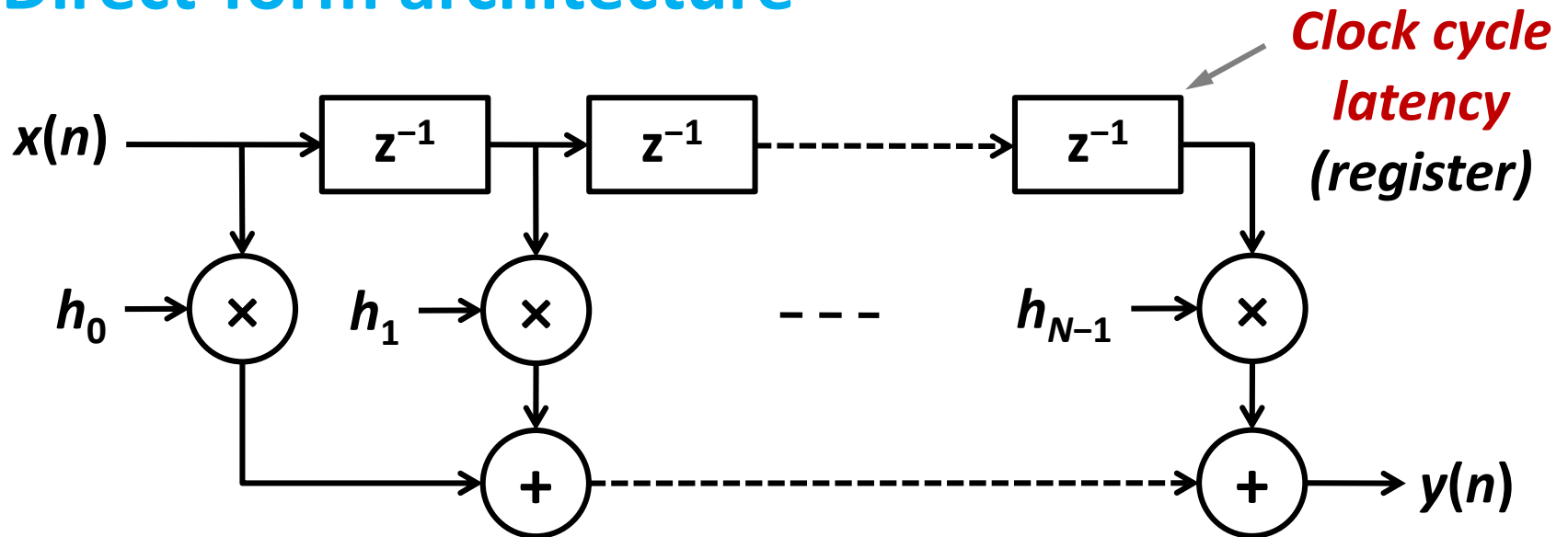
Direct (FIR) Filters

Implementing the Filter

$$y(n) = h_0x(n) + h_1x(n-1) + h_2x(n-2) + \dots + h_{N-1}x(n-N+1)$$

- An N -tap filter is a series of *multiply* and *add* operations

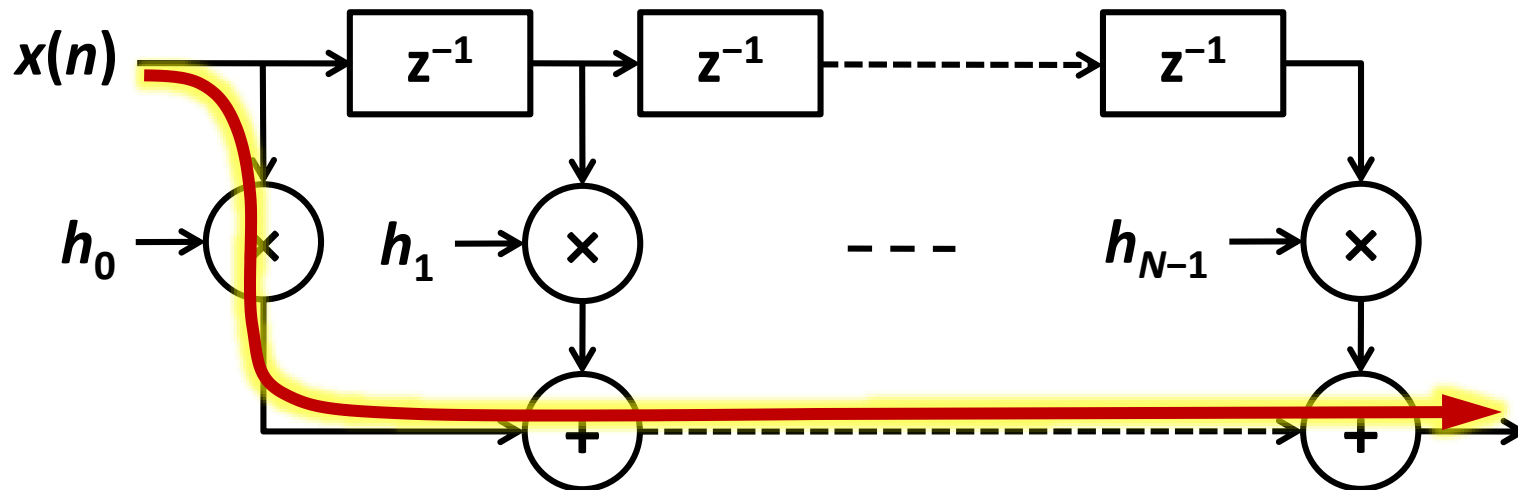
Direct-form architecture



Direct-Form FIR Filter: **Critical Path**

$$\text{Critical path} = t_{mult} + (N-1) \cdot t_{add}$$

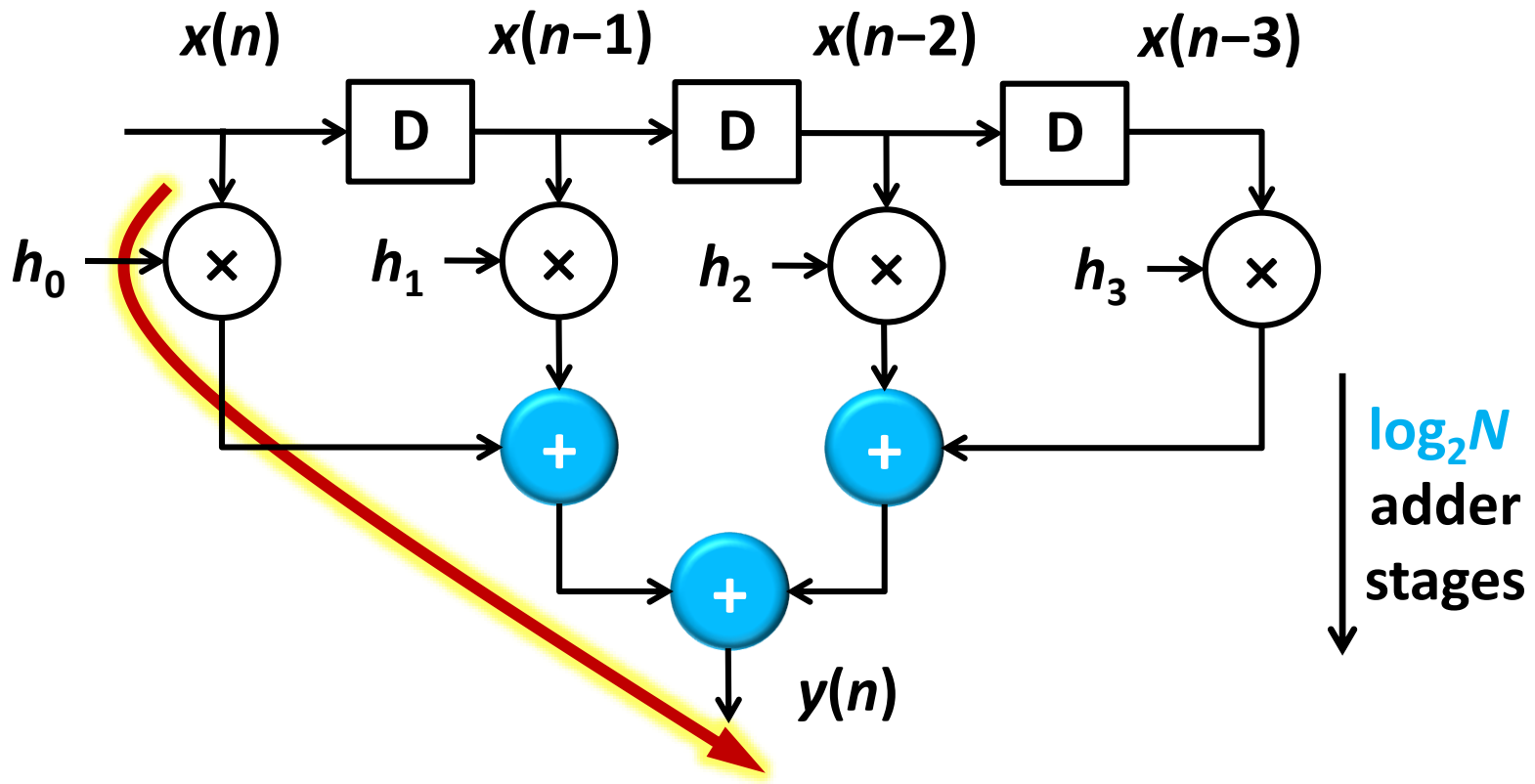
- **Critical-path delay proportional to filter order**
 - Suitable for small number of taps



Multi-Operand Addition

$$\text{Critical path} = t_{mult} + \log_2(N) \cdot t_{add}$$

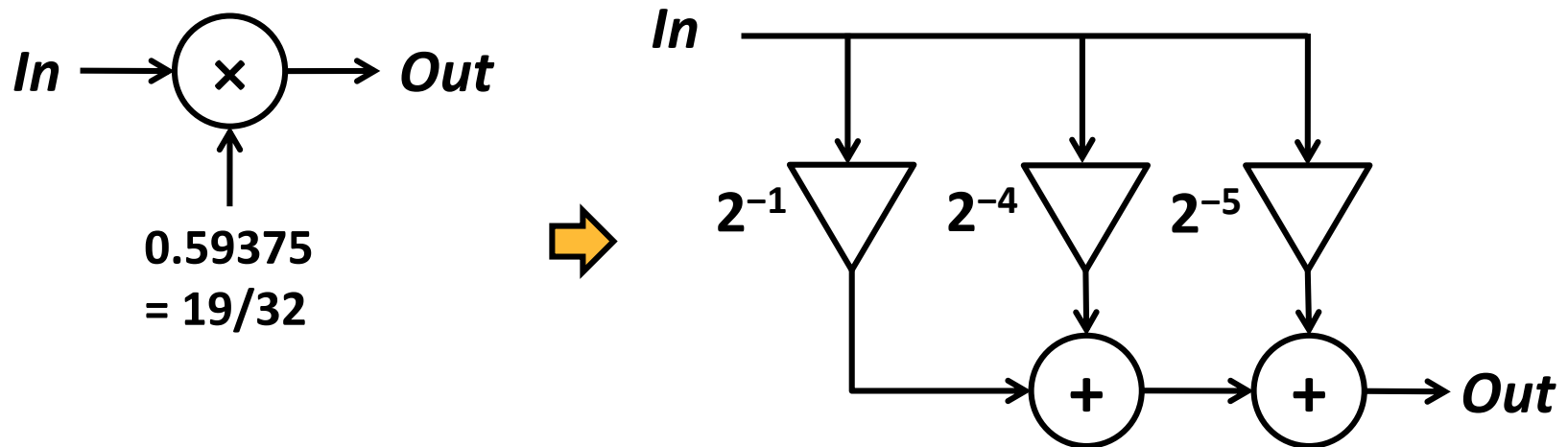
instead of $N - 1$



Use an **adder tree** instead of a chain

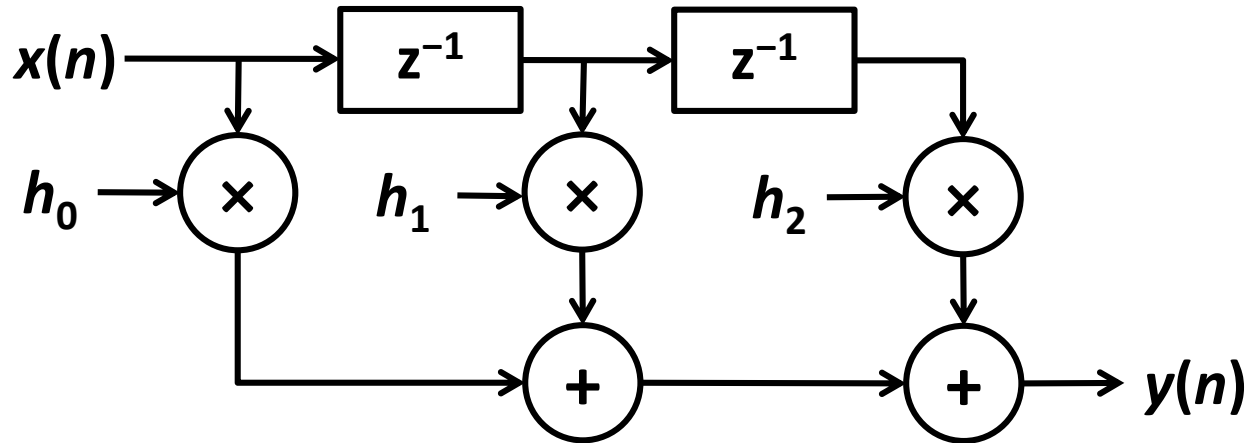
Multiplier-less FIR Filter Implementation

- **Low-complexity power-of-two based multiplications**
 - Obtained for free by simply shifting data buses
 - Round off multiplier coefficients to nearest power of 2
 - Small performance degradation in most cases

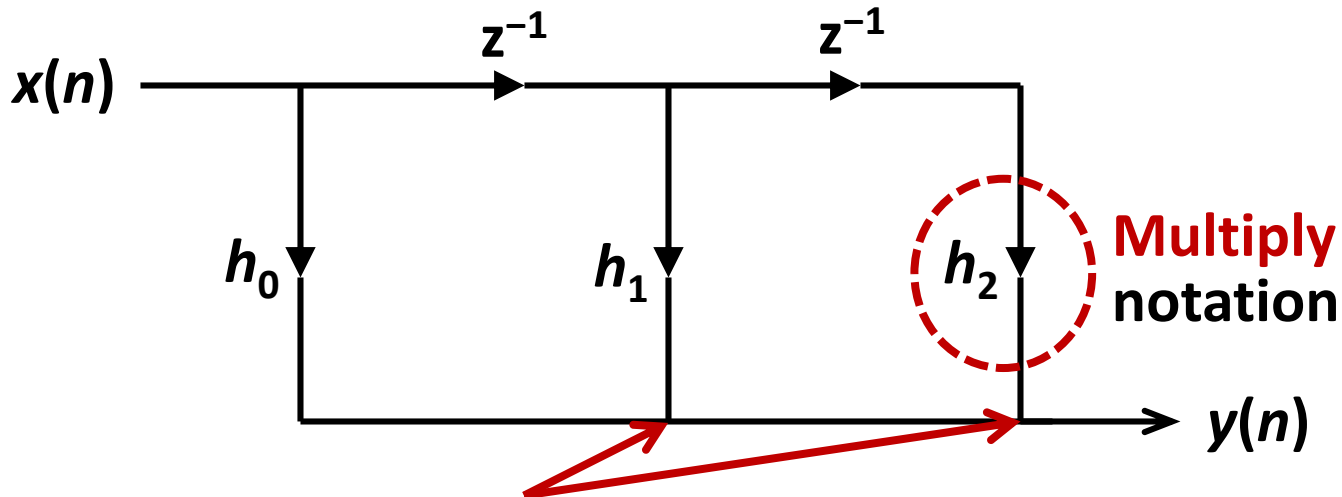


H. Samueli, "An Improved Search Algorithm for the Design of Multiplierless FIR Filters with Powers-of-Two Coefficients," *IEEE Trans. Circuits and Systems*, vol. 36, no. 7, pp. 1044-1047, July 1989.

