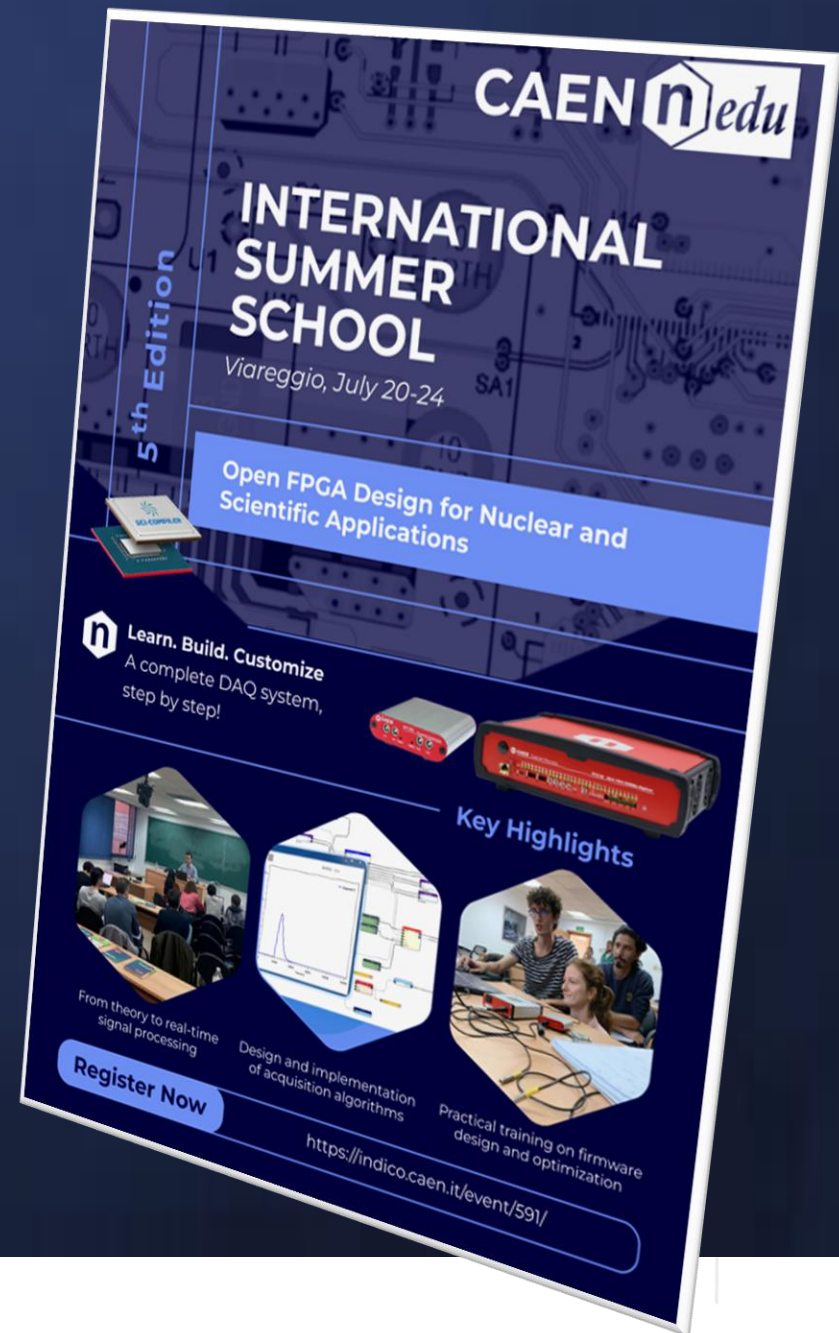


Digital Pulse Processing Algorithm

CAEN Summer School 2026

Viareggio, Italy, July 20th – 24th, 2026

Alberto Potenza



01 From Particle to Digital Domain

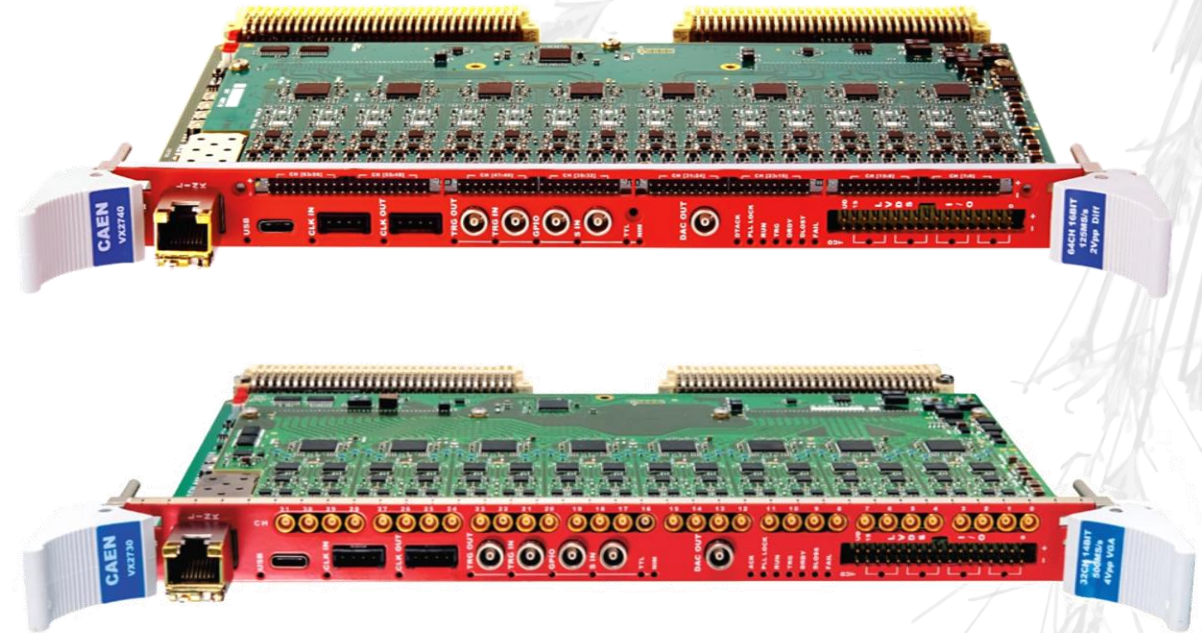
- › Physics of Nuclear Pulse
- › Analog Front-End
- › ADC Parameters

02 Real-Time Event Triggering

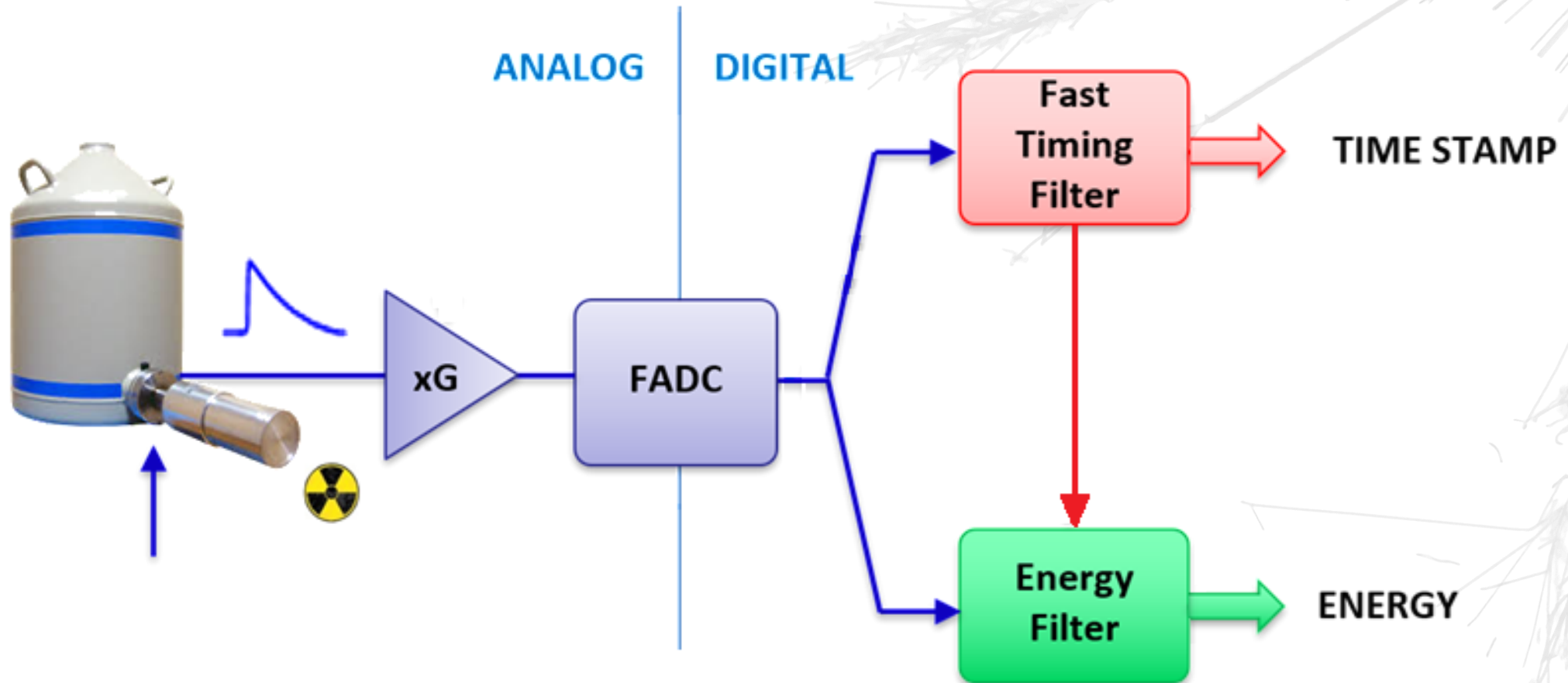
- › Leading-Edge
- › CFD
- › Sub-clock Interpolation

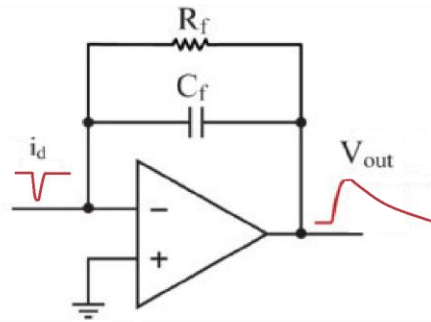
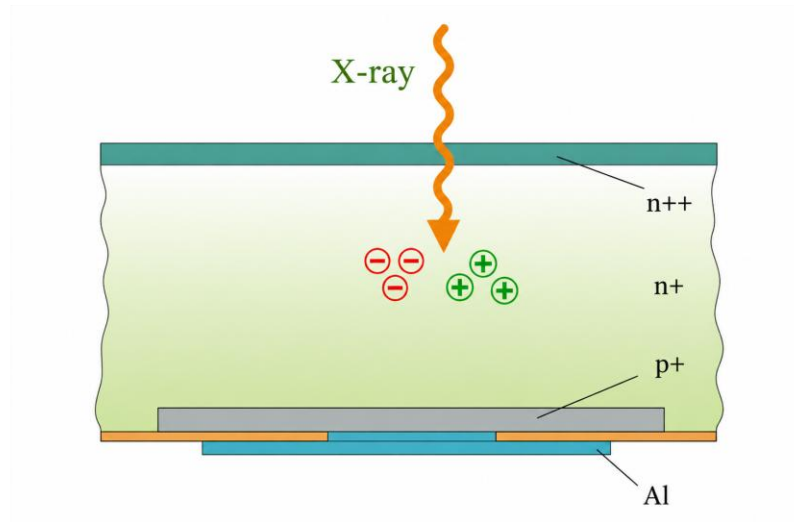
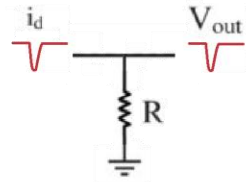
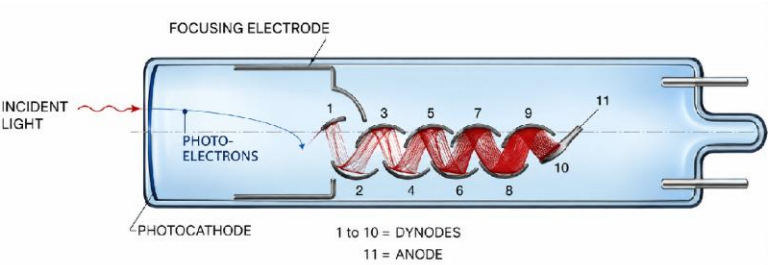
03 Digital Energy Spectroscopy