FIR Filter: Simplified Notation



A more abstract  *and efficient notation*



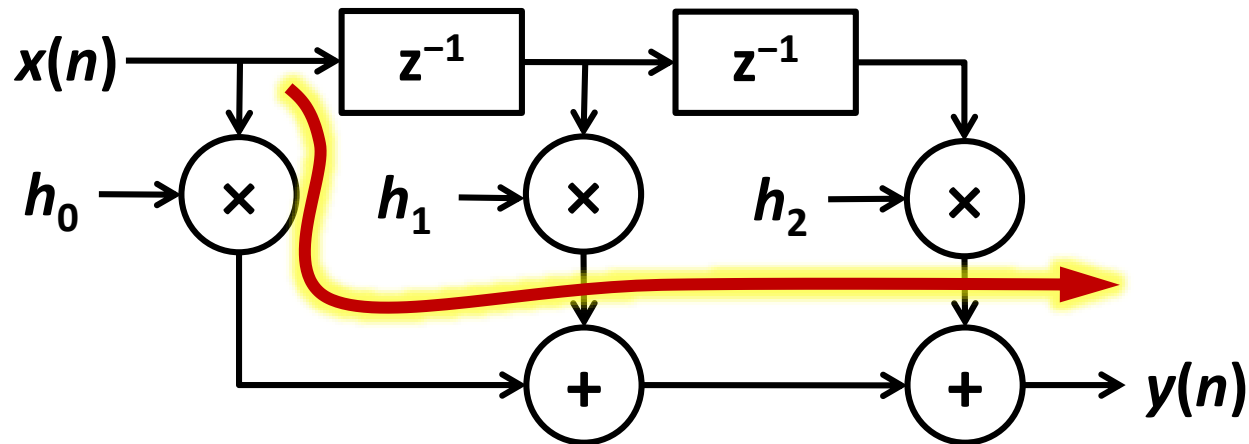
Add when branches merge

Pipelining

- **Direct-form architecture is throughput-limited**
 - Pipelining can be used to increase throughput
- **Pipelining: adding the same number of delay elements in each forward cut-set**
 - **Cut-set**: set of edges in a graph that if removed, graph becomes disjoint
 - **Forward cut-set**: all edges in the cut-set are in the same direction
- **Increases latency**
- **Register overhead (power, area)**