- › Charge Integration
- › Trapezoidal Filter
- › Pulse Shape Discrimination



From Particles to Digital Domain





Moving Charges

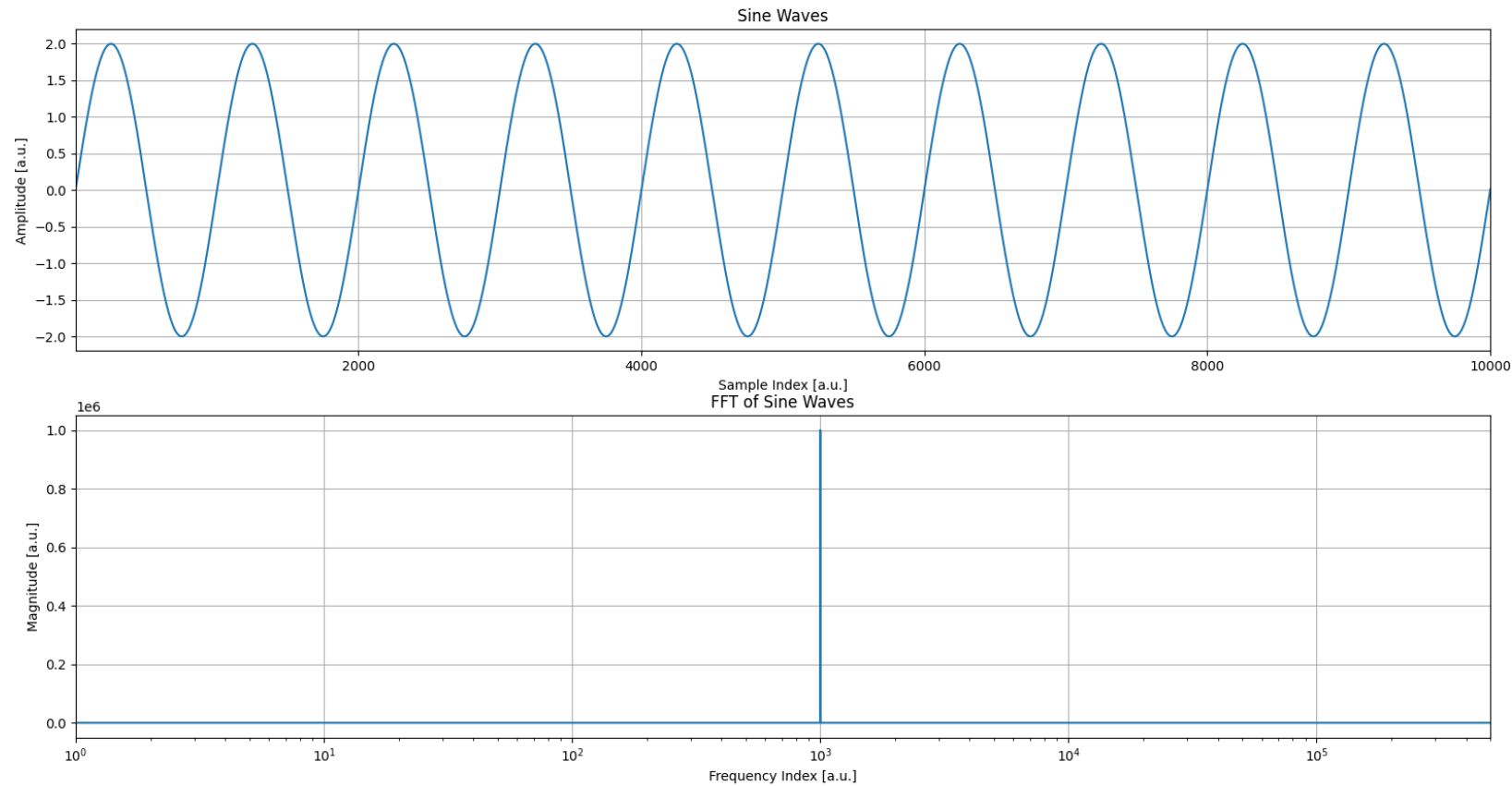
Current Signal

Ground Resistor

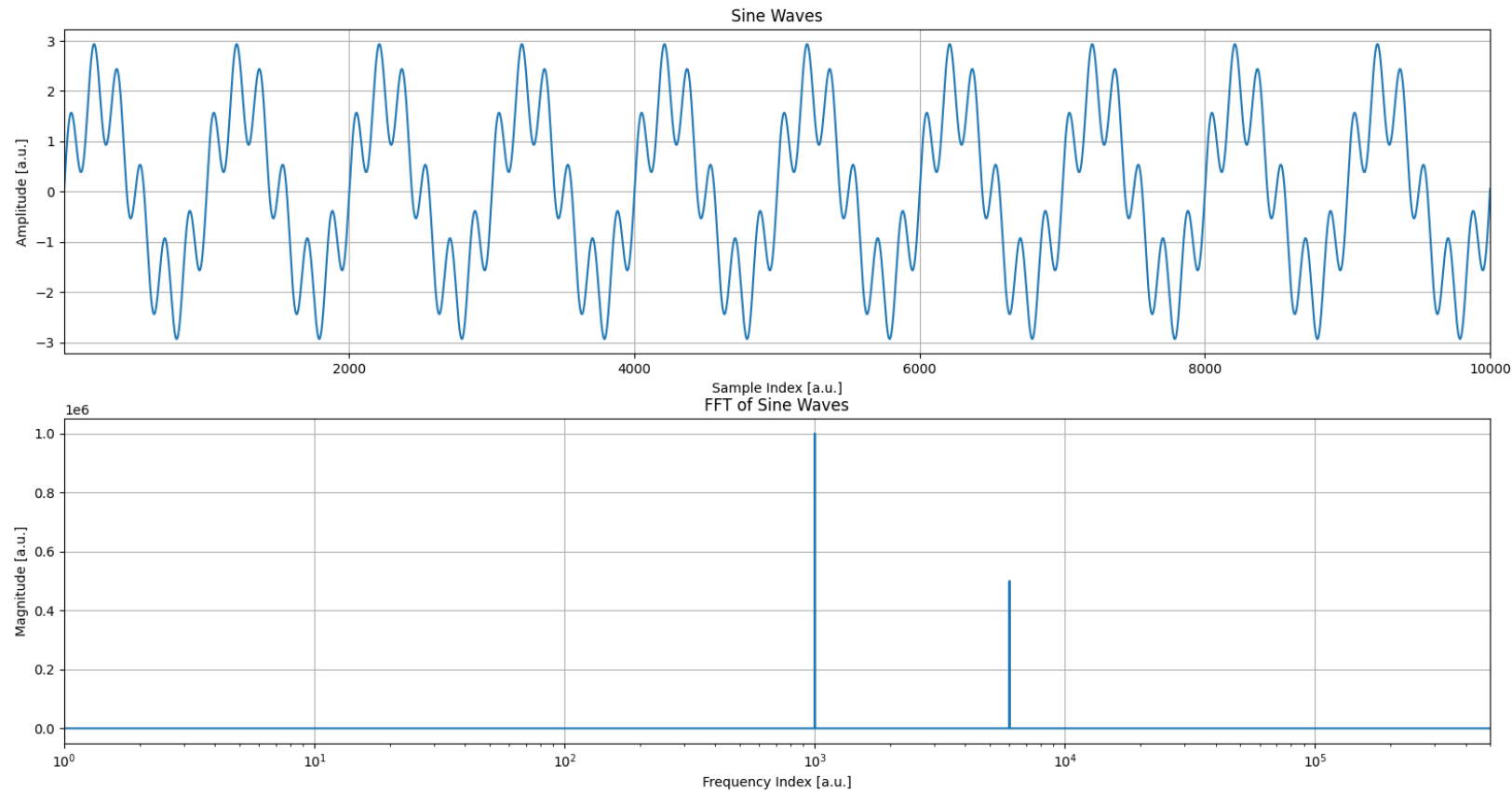
Charge Amplifier

$$V(t) = V_0 (e^{-t/\tau_f} - e^{-t/\tau_r})$$

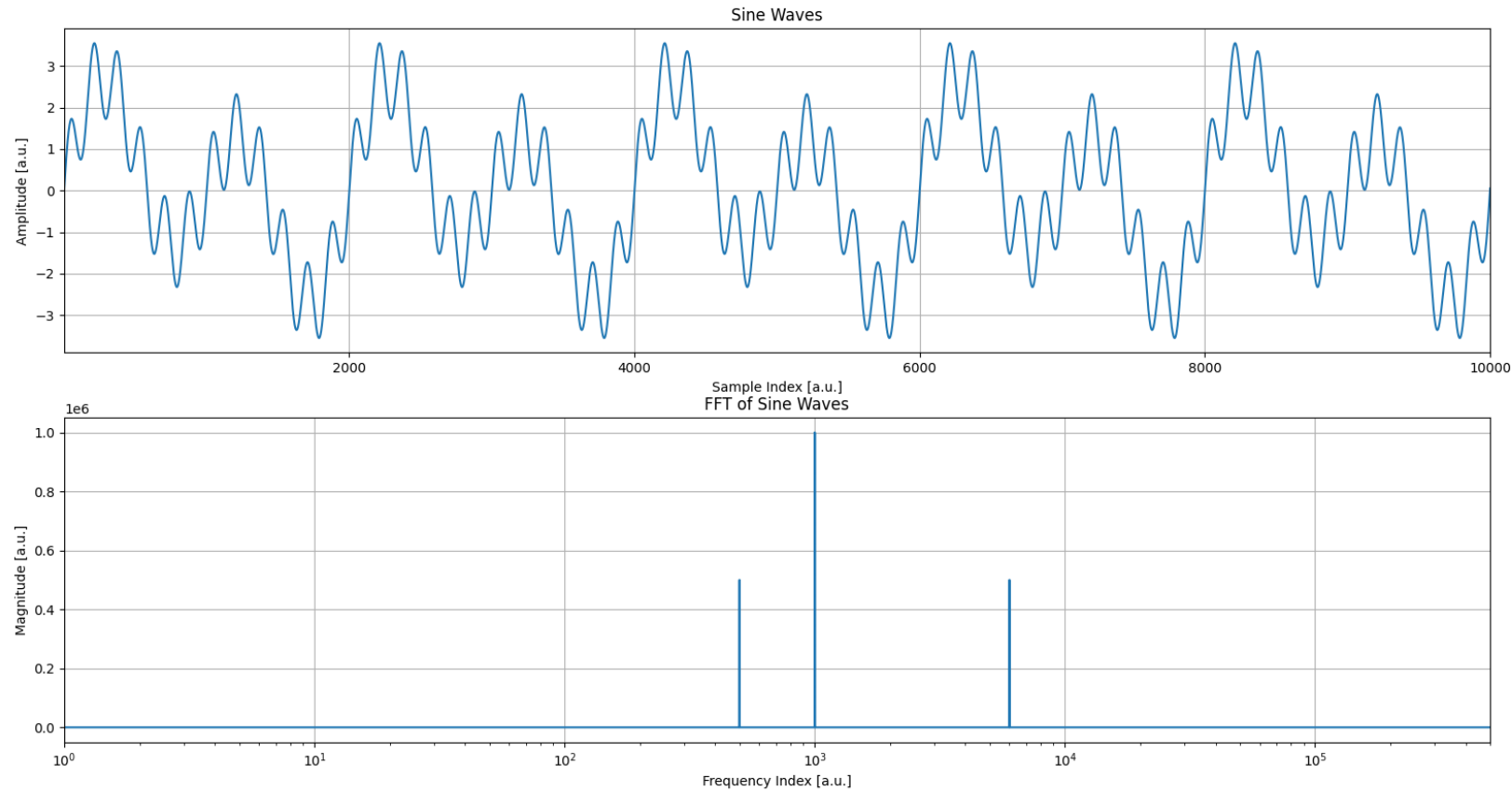
$$V(t) = V_0 \sin(\omega_0 t)$$



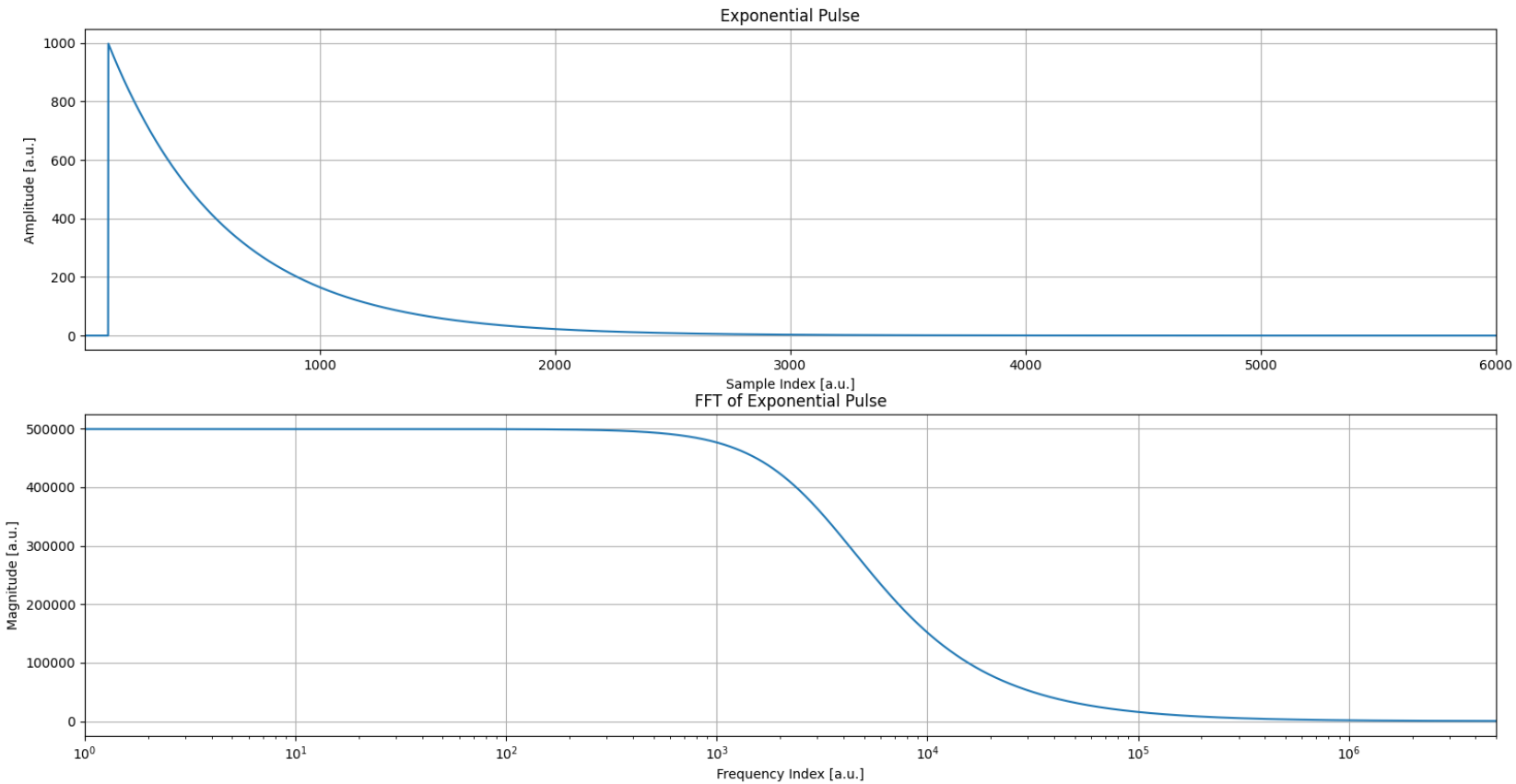
$$V(t) = V_0 \sin(\omega_0 t) + V_1 \sin(\omega_1 t)$$



$$V(t) = V_0 \sin(\omega_0 t) + V_1 \sin(\omega_1 t) + V_2 \sin(\omega_2 t)$$



$$V(t) = V_0 (e^{-t/\tau_f} - e^{-t/\tau_r})$$



$$V(t) = V_0 (e^{-t/\tau_f} - e^{-t/\tau_r})$$

$$\begin{array}{l} t \ll \tau_f \\ \tau_r \rightarrow RC \end{array}$$

$$V(t) = V_0 (1 - e^{-t/RC})$$

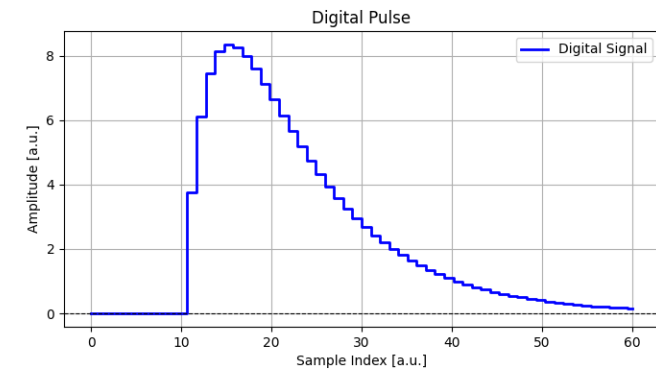
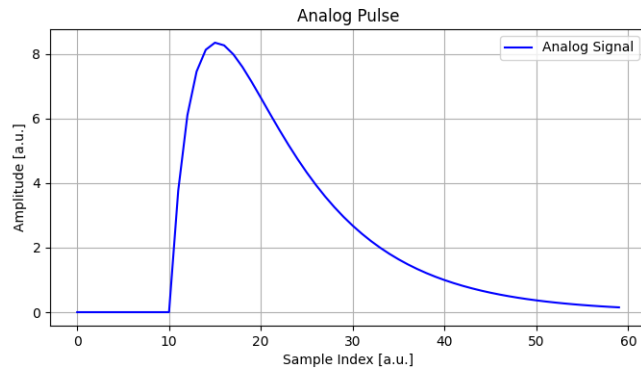
For a simple one-stage low-pass RC network, the 10% to 90% rise time is proportional to the time constant $\tau = RC$

$$\begin{array}{l} V(t_1) = 0.1 V_0 \\ V(t_2) = 0.9 V_0 \end{array}$$

$$t_{rise} = RC \ln(9)$$

$$RC = \frac{1}{2\pi BW}$$

$$BW \approx \frac{0.35}{t_{rise}}$$



$$t_{\text{rise}} \approx 100 \text{ ns}$$

$$BW \approx 3.5 \text{ MHz}$$

$$f_{\text{sampleNyq}} \approx 7 \text{ MHz}$$

Why $f_{\text{sampling}} \gg f_{\text{sampleNyq}}$?

Beyond Nyquist Limits

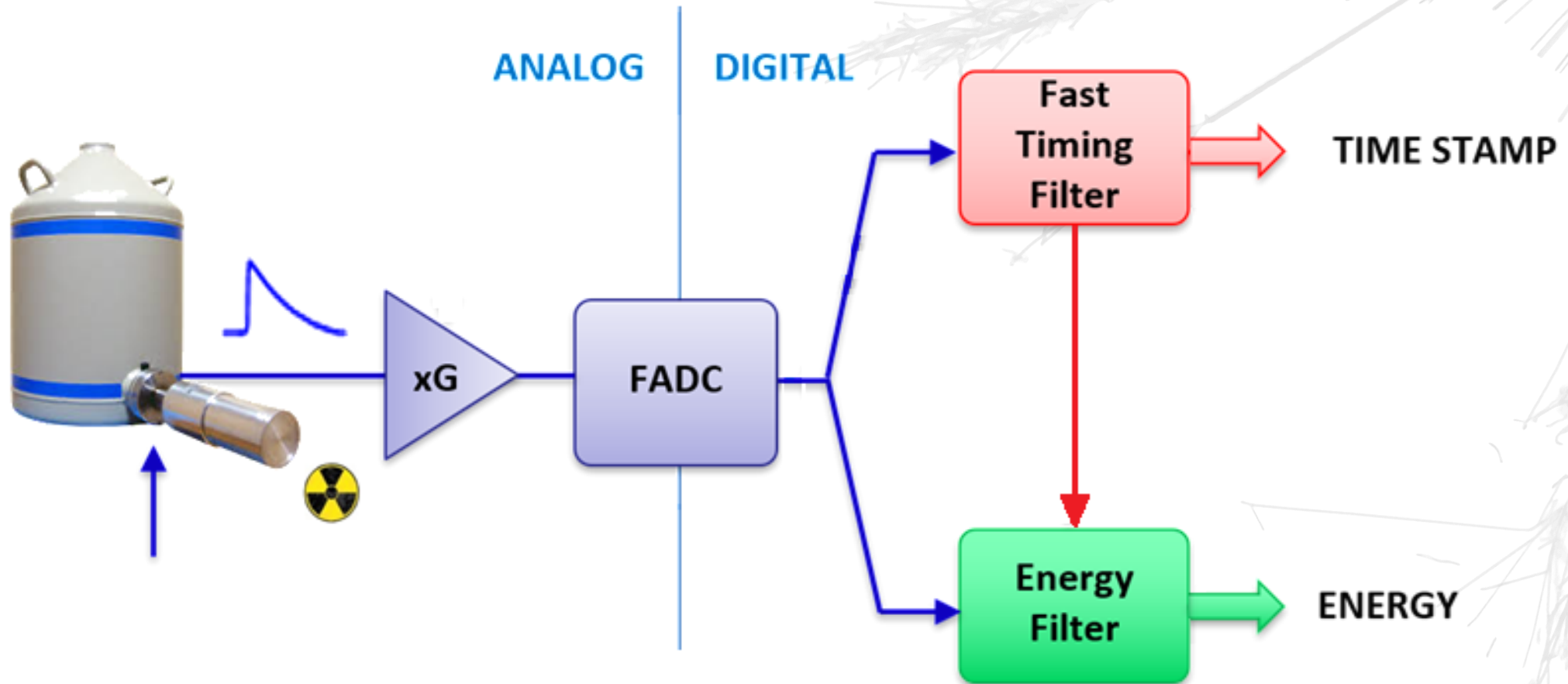
- Nyquist guarantees information preservation, but exact reconstruction requires ideal sinc interpolation.
- Ideal sinc interpolation is computationally expensive and never performed in real-time.

Real ADC Performance

- Resolution is degraded by ENOB, clock noise, INL/DNL non-linearity.
- High-rate oversampling directly helps to average out and mitigate these effects.

Filter Performance Margin

- High sample density reduces interpolation error at the timing threshold and pulse peaks.
- Digital filters perform better with more samples.



Trigger Filters

Primary Objective: Precise Event Timestamping

Target: Determine the exact arrival time with maximum temporal precision.

Application: Time-of-Flight (ToF), coincidence and event timestamping.

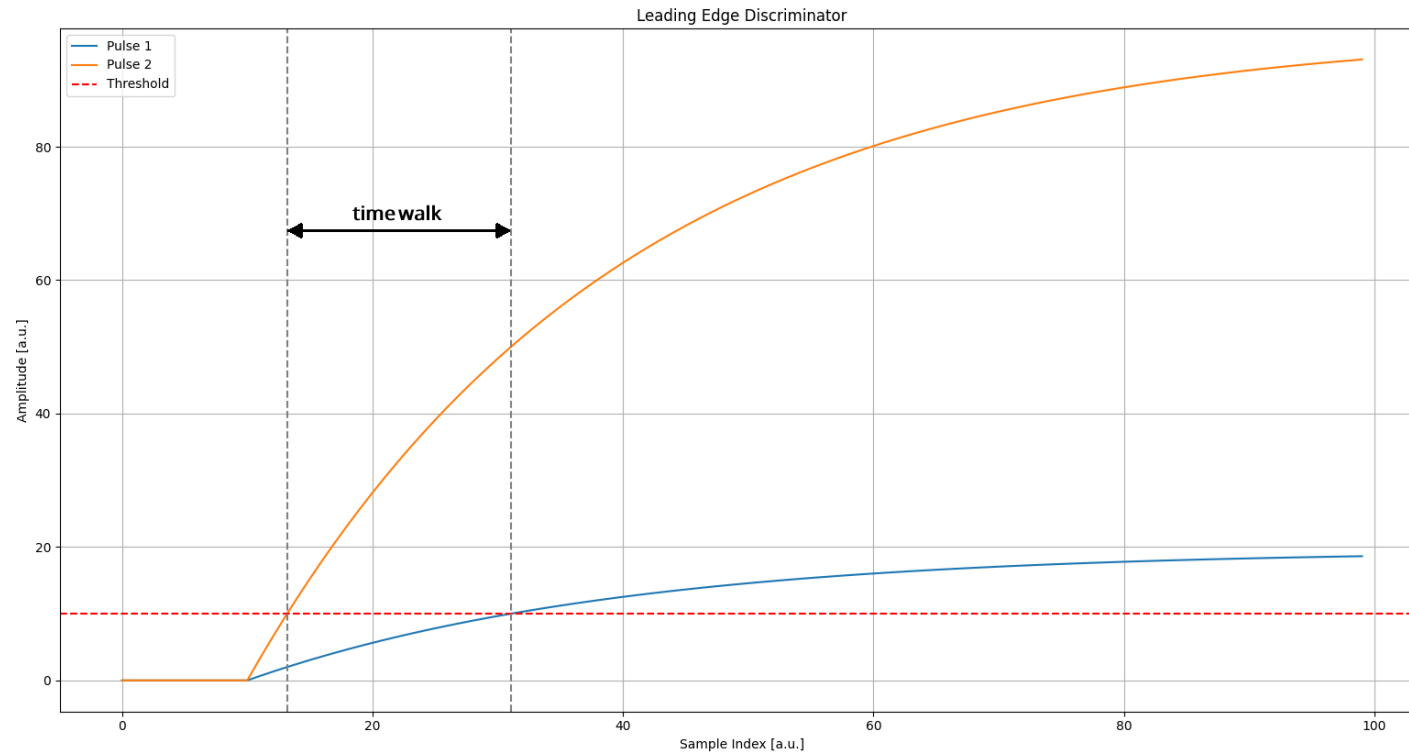
Filter Optimization: Minimizing Temporal Jitter