Example 9.1: Pipelining FIR Filter

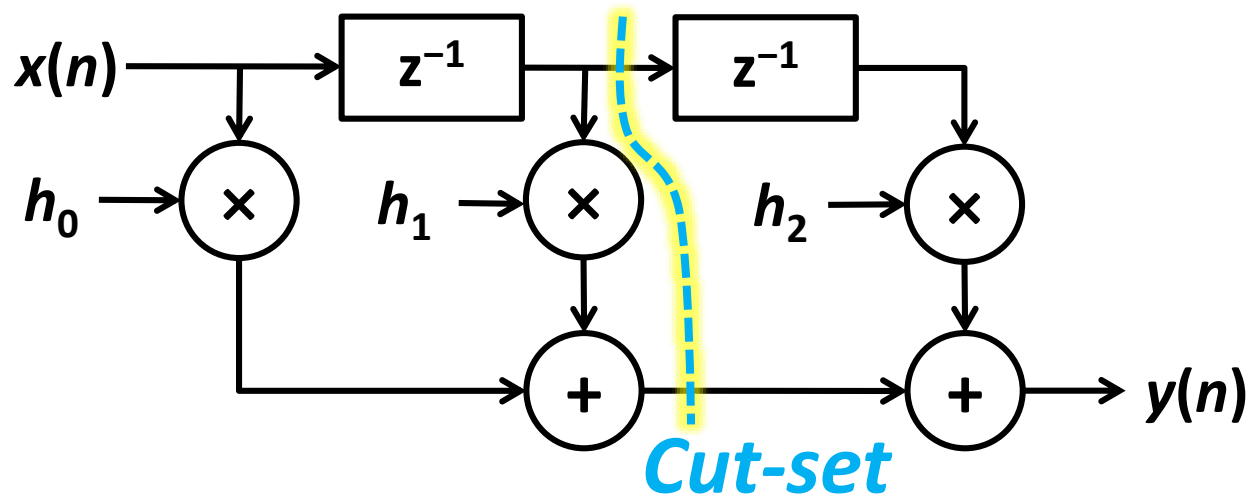
- Calculate initial critical path



$$\text{Critical path} = t_{mult} + 2 \cdot t_{add}$$

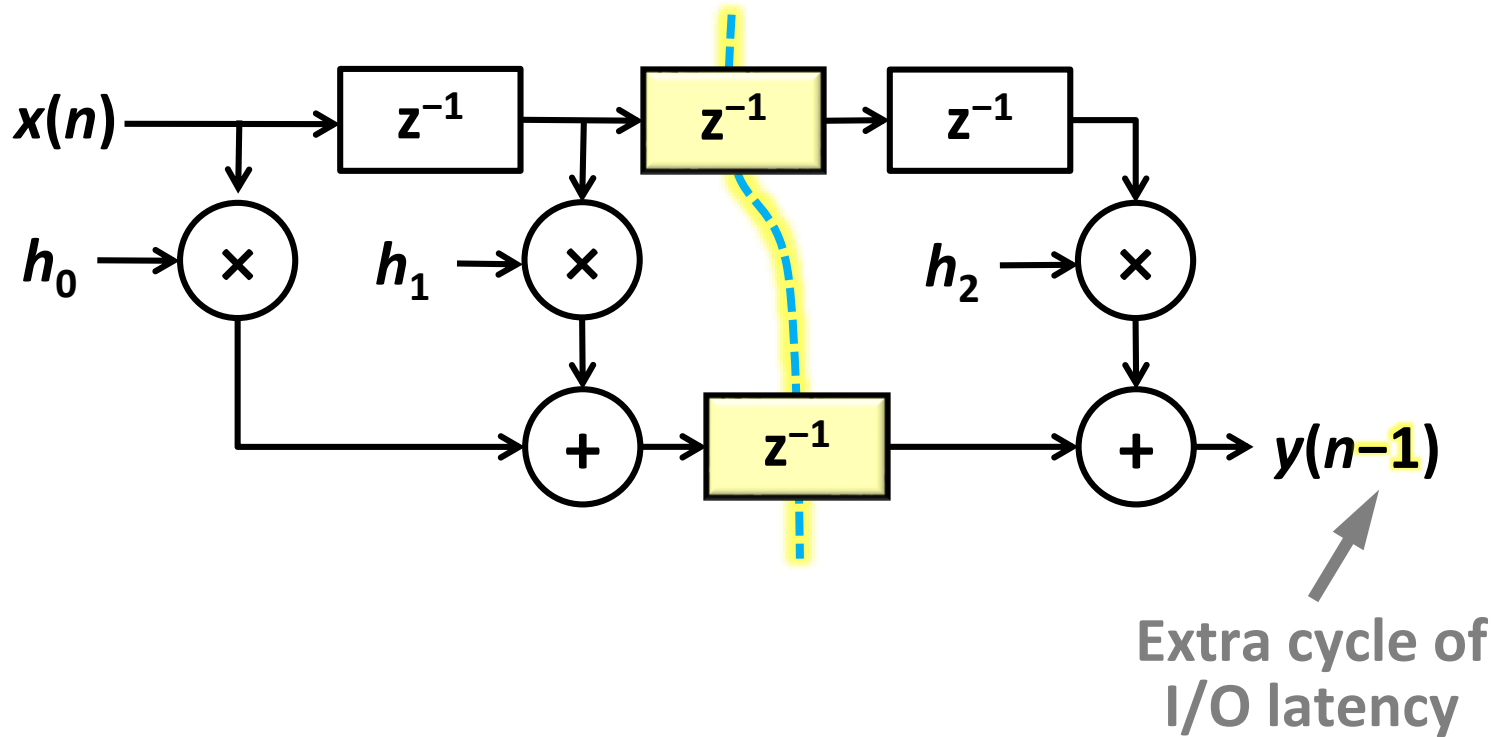
Example 9.1: Pipelining FIR Filter

- Find forward cut-set



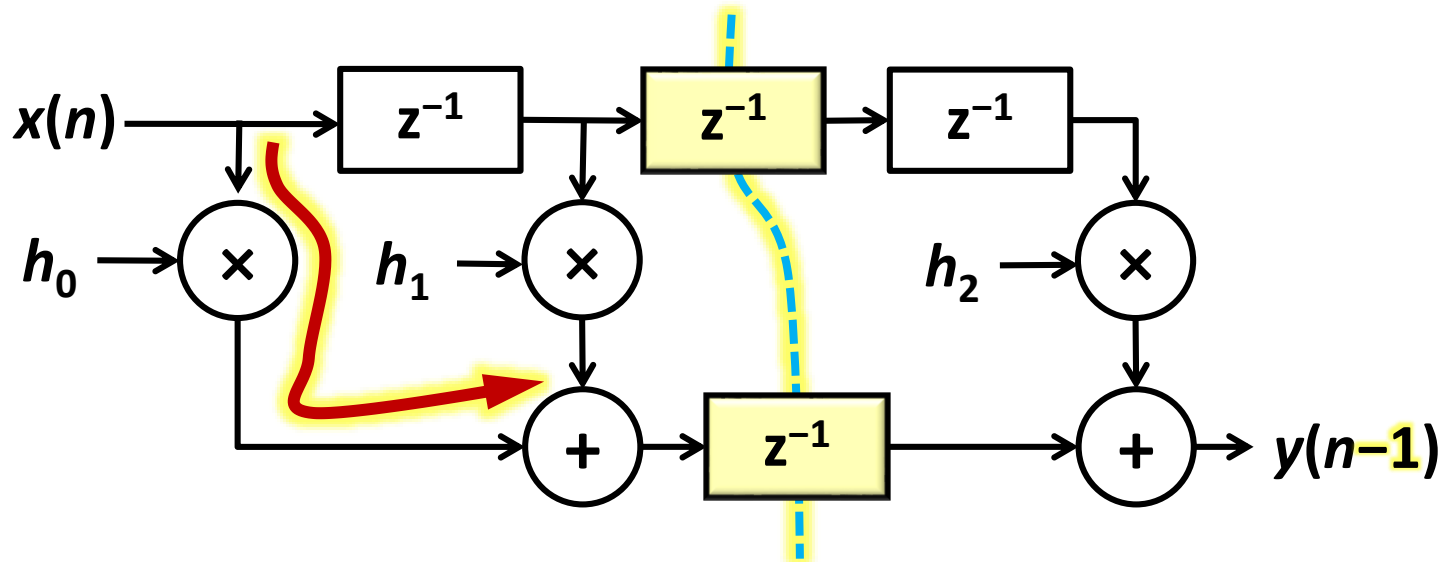
Example 9.1: Pipelining FIR Filter

- Insert **pipeline registers**
 - I/O latency increases



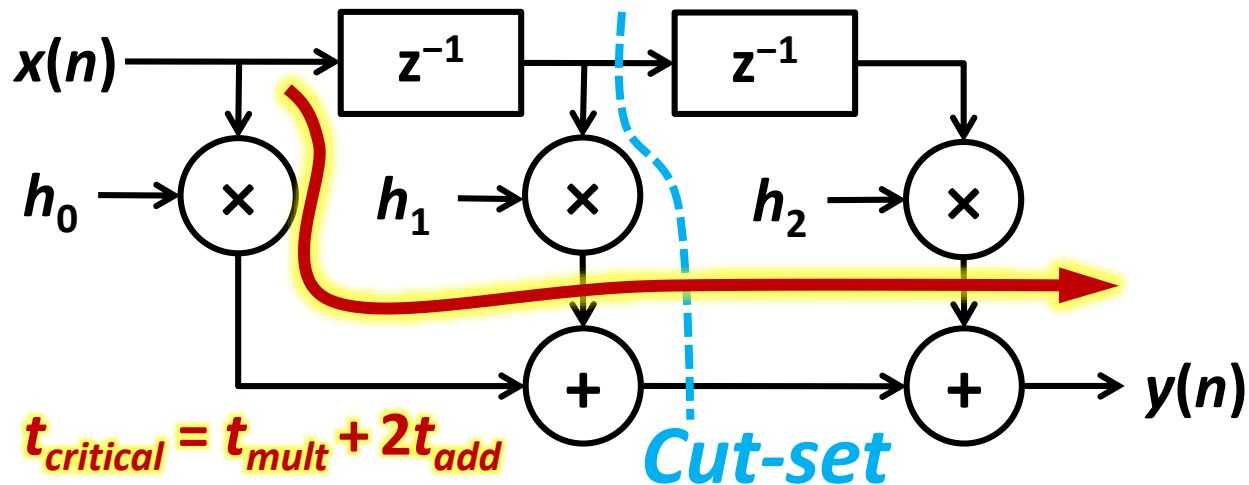
Example 9.1: Pipelining FIR Filter

- Calculate new critical path



$$\text{Critical path} = t_{mult} + t_{add}$$

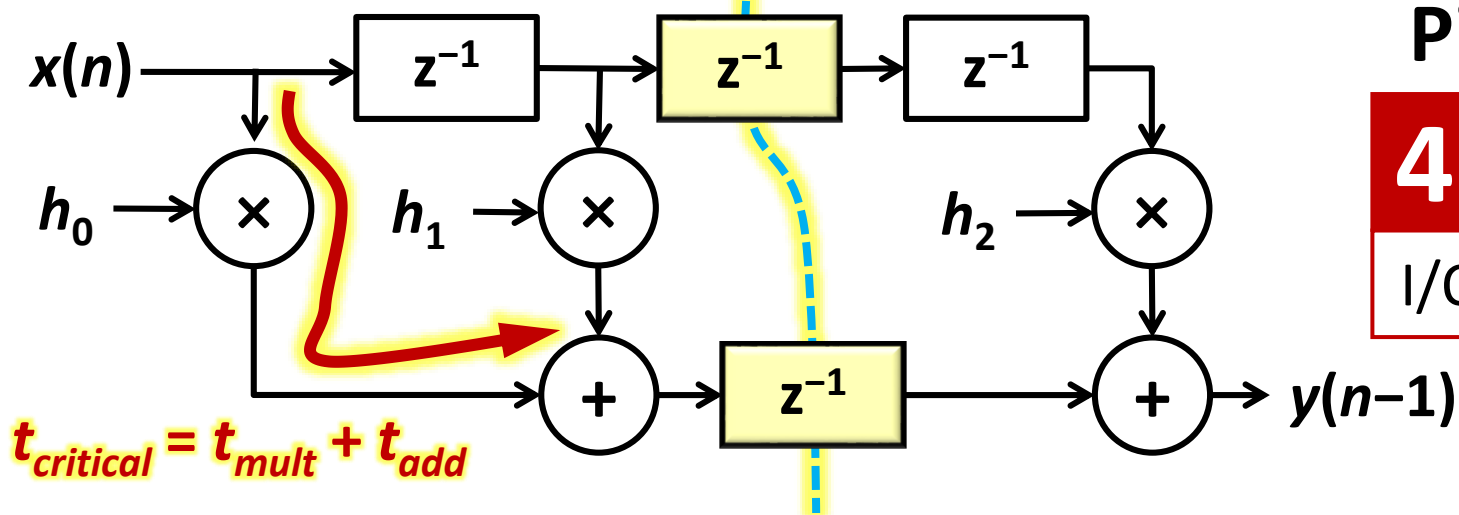
Example 9.1: Pipelining FIR Filter



Reference

2 registers

I/O latency: **0**



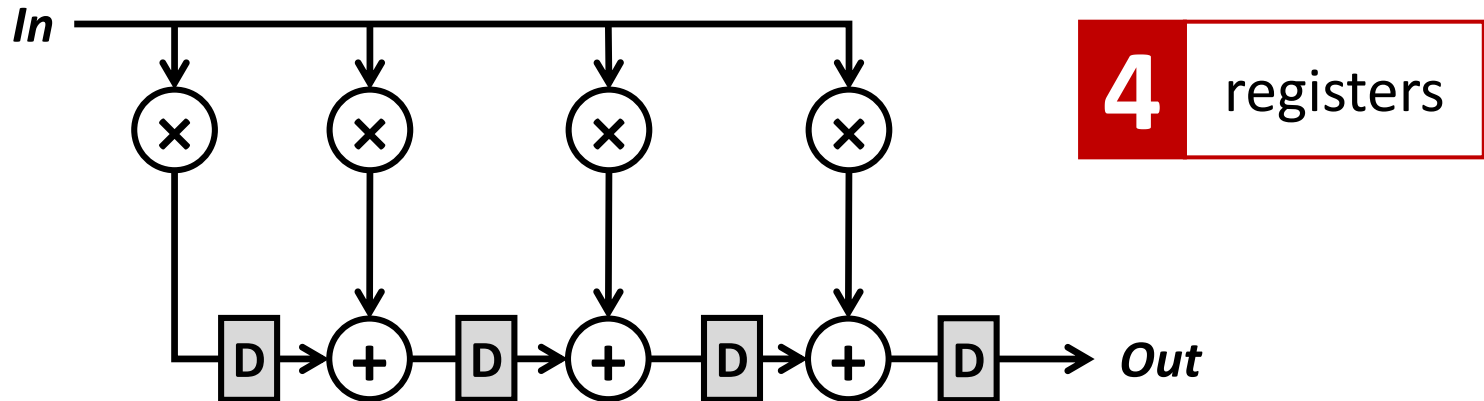
Pipelined

4 registers

I/O latency: **1**

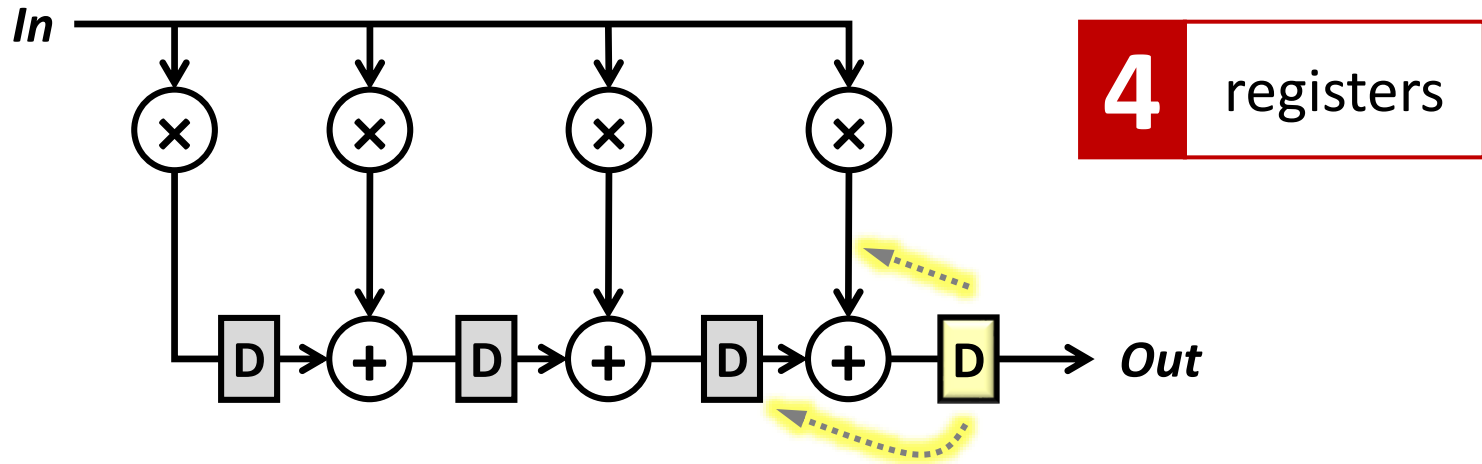
High-Level Retiming

- Optimal placement of **existing** data-path registers
 - Register movement does not alter functionality
- **Objective:** balance t_{critical} for maximum throughput



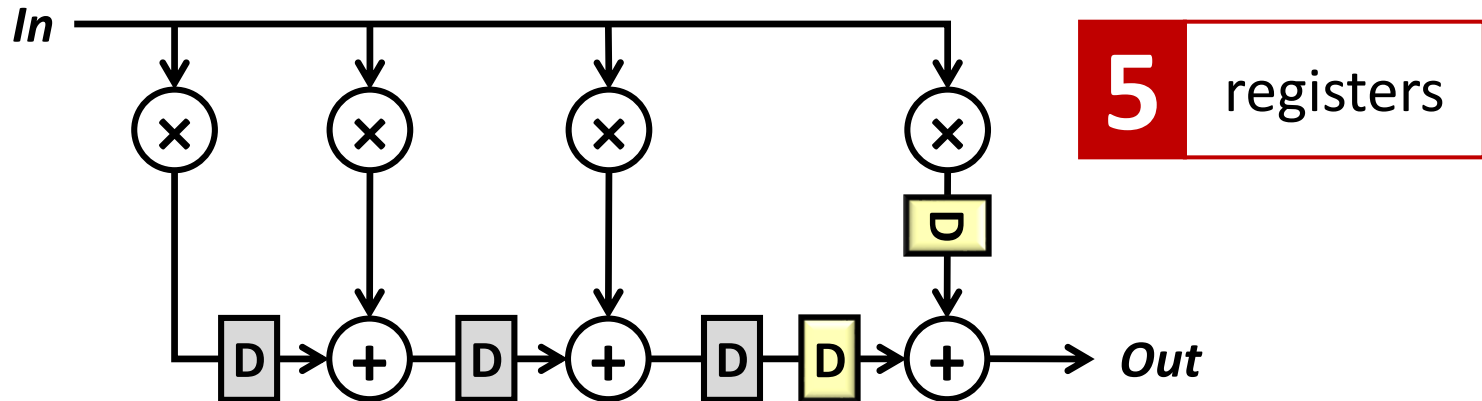
Example 9.2: Retiming of FIR Filter

- **Step 1:** move output register inside



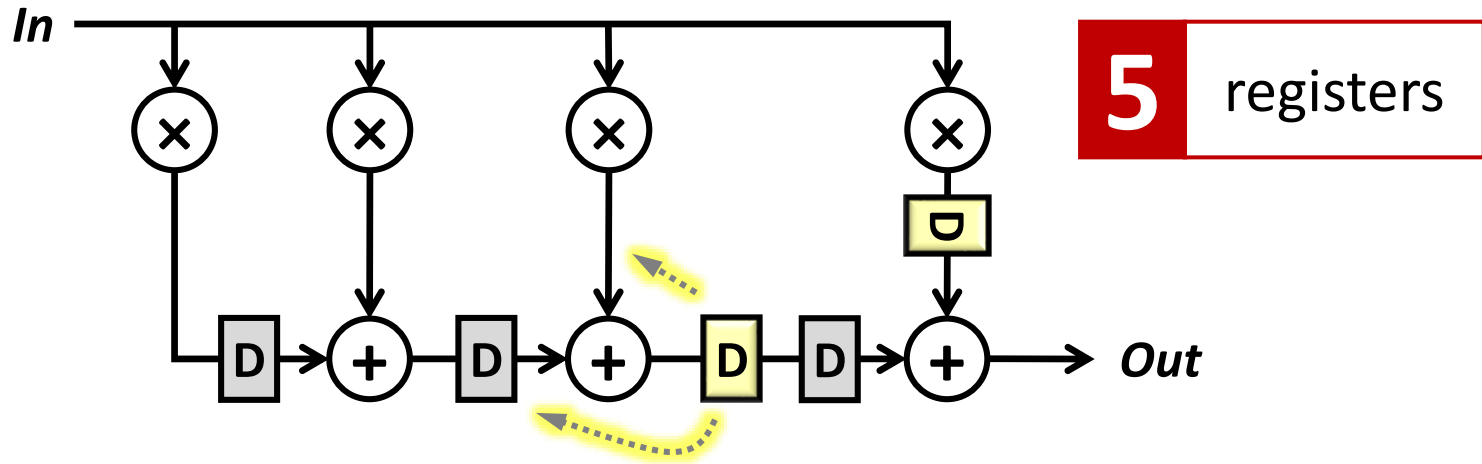
Example 9.2: Retiming of FIR Filter

- Output register splits into two input registers after the first step
 - Total number of registers increases



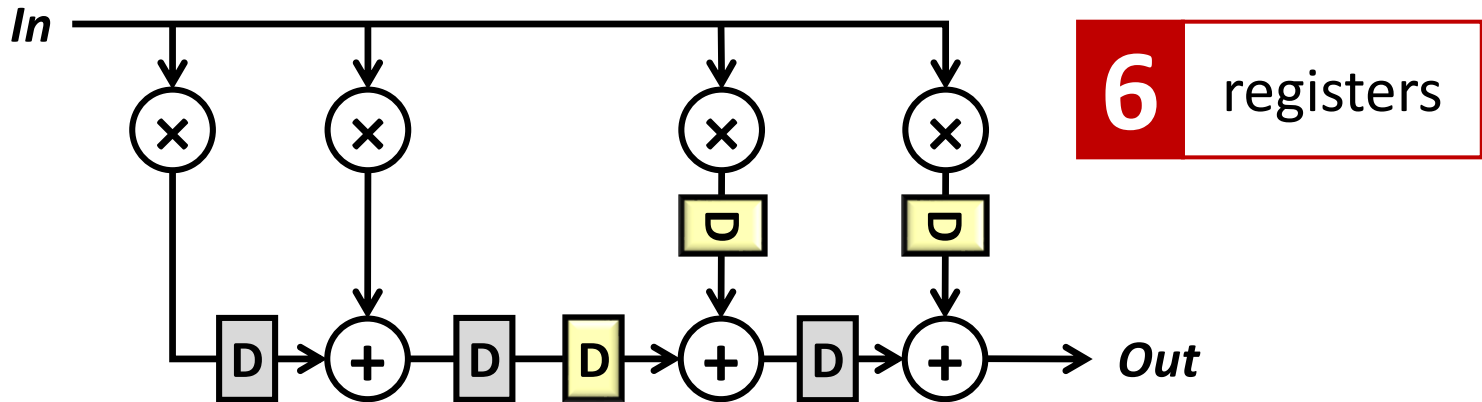
Example 9.2: Retiming of FIR Filter

- **Step 2:** move internal register further inside



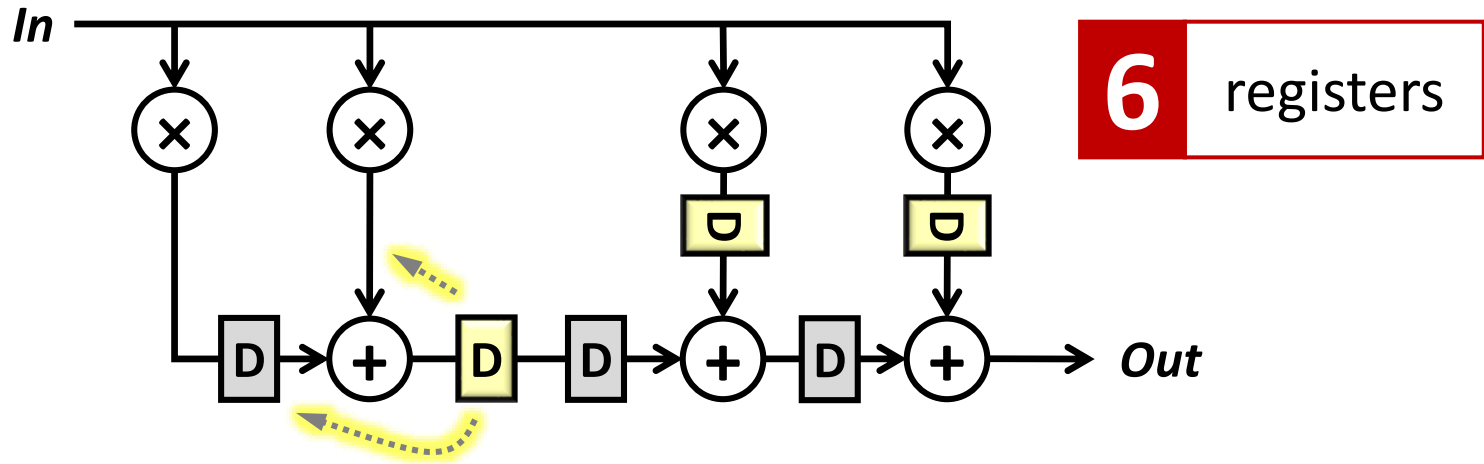
Example 9.2: Retiming of FIR Filter

- Internal register splits into two input registers after the second step
 - Total number of registers increases