- Jitter Equation: Temporal jitter depends strictly on noise and the signal slope at the threshold crossing point:

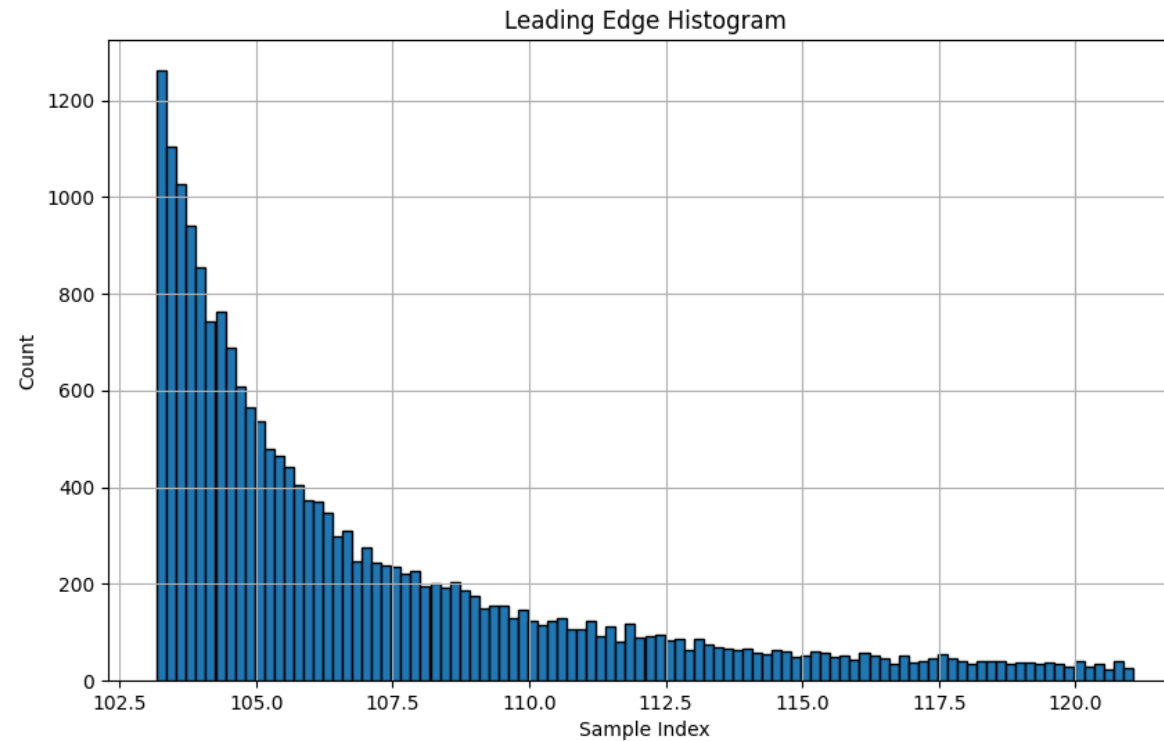
$$\sigma_t = \frac{\sigma_V}{\frac{dV}{dt}}$$

- Bandwidth Requirement: Must preserve high frequencies to maintain a steep rise time (dV/dt)

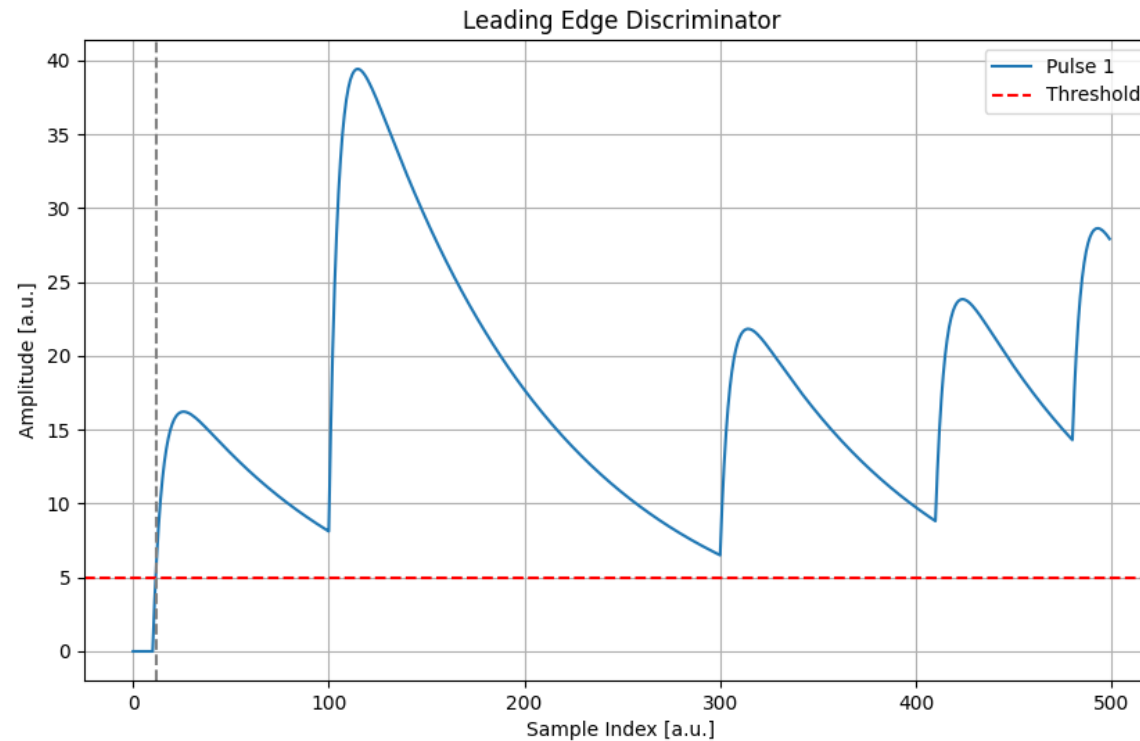
$$t_{trig} = \min\{t : V(t) \geq V_{th}\}$$



Time Walk Problem

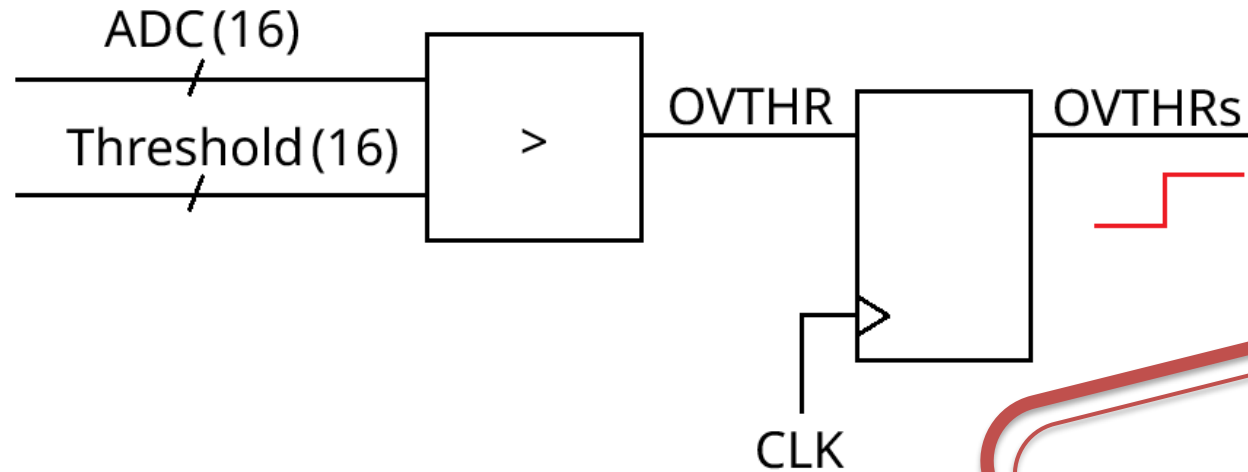


Pile-up Problem



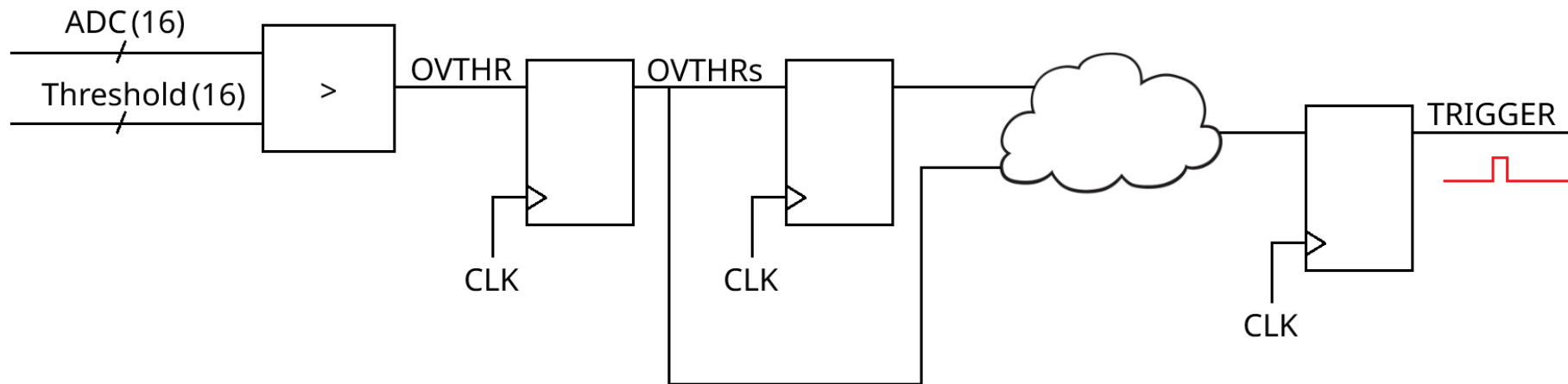
Why a Leading Edge Discriminator ?

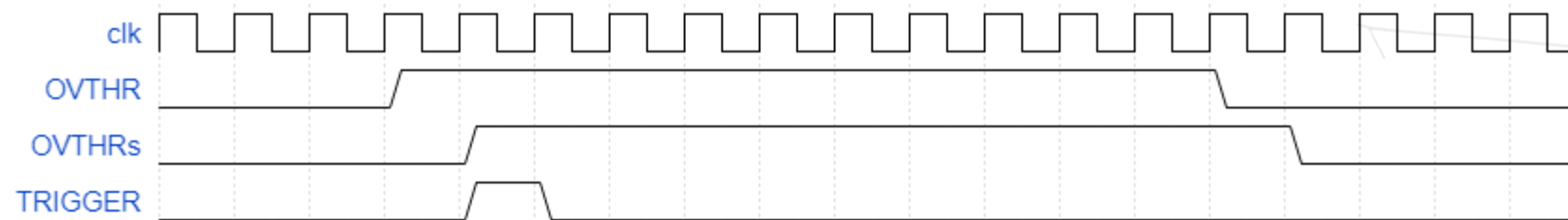
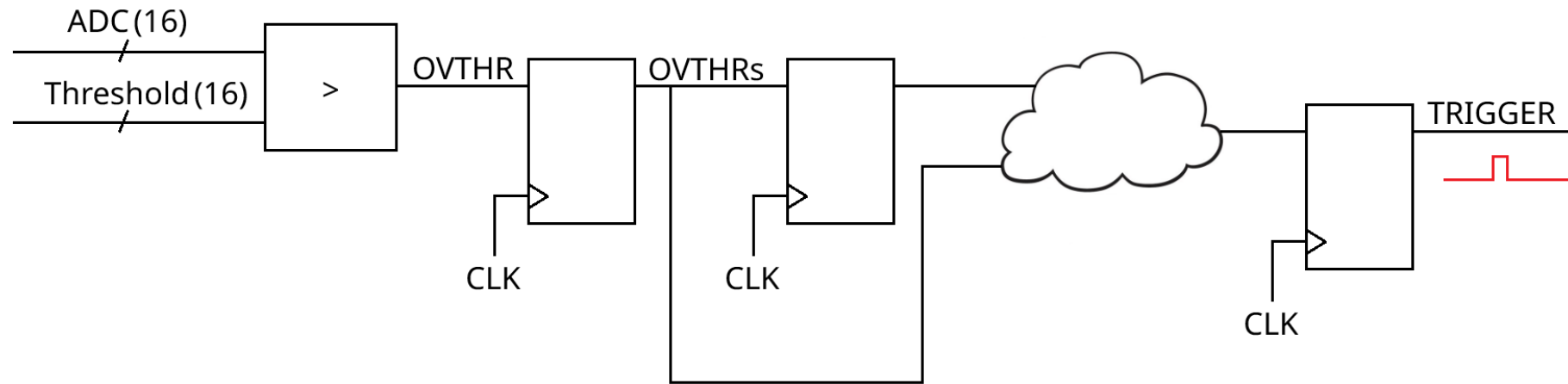
Why a Leading Edge Discriminator ?



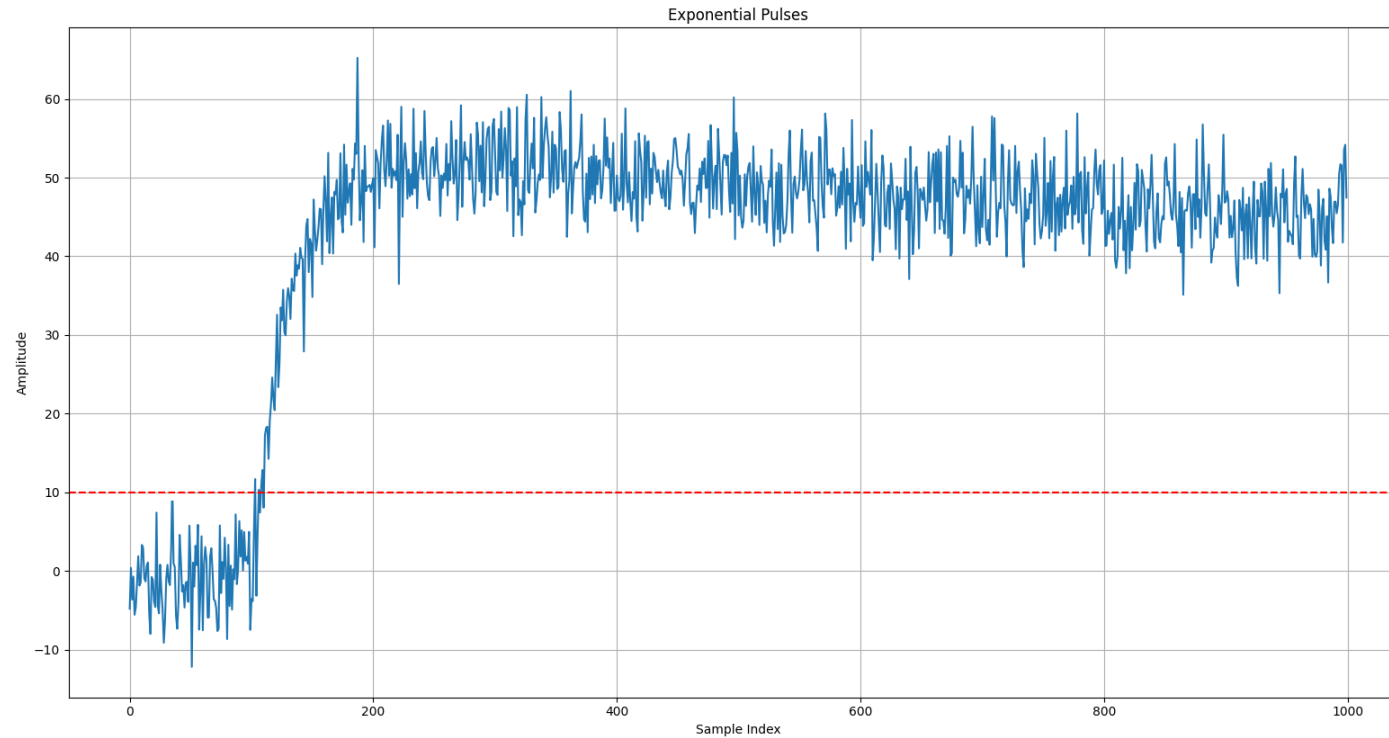
SIMPLE !!!

Why a Leading Edge Discriminator ?