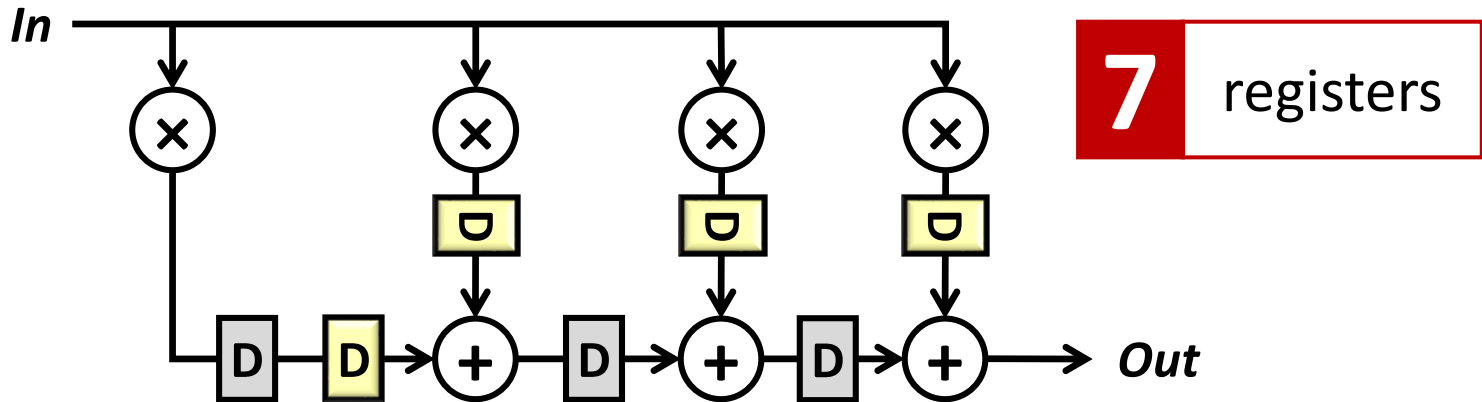
Example 9.2: Retiming of FIR Filter

- **Step 2+:** continue moving internal registers until you reach input



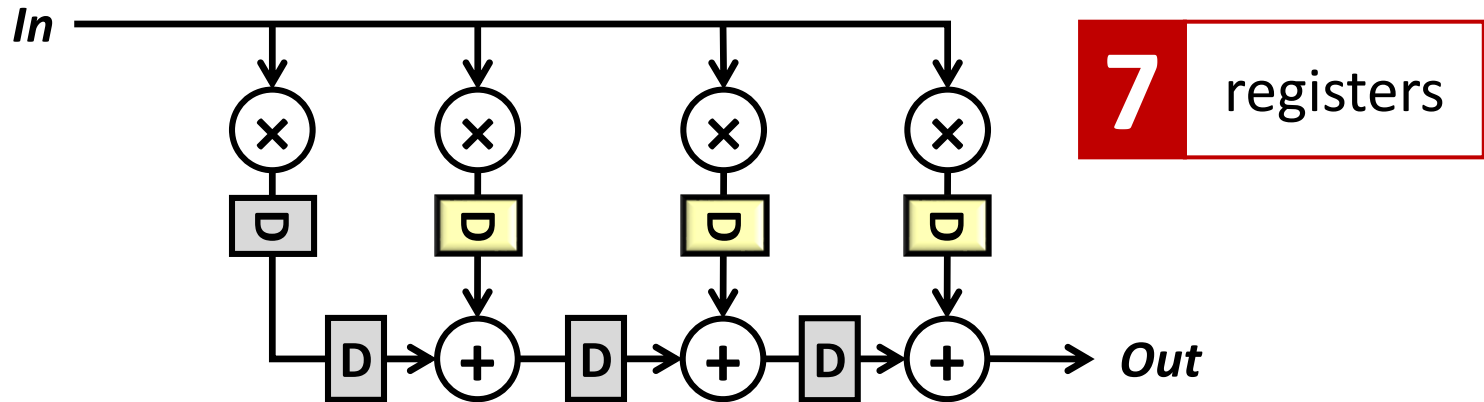
Example 9.2: Retiming of FIR Filter

- Each out/in split added an extra register
 - 4 filter stages: 3 additional registers



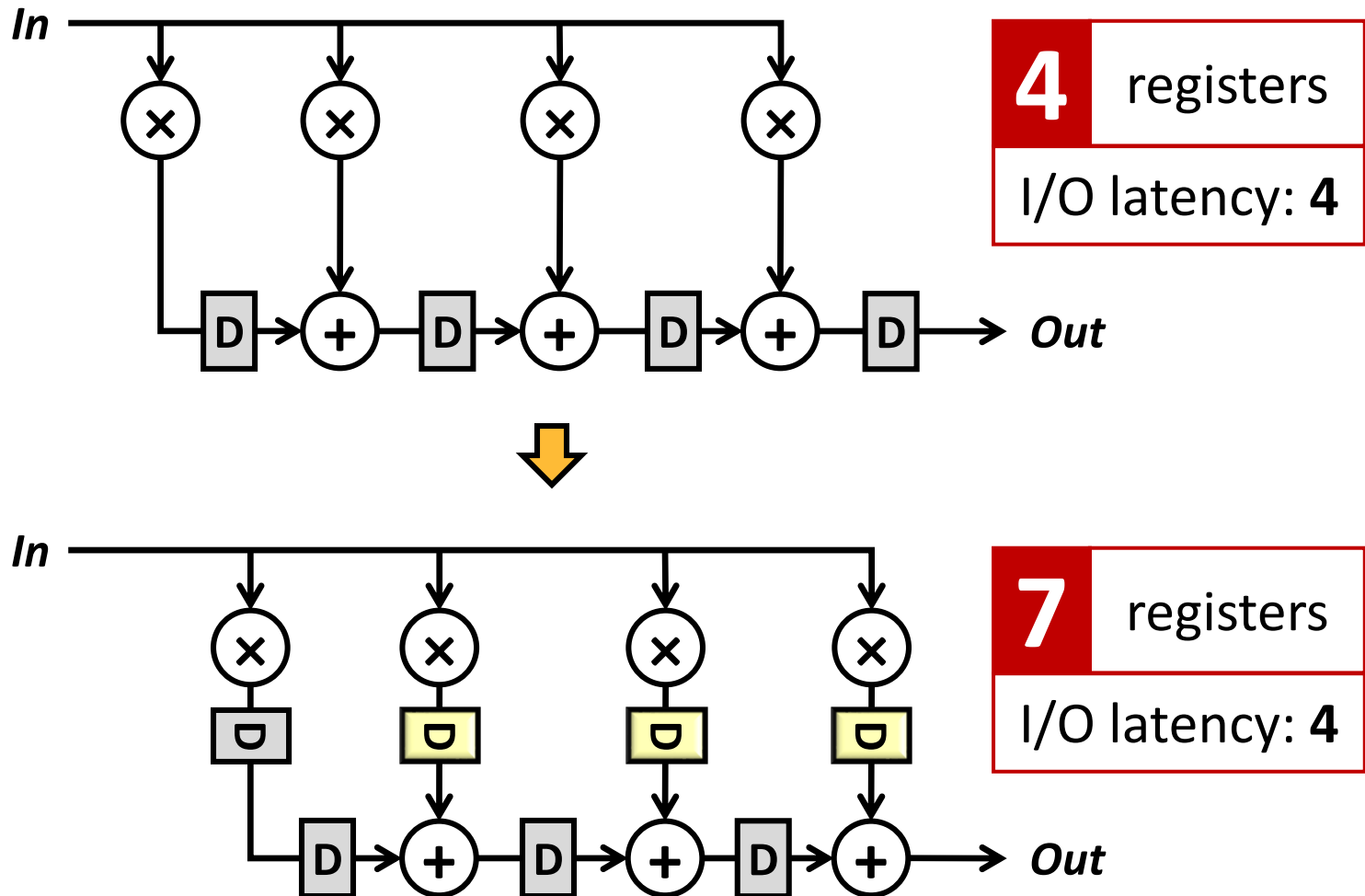
Example 9.2: Retiming of FIR Filter

- Each out/in split added an extra register
 - 4 filter stages: 3 additional registers



High-Level Retiming

Number of registers increased, I/O latency unchanged

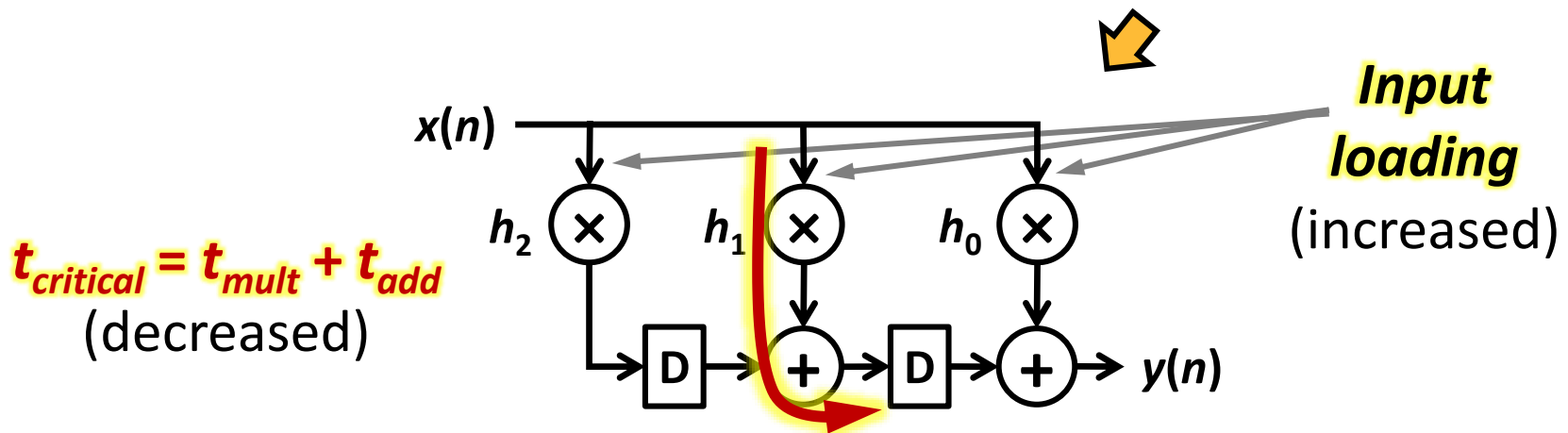
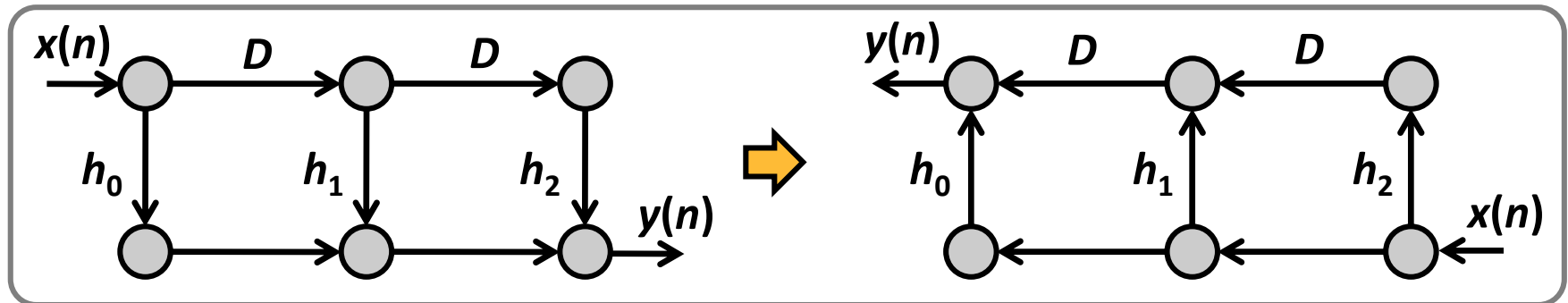


Pipelining vs. Retiming

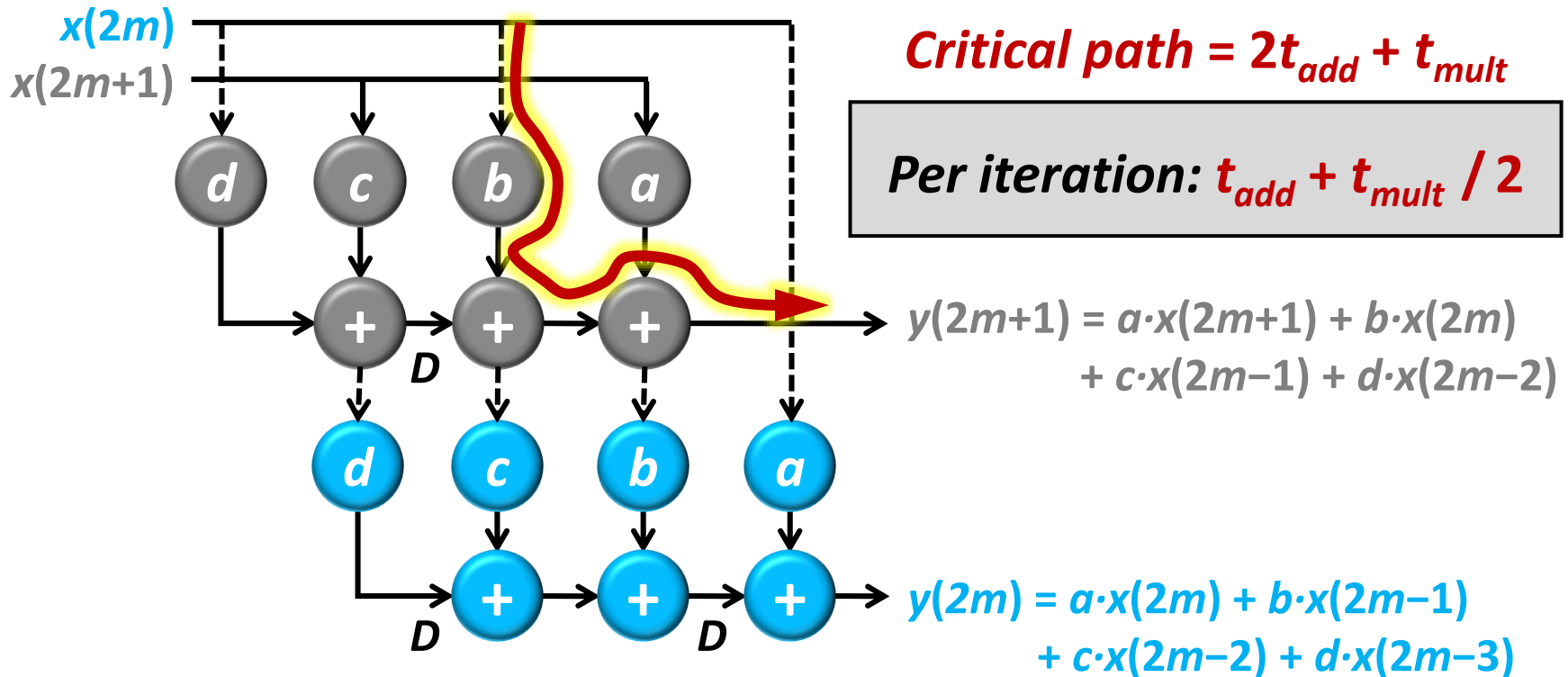
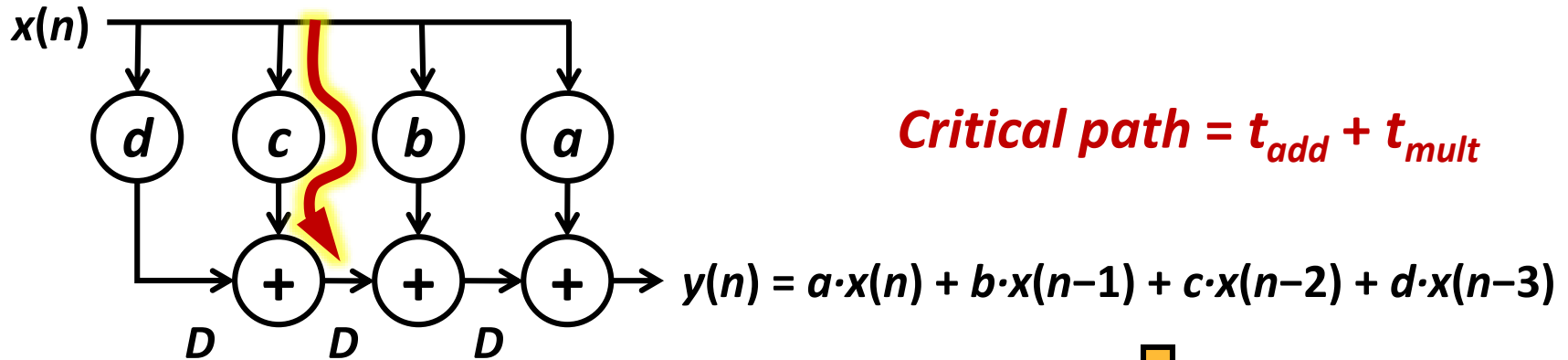
- **Pipelining**
 - Inserts registers into cut-sets
 - I/O latency increases
- **Retiming**
 - Moves existing registers
 - I/O latency unchanged

Transposing FIR

- Reverse the direction of edges in a signal-flow graph
- Interchange the input and output ports
- Functionality unchanged



Transposed + Parallel FIR



FIR Summary

Main features:

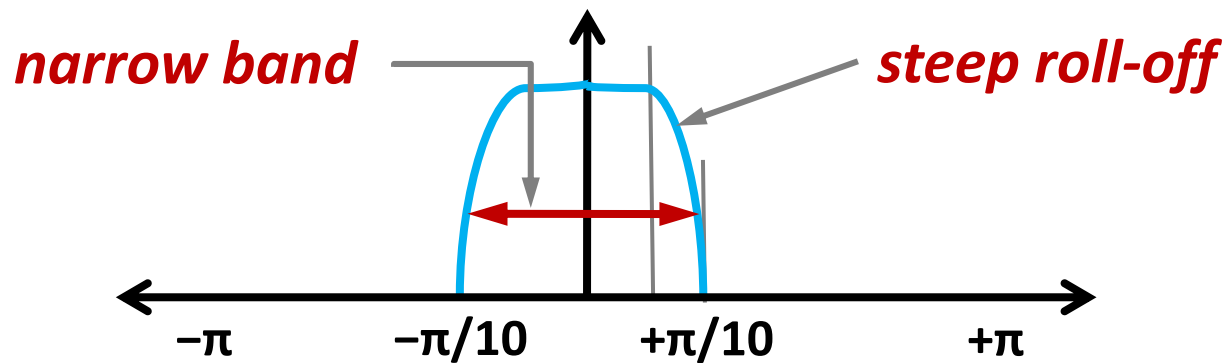
- Easy to design, always stable
- Feed-forward: can be pipelined, parallelized
- Linear phase response

Realize narrow band, steep roll-off?

- FIR filters require large number of taps
- Area and power cost can make FIR unsuitable

Recursive (IIR) Filters

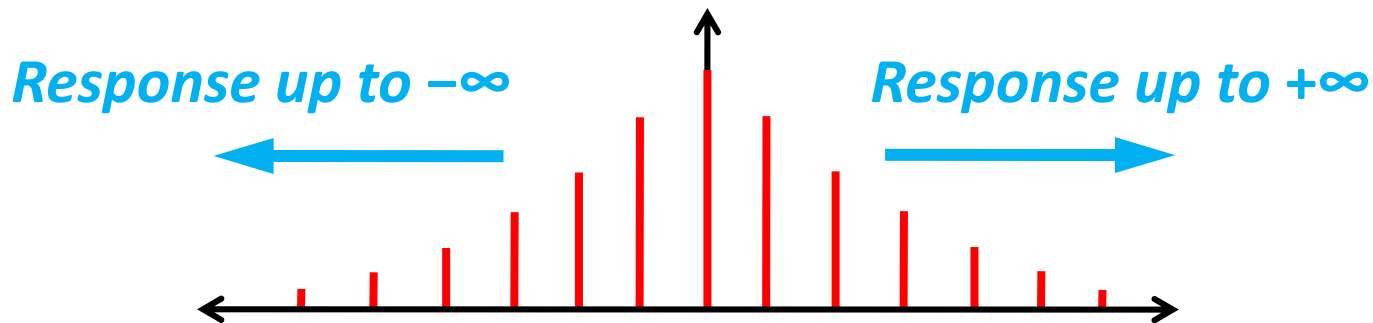
IIR Filters for Narrow Band, Steep Roll-Off



Infinite impulse response (IIR) filters are more suited to achieve such a frequency response with low area, power

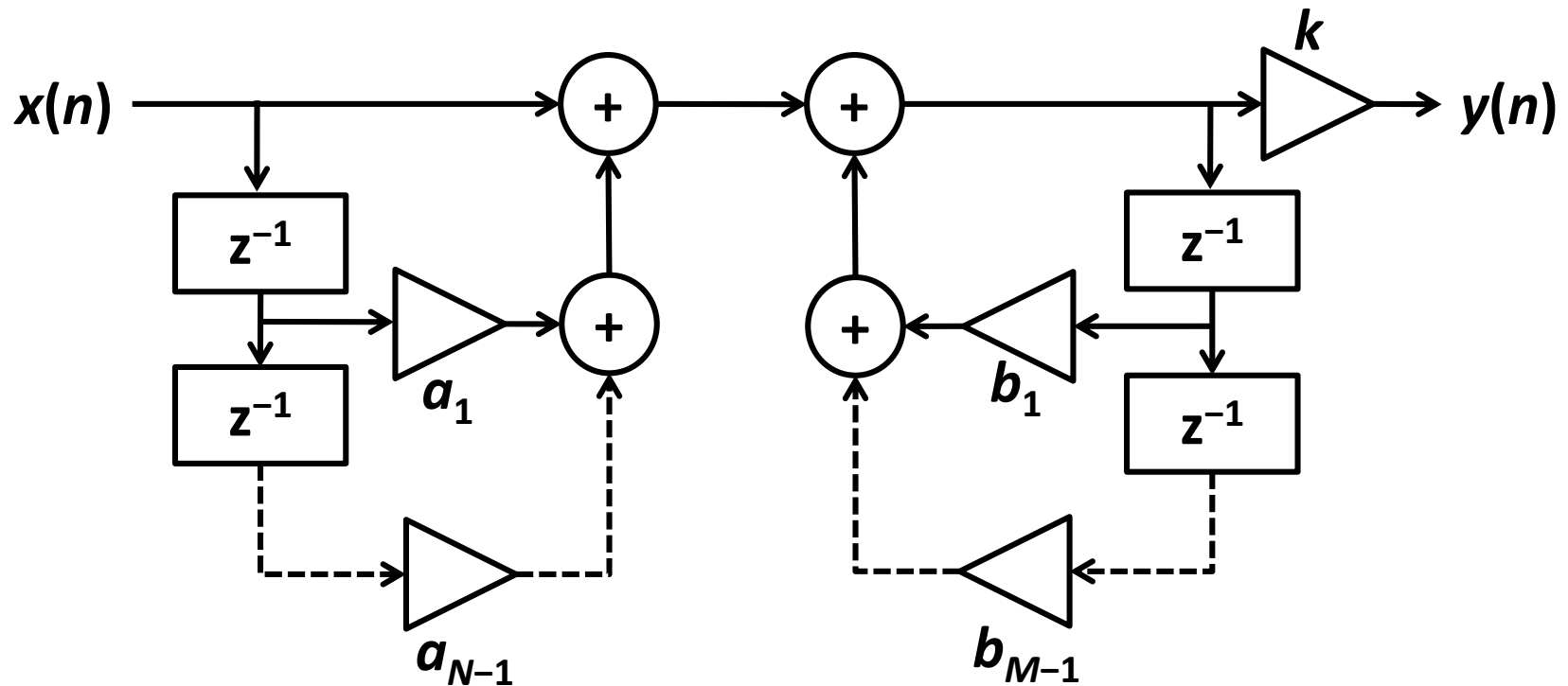
Generalized IIR Transfer Function

$$y(n) = \underbrace{\sum_{m=1}^N b_m y(n-m)}_{\text{Feedback}} + \underbrace{\sum_{p=0}^N a_p x(n-p)}_{\text{Feed-forward}}$$



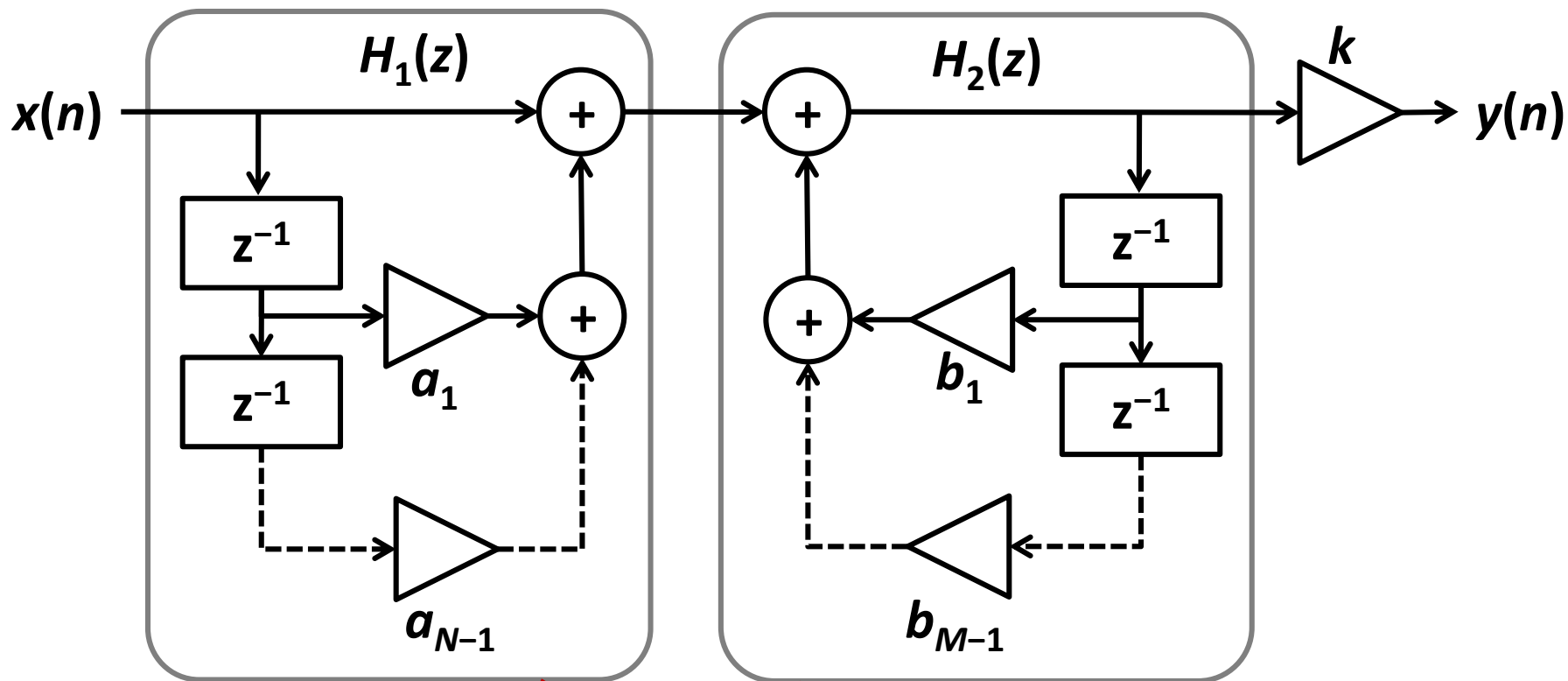
Direct-Form IIR Architecture

$$H(z) = k \frac{1 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_{N-1} z^{-(N-1)}}{1 - b_1 z^{-1} - b_2 z^{-2} - \dots - b_{M-1} z^{-(M-1)}}$$



IIR Architecture Optimization

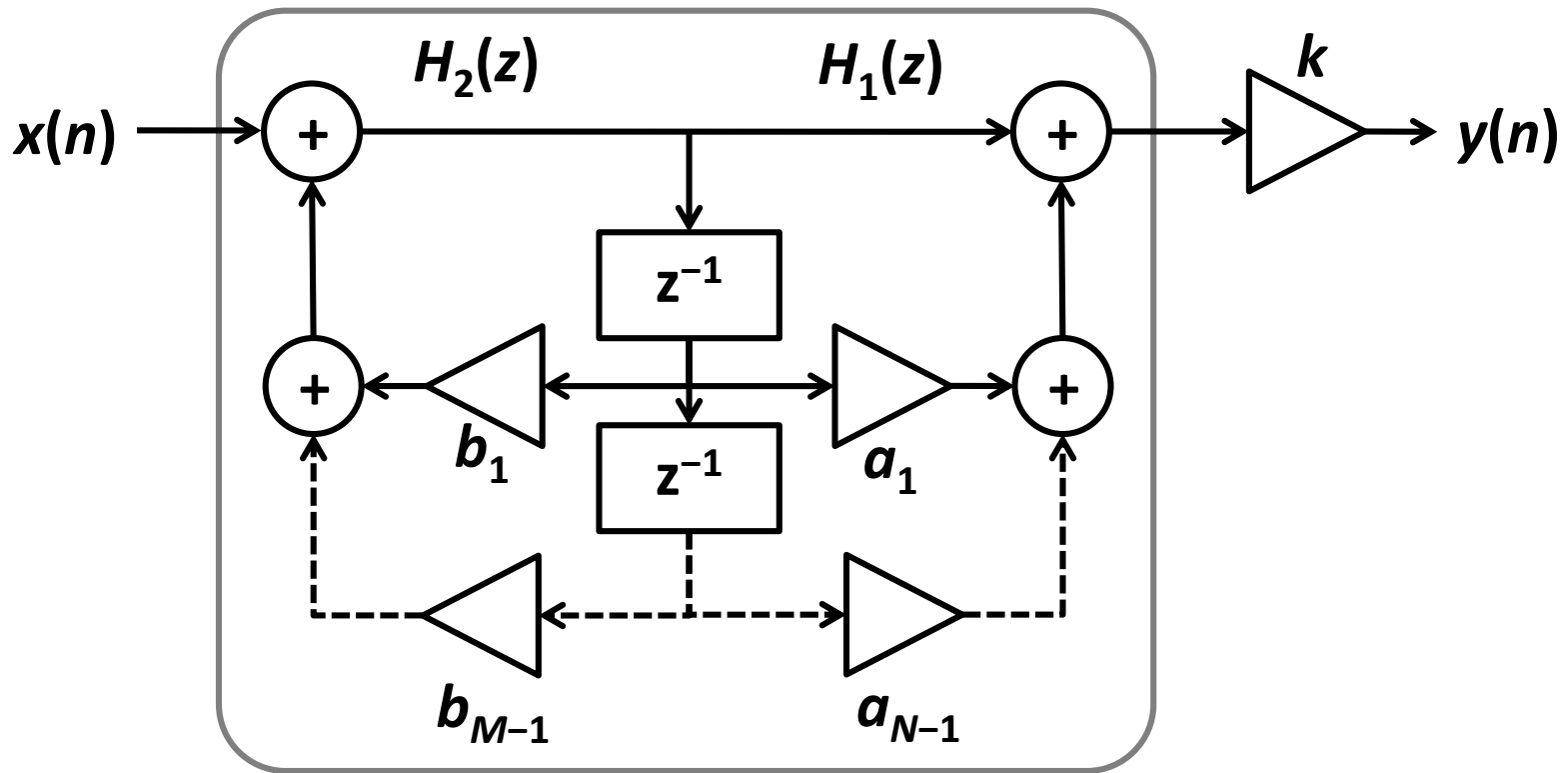
Functionality unchanged if H_1 , H_2 are swapped