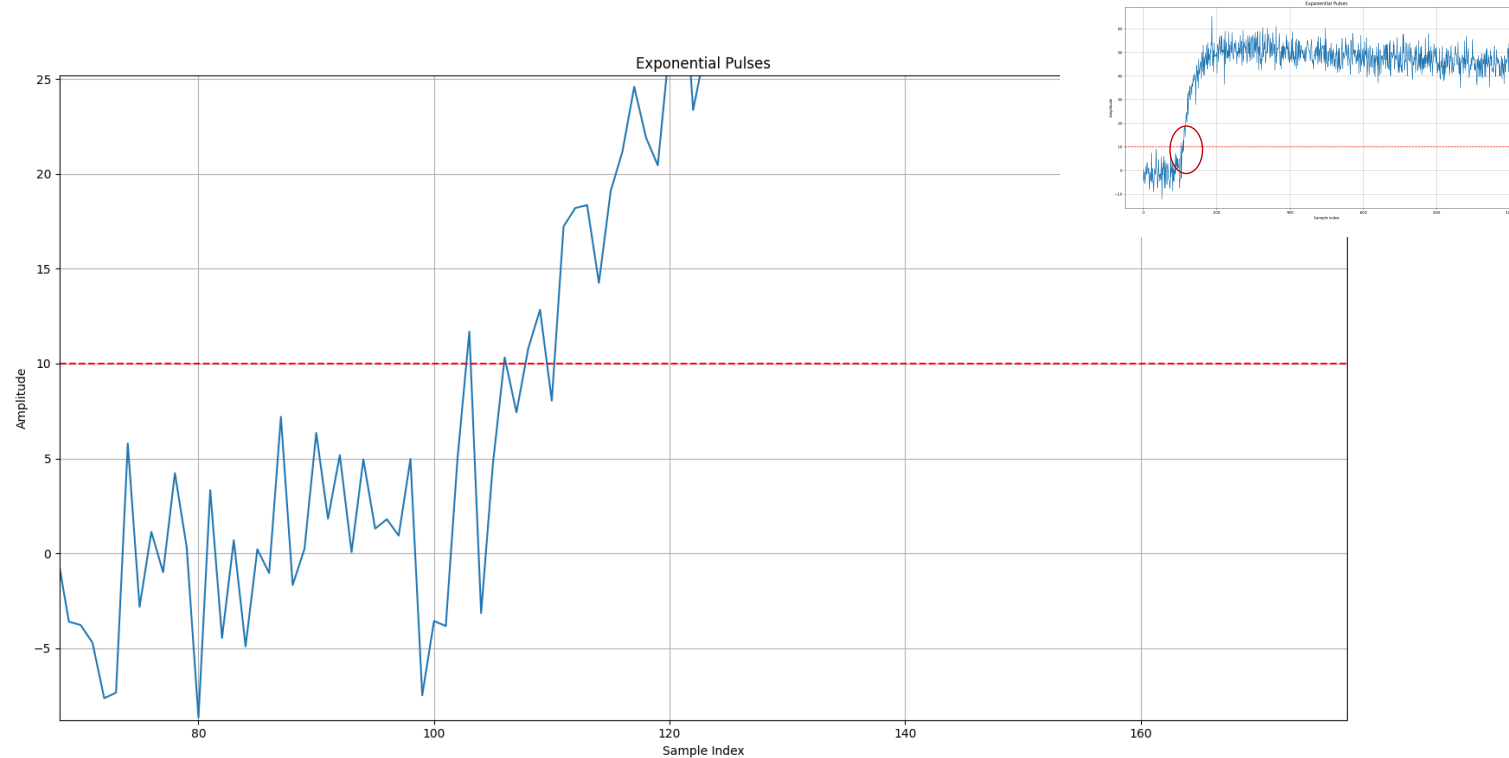


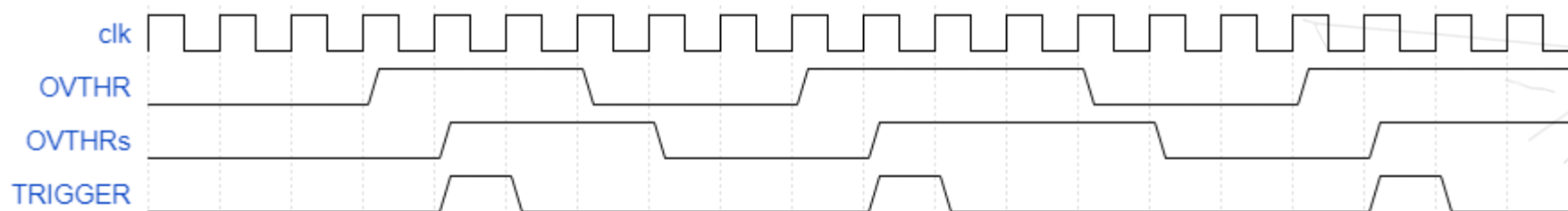
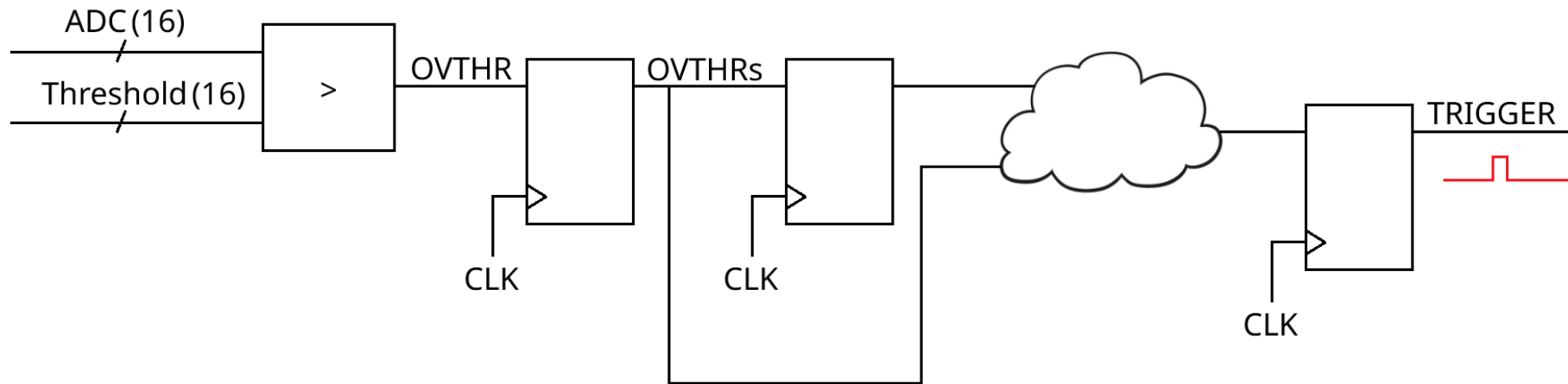


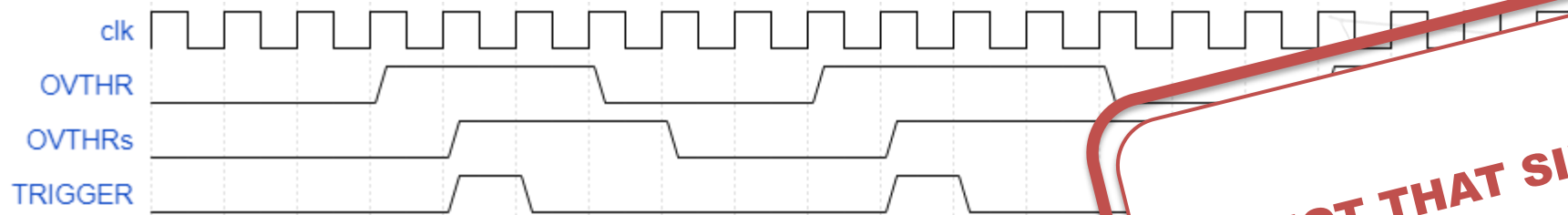
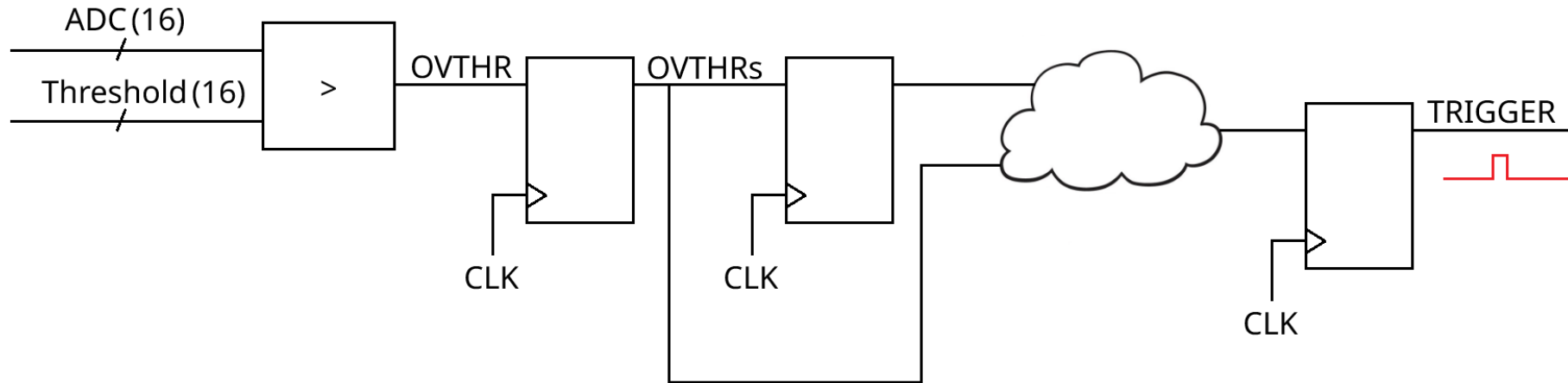
Noise Effects



Noise Effects

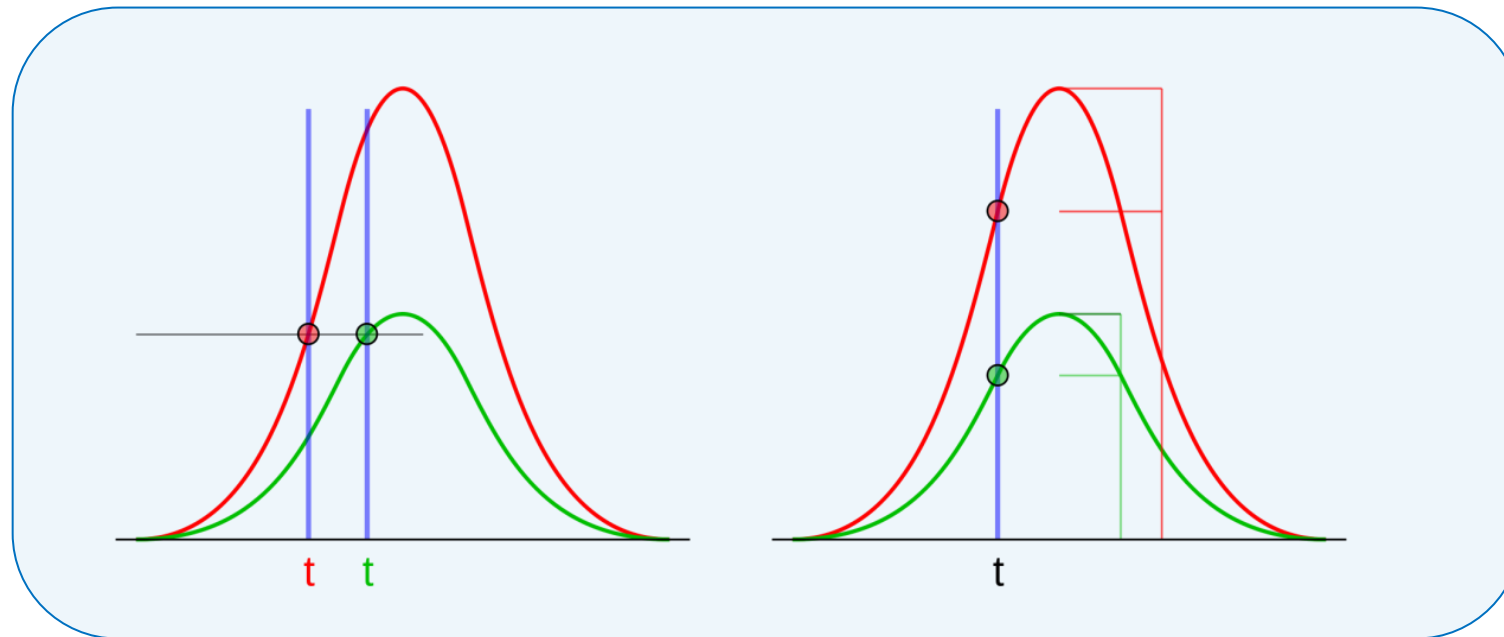






NOT THAT SIMPLE !!!