Swap the order of execution

IIR Architecture Optimized

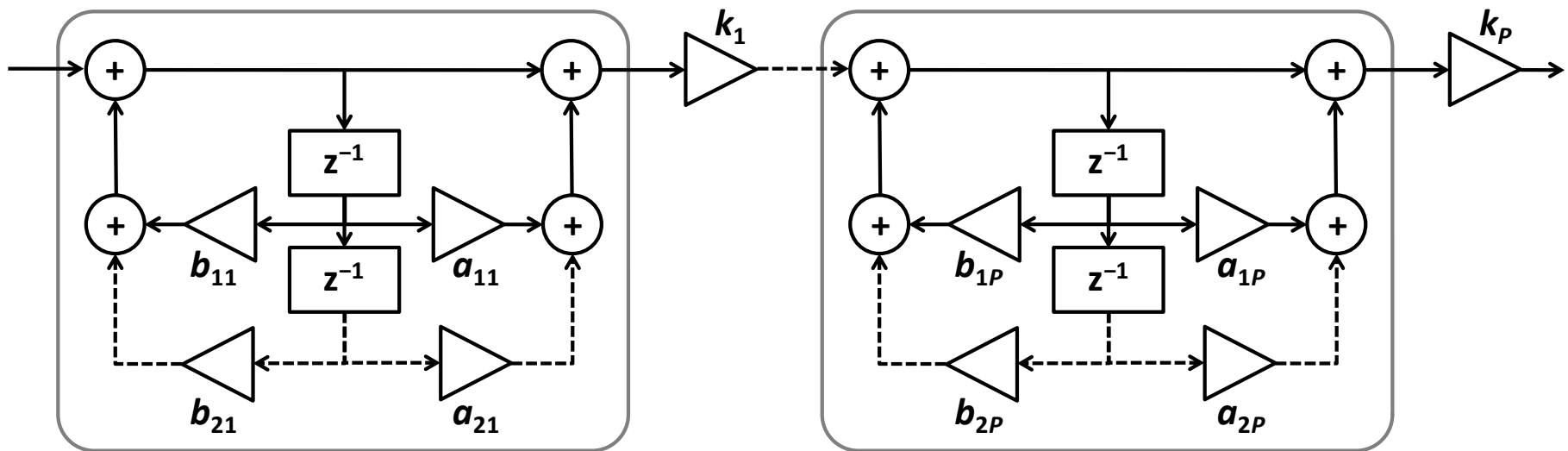
- H_1, H_2 can share the central register bank
 - If $M \neq N$, the central bank will have $\max(M, N)$ registers



Cascaded IIR

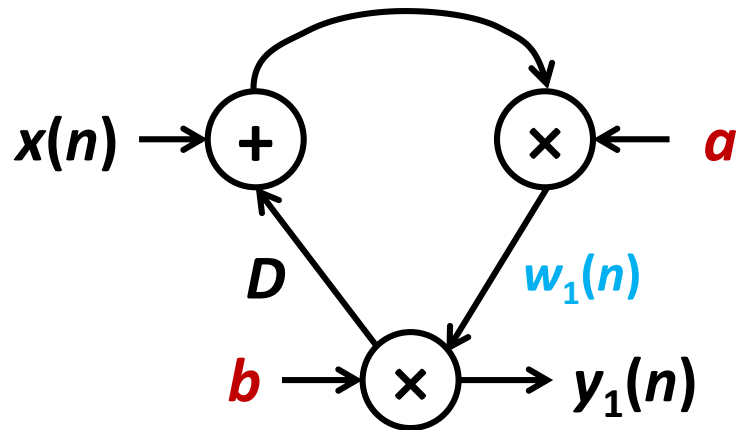
$$H(z) = k_1 \frac{1 + a_{11}z^{-1} + a_{21}z^{-2}}{1 - b_{11}z^{-1} - b_{21}z^{-2}} \dots k_p \frac{1 + a_{1p}z^{-1} + a_{2p}z^{-2}}{1 - b_{1p}z^{-1} - b_{2p}z^{-2}}$$

- **Implement IIR as cascade of 2nd order sections**
 - Shorter wordlengths in the feedback loop adders
 - More area and power efficient architecture

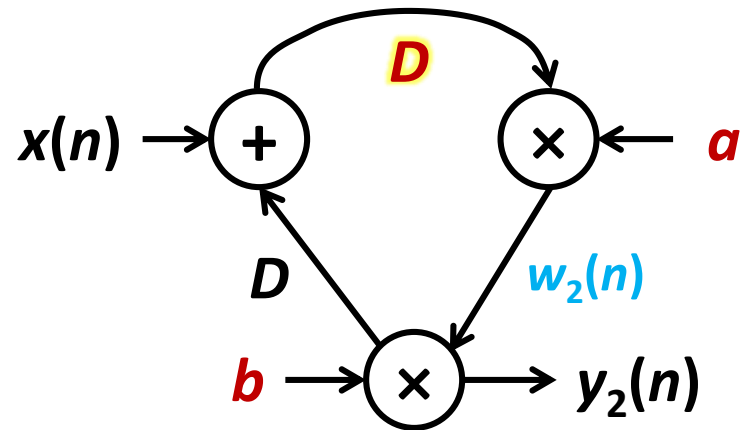


Recursive-Loop Bottlenecks

- Pipelining loops not possible
 - # registers in feedback loops must remain fixed



$$w_1(n) = a \cdot (y_1(n-1) + x(n))$$
$$y_1(n) = b \cdot a \cdot y_1(n-1) + b \cdot a \cdot x(n)$$



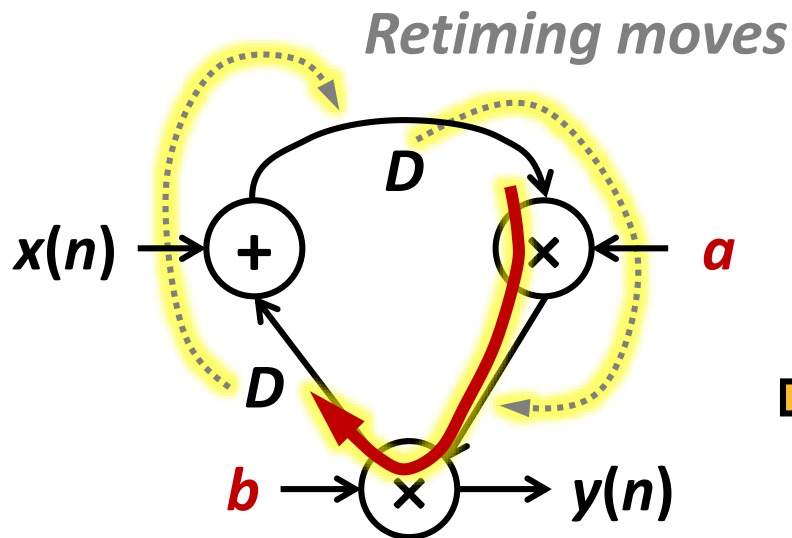
$$w(n) = a \cdot (y_2(n-2) + x(n-1))$$
$$y_2(n) = b \cdot a \cdot y_2(n-2) + b \cdot a \cdot x(n-1)$$

$$y_1(n) \neq y_2(n)$$

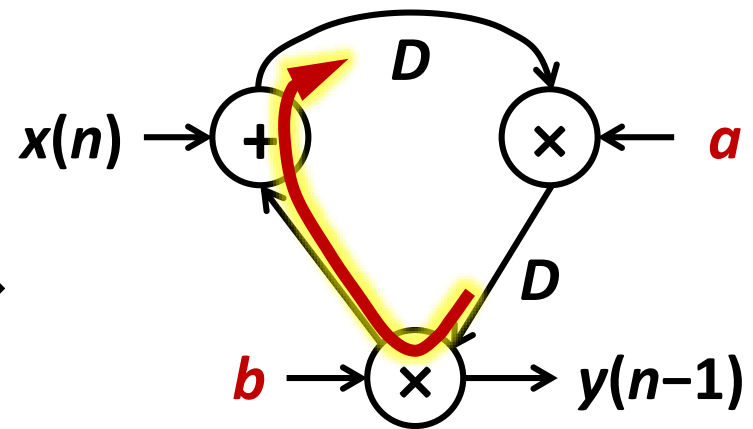
Changing the # delays in a loop alters functionality

High-Level Retiming of an IIR Filter

- IIR throughput is limited by the retiming in the feedback sections
- Optimal placement of registers in the loops leads to max speed



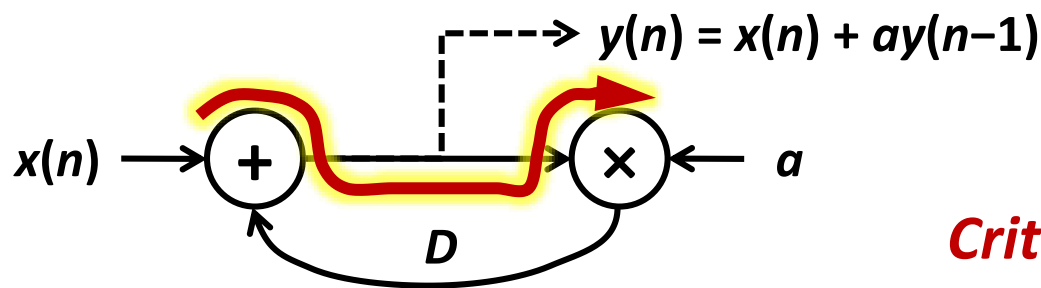
$$t_{critical} = 2t_{mult}$$



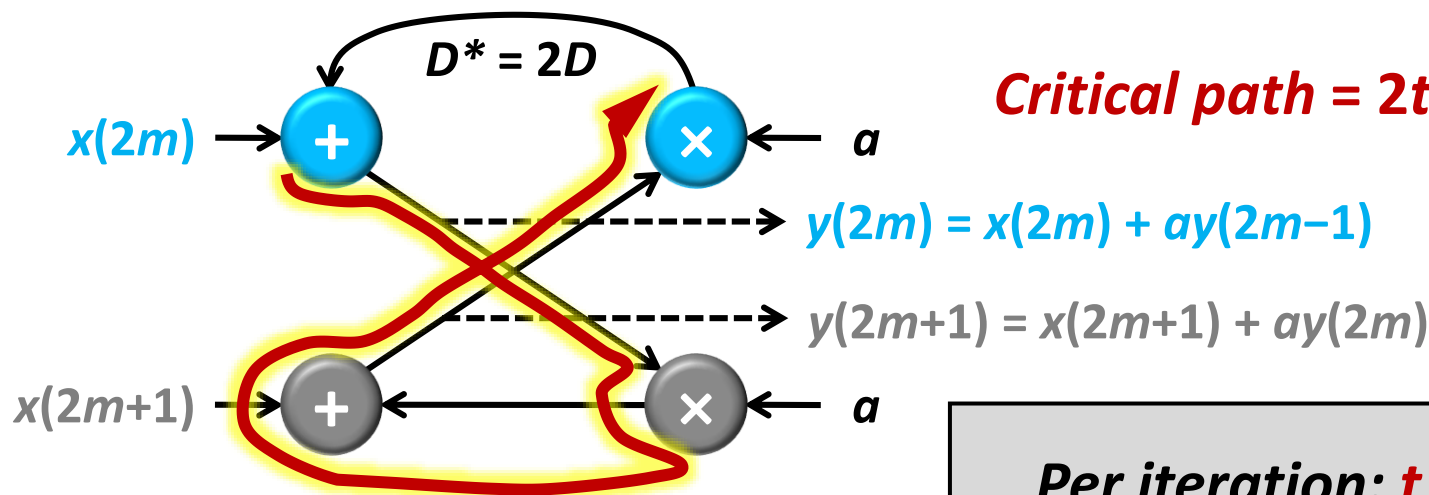
$$t_{critical} = t_{add} + t_{mult}$$

Unfolding: Constant Throughput

- Maximum throughput limited by iteration bound (IB)
- Unfolding does not help if IB is already achieved



Critical path = $t_{add} + t_{mult}$



Critical path = $2t_{add} + 2t_{mult}$

Per iteration: $t_{add} + t_{mult}$

IIR Summary

- **Pros**

- Suitable when filter response has sharp roll-off, has narrow-band, or large attenuation in the stop-band
- More area- and power-efficient compared to FIR

- **Cons**

- Difficult to ensure filter stability
- Sensitive to finite-precision arithmetic effects (limit cycles)
- Does not have linear phase response unlike FIR filters
- All-pass filters required if linear phase response desired
- Difficult to increase throughput
 - Pipelining not possible
 - Retiming and parallelism has limited benefits

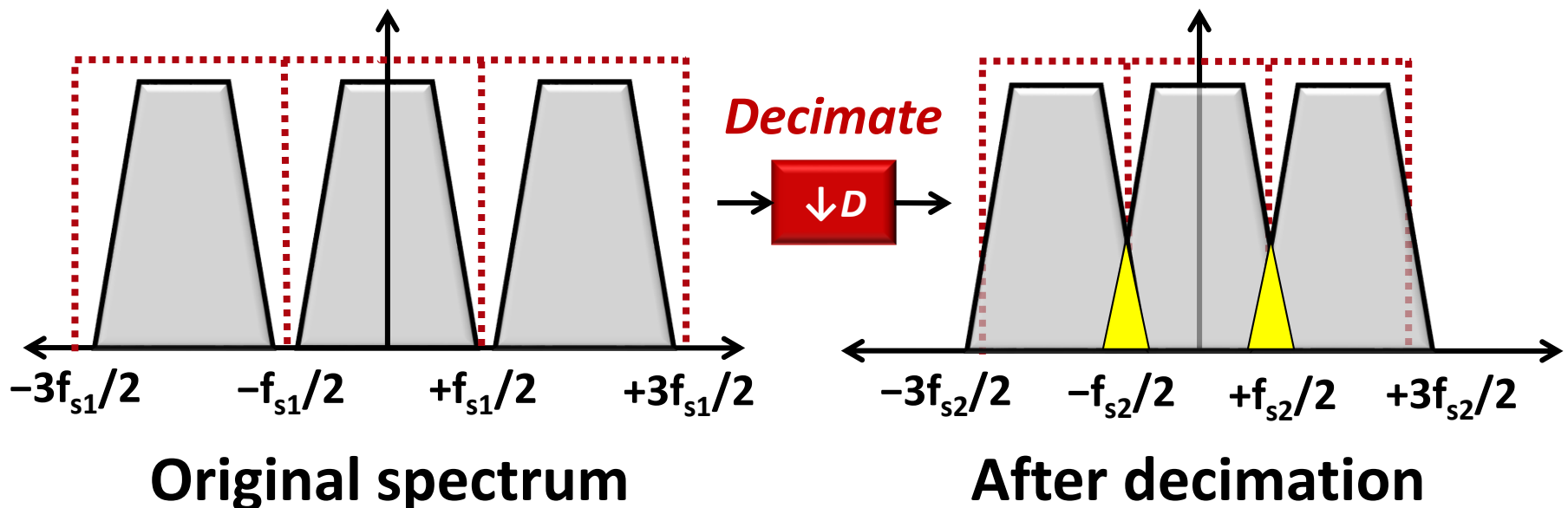
Multi-Rate Filters

Multi-Rate Filters

- Data transfer between blocks at different sample rate
 - **Decimation:** higher rate f_{s1} to lower rate f_{s2}
 - **Interpolation:** lower rate f_{s2} to higher rate f_{s1}
- ⇒ • For integer f_{s1}/f_{s2}
 - **Drop samples** when decimating
 - Leads to **aliasing** in the original spectrum
 - **Stuff zeros** when interpolating
 - Leads to **images** at multiples of f_{s2}

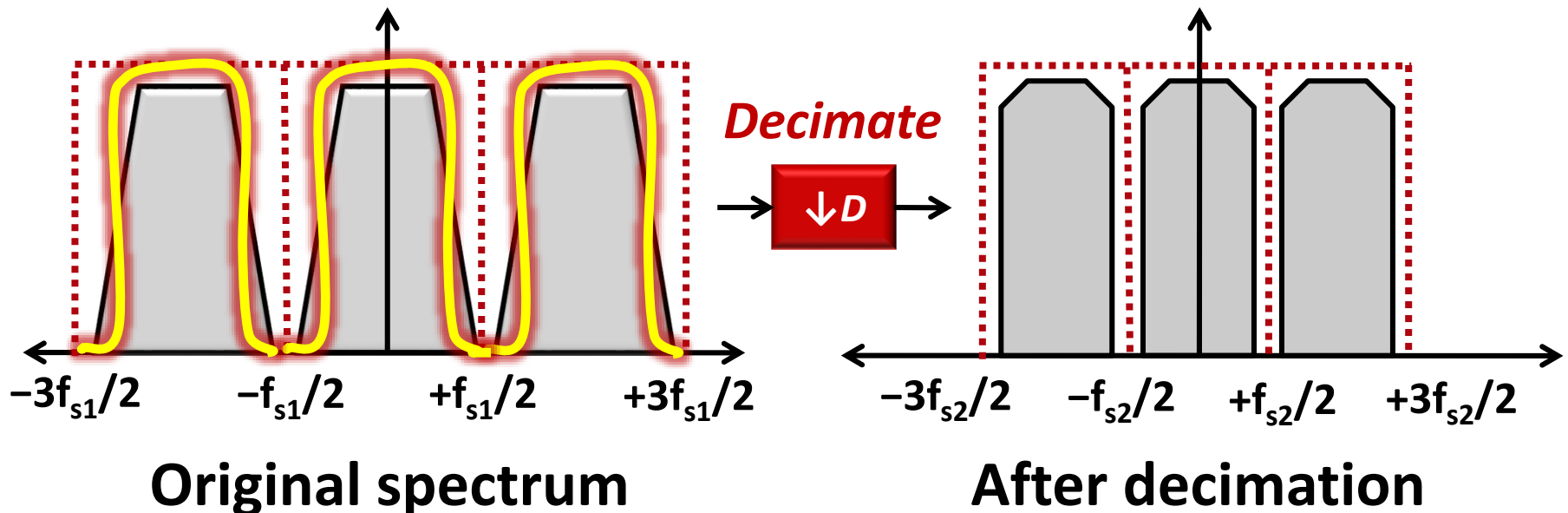
Decimation

- **Frequency-domain representation**
 - Spectrum replicated at intervals of f_{s1} originally
 - Spectrum replicated at intervals of f_{s2} after decimation
 - Aliasing of spectrum lying beyond B/W f_{s2} in original spectrum
- **Decimation filters to remove content beyond f_{s2}**



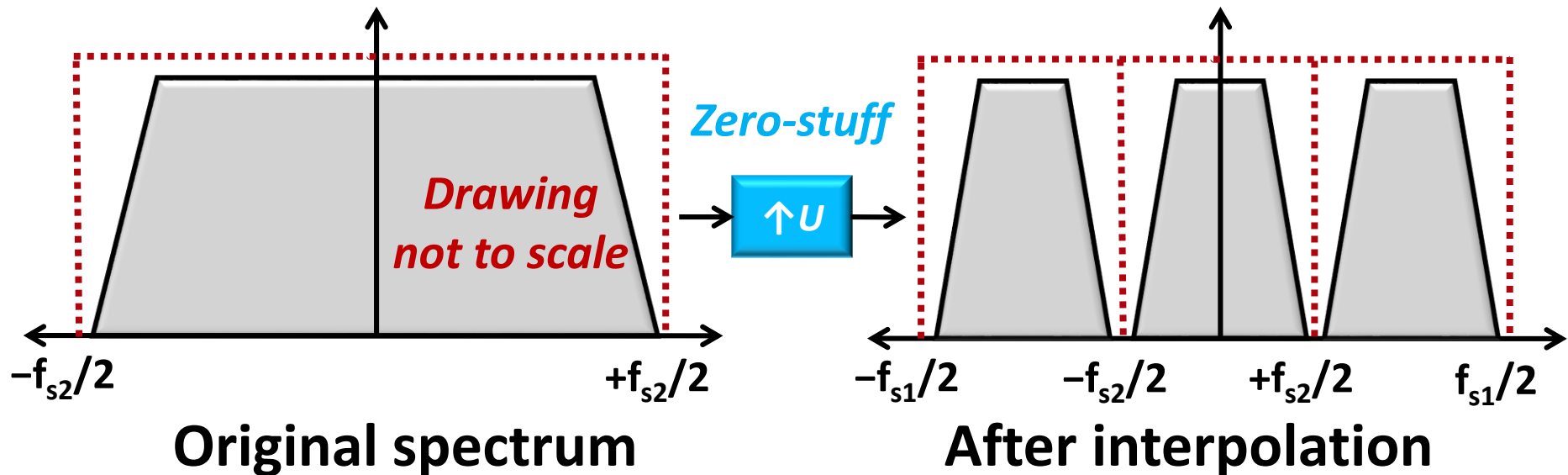
Decimation Filters

- **Low-pass filters** used before decimation
 - Usually FIR realization (IIR if linear phase is not necessary)
 - Cascade integrated comb filter for hardware efficient realization
 - Much cheaper to implement than FIR or IIR realizations
 - Less attenuation, useful in heavily over-sampled systems



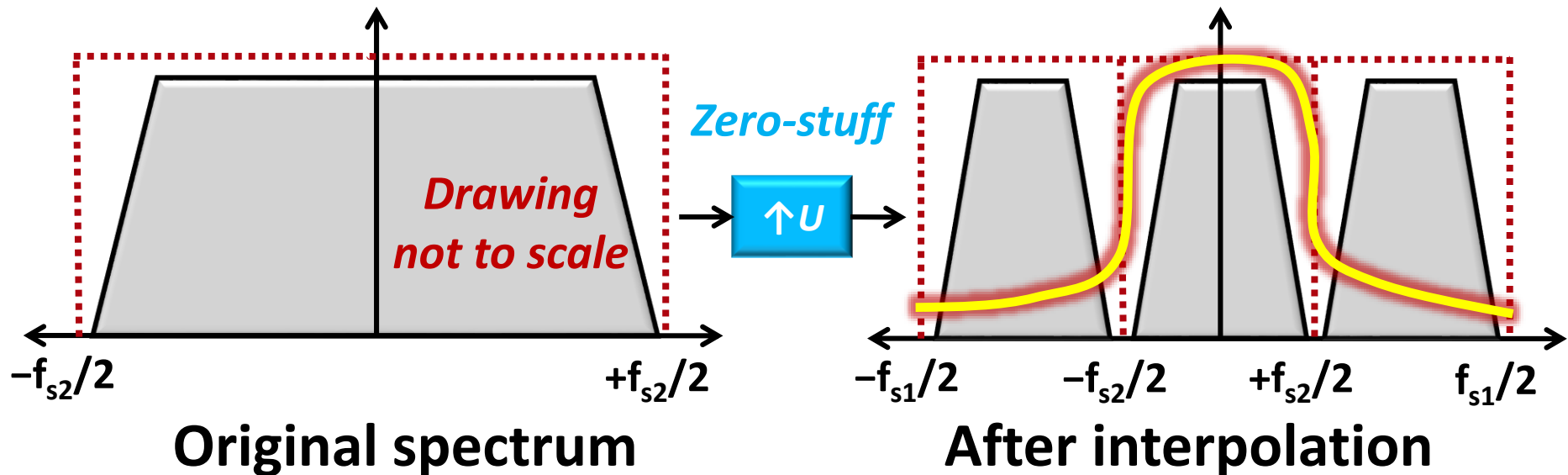
Interpolation

- **Frequency domain representation**
 - Spectrum spans bandwidth of f_{s2} originally
 - Spectrum spans bandwidth of f_{s1} after interpolation
 - Images of spectrum at intervals of f_{s2} after interpolation
- **Interpolation filters to remove images at multiples of f_{s2}**



Interpolation Filters

- **Low-pass filter** used after interpolation
 - Suppress spectrum images at multiples of f_{s2}
 - Usually FIR realizations (IIR if linear phase not necessary)
 - Cascade integrated comb filter for hardware efficient realization
 - Much cheaper to implement than FIR or IIR realizations
 - Less attenuation, useful for large interpolation factors (U)

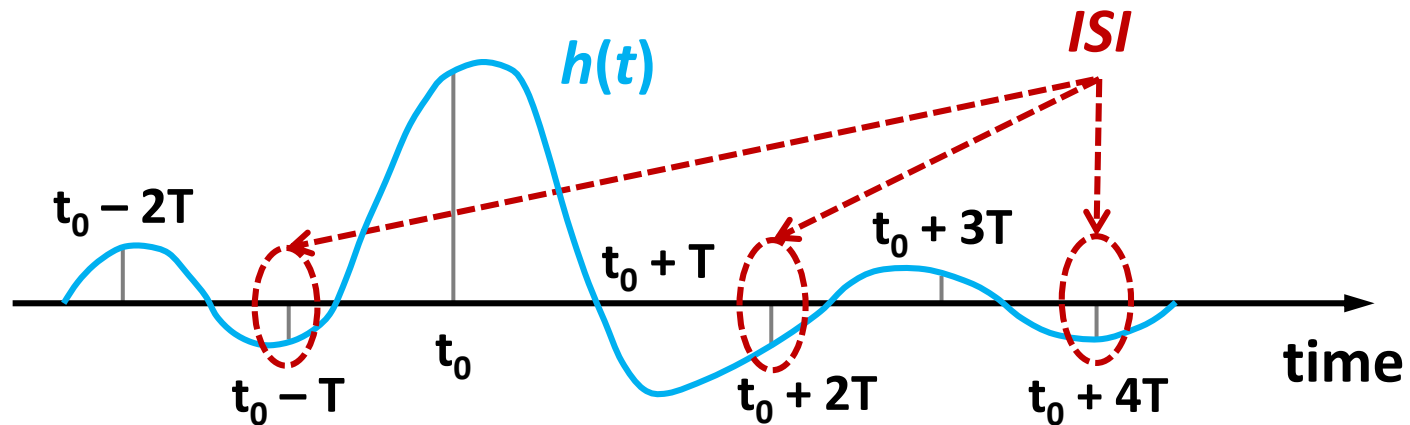


Adaptive Filters:

Equalization

Inter-Symbol Interference (ISI)

- Channel response causes delay spread in Tx symbols
- Adjacent symbols contribute at sampling instants



- ISI and adaptive noise modeling

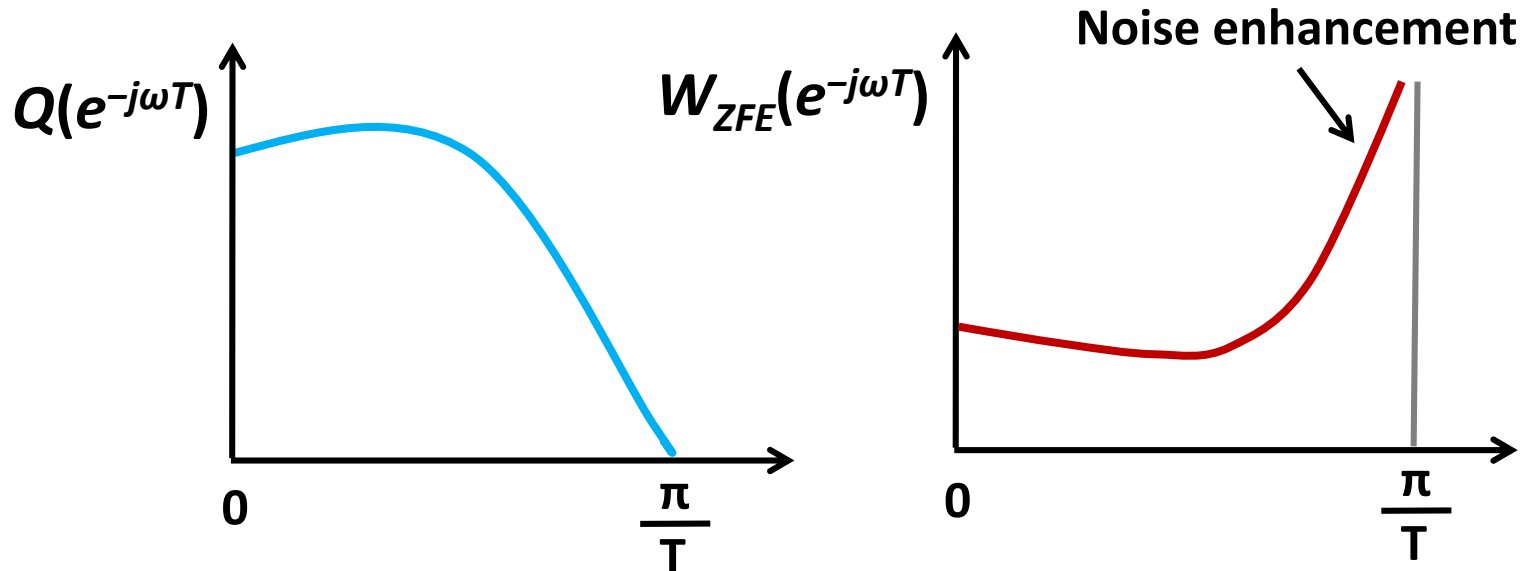
$$r(t_0 + kT) = x_k h(t_0) + \underbrace{\sum_{j \neq k} x_j h(t_0 + kT - jT)}_{\text{ISI}} + \underbrace{n(t_0 + kT)}_{\text{noise}}$$

Zero-Forcing Equalizer

- **Basic equalization techniques**

- Zero-forcing (ZF)

- Causes noise enhancement



- **Adaptive equalization**

- Least-mean squares (LMS) algorithm

- More robust

Adaptive Equalization

- **Achieve minimum mean-square error (MMSE)**
 - Equalized signal: $\mathbf{z}_k$, transmitted signal: $\mathbf{x}_k$
 - Error: $\mathbf{e}_k = \mathbf{z}_k - \mathbf{x}_k$
 - **Objective:** Minimize expected error, $\min E[e_k^2]$
 - Equalizer coefficients at time instant k : $c_n(k)$, $n \in \{0, 1, \dots, N\}$
 - For real optimality: set

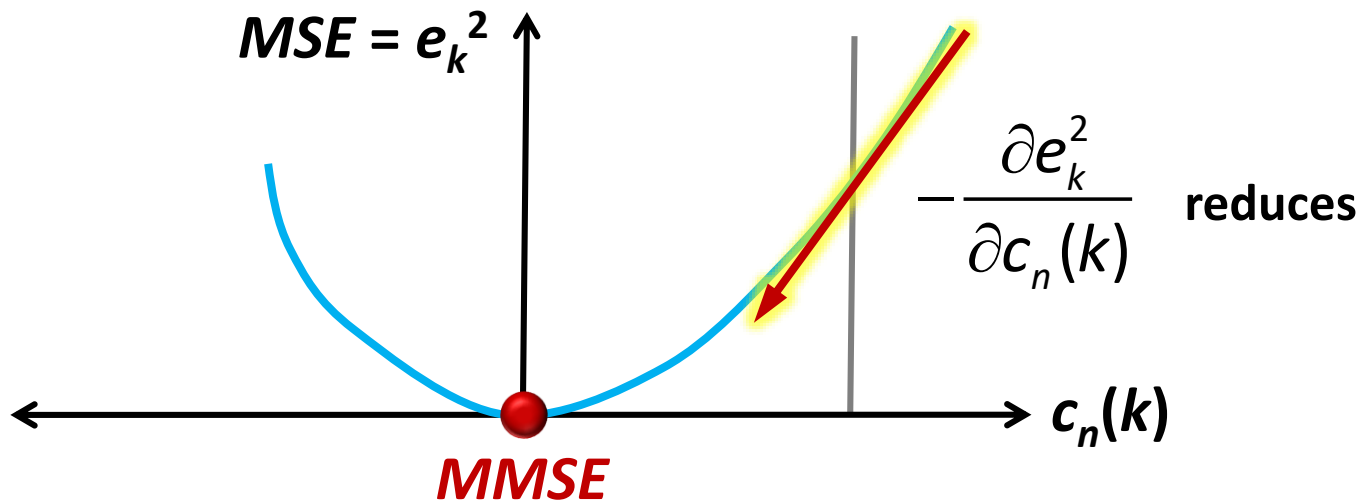
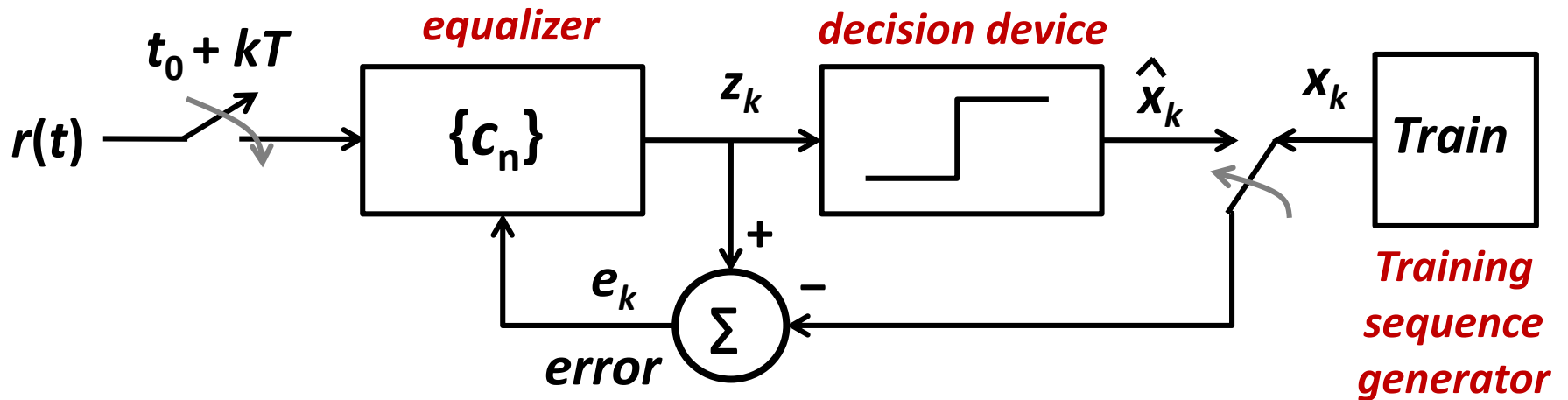
$$\frac{\partial E[e_k^2]}{\partial c_n(k)} = 0$$

- Equalized signal $\mathbf{z}_k$ given by:

$$\mathbf{z}_k = c_0 \mathbf{r}_k + c_1 \mathbf{r}_{k-1} + c_2 \mathbf{r}_{k-2} + \dots + c_{n-1} \mathbf{r}_{k-n+1}$$

- $\mathbf{z}_k$ is convolution of received signal $\mathbf{r}_k$ w/ tap coefficients

LMS Adaptive Equalization

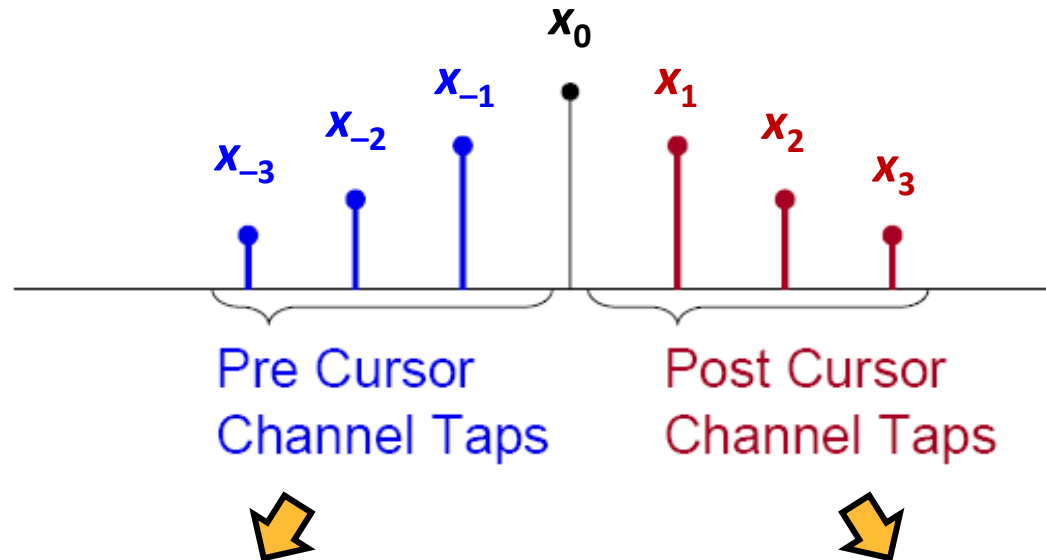


Approximate Calculation of MMSE

- For computational optimality
 - Set: $\partial E[e_k^2]/\partial c_n(k) = 2e_k r_{k-n} = 0$
- Tap update equation: $e_k = z_k - x_k$
- Step size: Δ
- Good approximation if using:
$$c_n(k+1) = c_n(k) - \Delta e_k r(k-n), n = 0, 1, \dots, N-1$$
 - Small step size
 - Large number of iterations
- Error-signal power comparison: $\sigma_{\text{LMS}}^2 \leq \sigma_{\text{ZF}}^2$

Decision-Feedback Equalizers

Cancel the interference from previously detected symbols



Feed-forward equalizer

- Remove the pre-crs ISI
- FIR (linear)

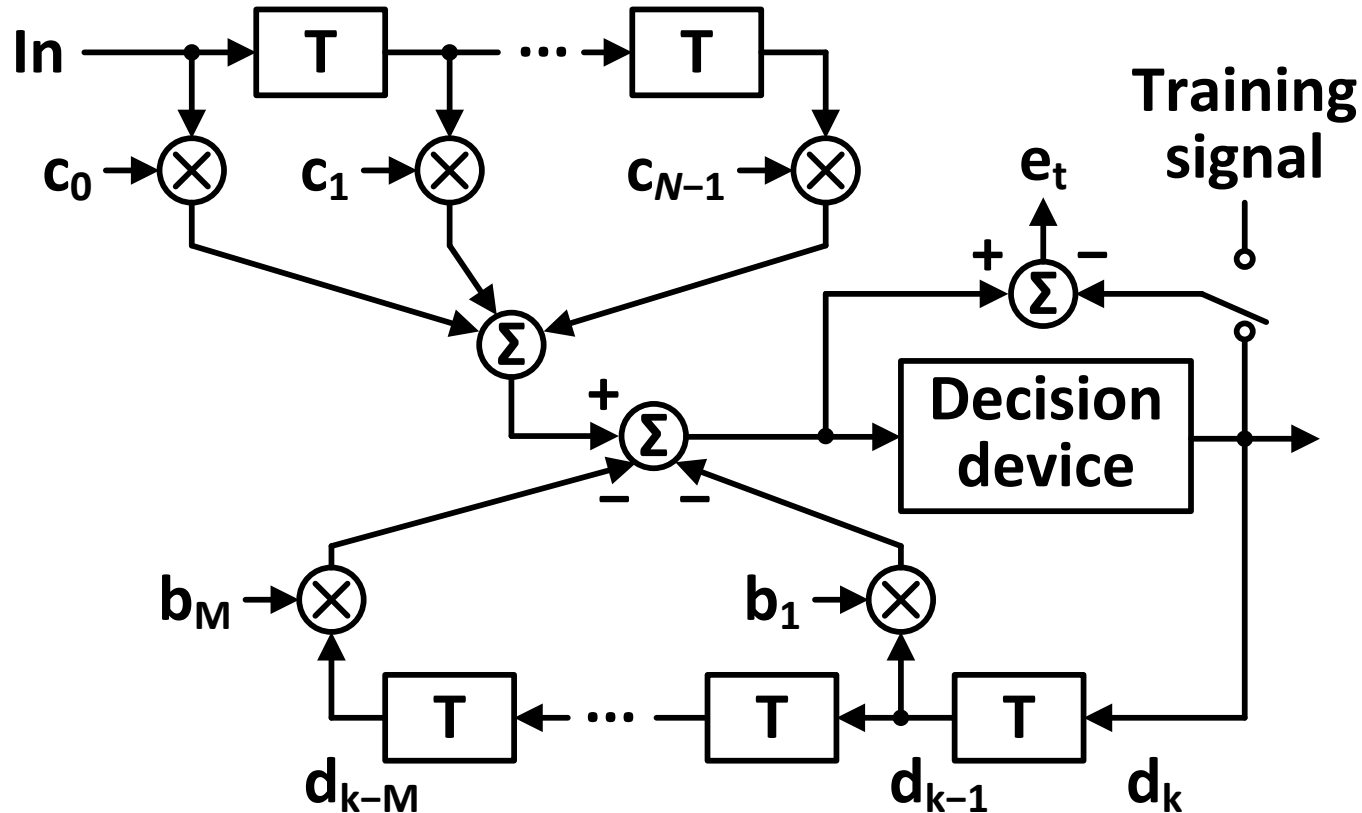
Feedback equalizer

- Remove the post-crs ISI
- Like a ZF equalizer

$$b_{m+1}(k+1) = b_m(k) - \Delta e_k d_{k-m}$$

DFE Architecture

Feed-forward equalizer: $c_0, c_1, \dots, c_{N-1}$



Feedback equalizer: $b_1, b_2, \dots, b_M$

DFE Properties

- **Less noise enhancement compared with ZF or LMS**
- **More freedom in selecting feed-forward coefficients**
 - Feed-forward equalizer need not fully invert channel response
- **Symbol decision may be incorrect**
 - Error propagation (slight)

Fractionally-Spaced Equalizers

- **Sampling at symbol period**
 - Equalizes the aliased response
 - Sensitive to sampling phase
- **Can over-sample (e.g. 2x higher rate)**
 - Avoid spectral aliasing at the equalizer input
 - Sample the Rx input at higher rate (e.g. 2x faster)
 - Produce equalizer output signal at symbol rate
 - Can update coefficients at symbol rate
 - Less sensitive to sampling phase

$$c_n(k+1) = c_n(k) - \Delta e_k \cdot r(t_0 + kT - NT/2)$$

Equalizers Summary

- Use equalizers to reduce ISI and achieve high data rate
- Use adaptive equalizers to track time-varying channel
- LMS-based equalization prevails in MODEM design
- DFE uses previous decisions to estimate current symbol
- Fractionally spaced equalizers resilient to sampling phase variation
- Properly select step size for convergence

S. Qureshi, "Adaptive Equalization," *Proceedings of the IEEE*, vol. 73, no. 9, pp. 1349-1387, Sep. 1985.

Digital Filters Summary

- **Digital filters are key building elements in DSP systems**
- **FIR filters can be realized in direct or transposed form**
 - Direct form has long critical-path delay
 - Transposed form has large input loading
 - Multiplications can be simplified by using coefficients that can be derived as sum of power-of-two numbers
- **Performance of IIR filters is limited by the longest loop delay (iteration bound)**
 - IIR filters are suitable for sharp roll-off characteristics
 - More power and area efficient than FIR
- **Multi-rate filters for decimation and interpolation**