*Shifting the paradigm:
from an absolute voltage threshold to an amplitude-independent fraction of the pulse.*



$$V_{cfd}(t) = kV_{in}(t) - V_{in}(t - t_d)$$

$$0 = kV_{in}(t) - V_{in}(t - t_d)$$



$$kV_0 \left(e^{-t/\tau_f} - e^{-t/\tau_r} \right) = V_0 \left(e^{-(t-t_d)/\tau_f} - e^{-(t-t_d)/\tau_r} \right)$$

$$V_{cfd}(t) = kV_{in}(t) - V_{in}(t - t_d)$$

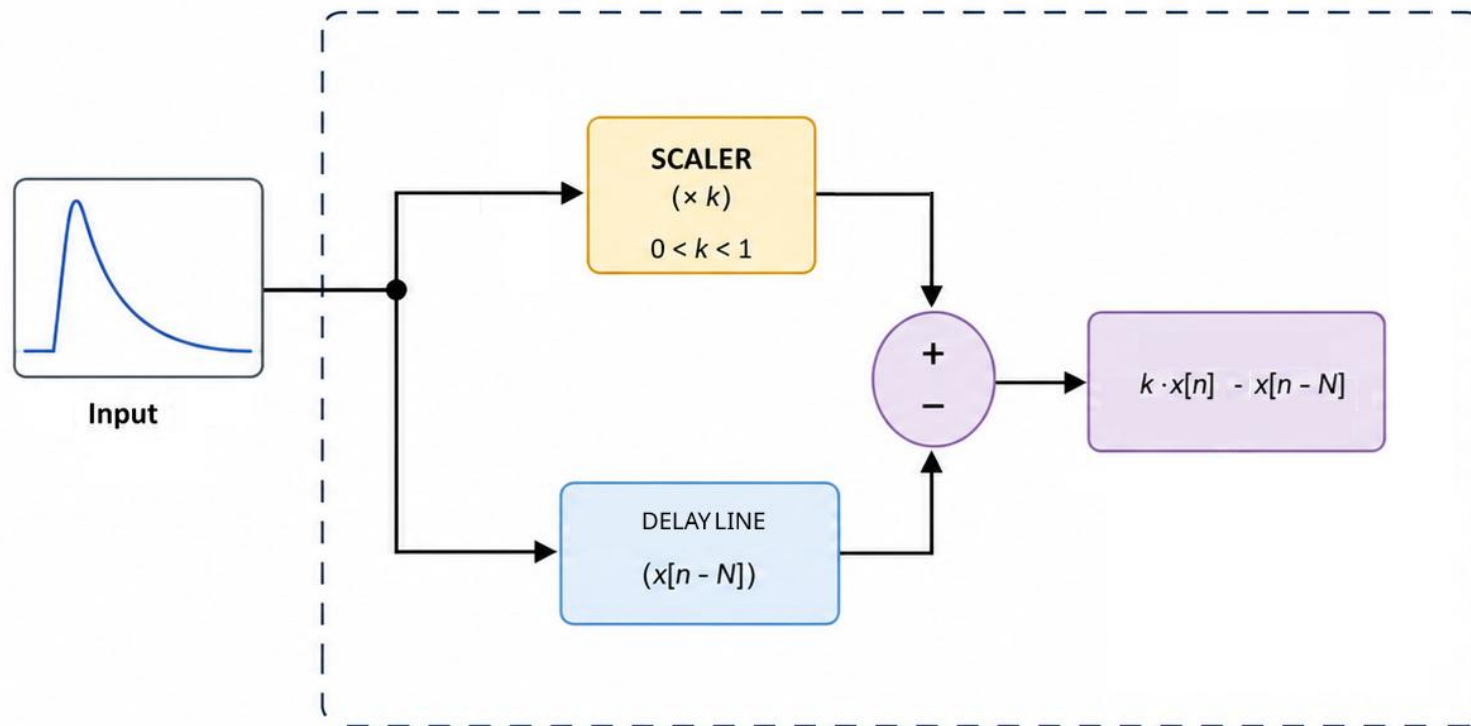
$$0 = kV_{in}(t) - V_{in}(t - t_d)$$



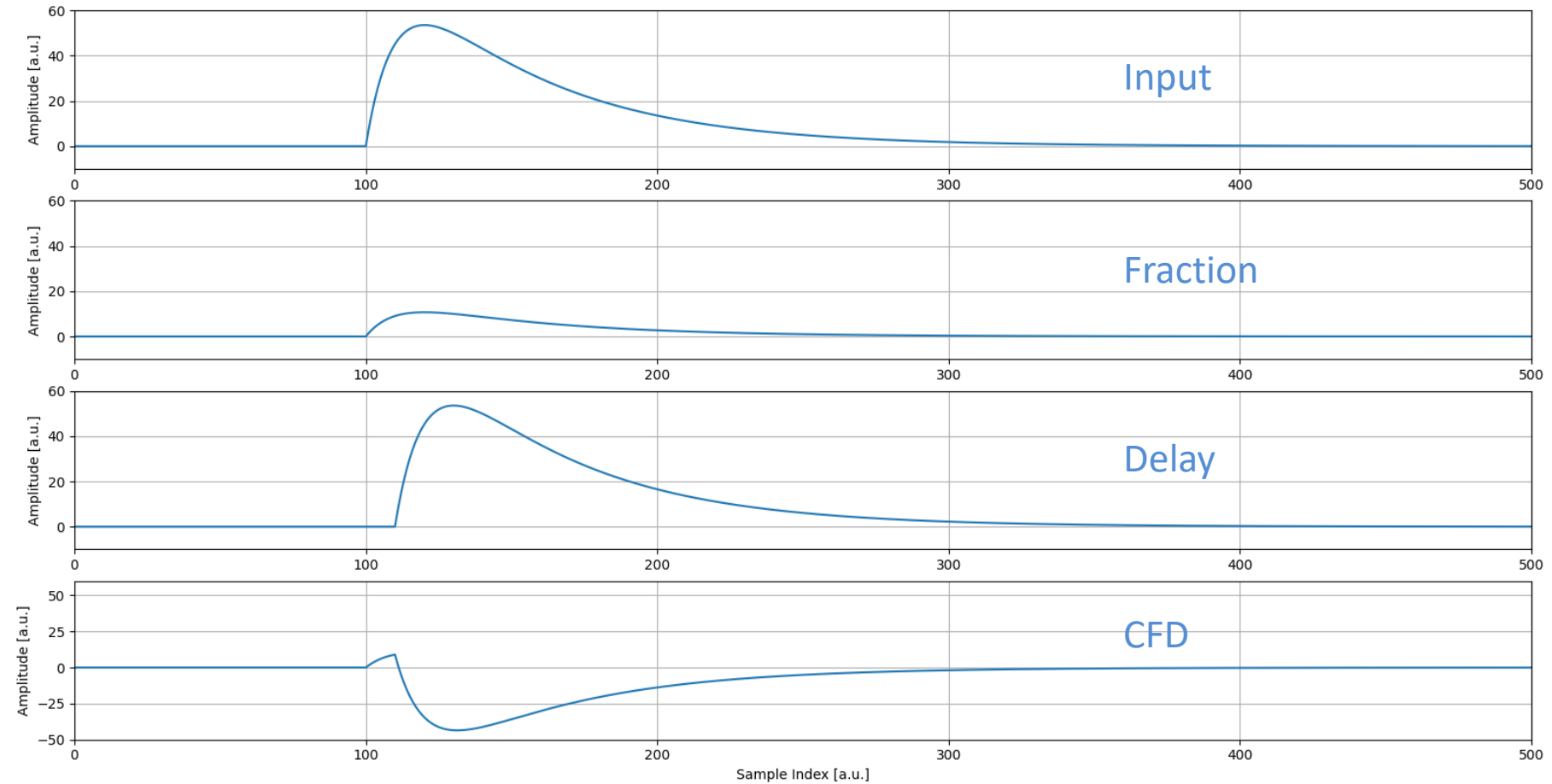
$$\cancel{kV_0} (e^{-t/\tau_f} - e^{-t/\tau_r}) = \cancel{V_0} (e^{-(t-t_d)/\tau_f} - e^{-(t-t_d)/\tau_r})$$

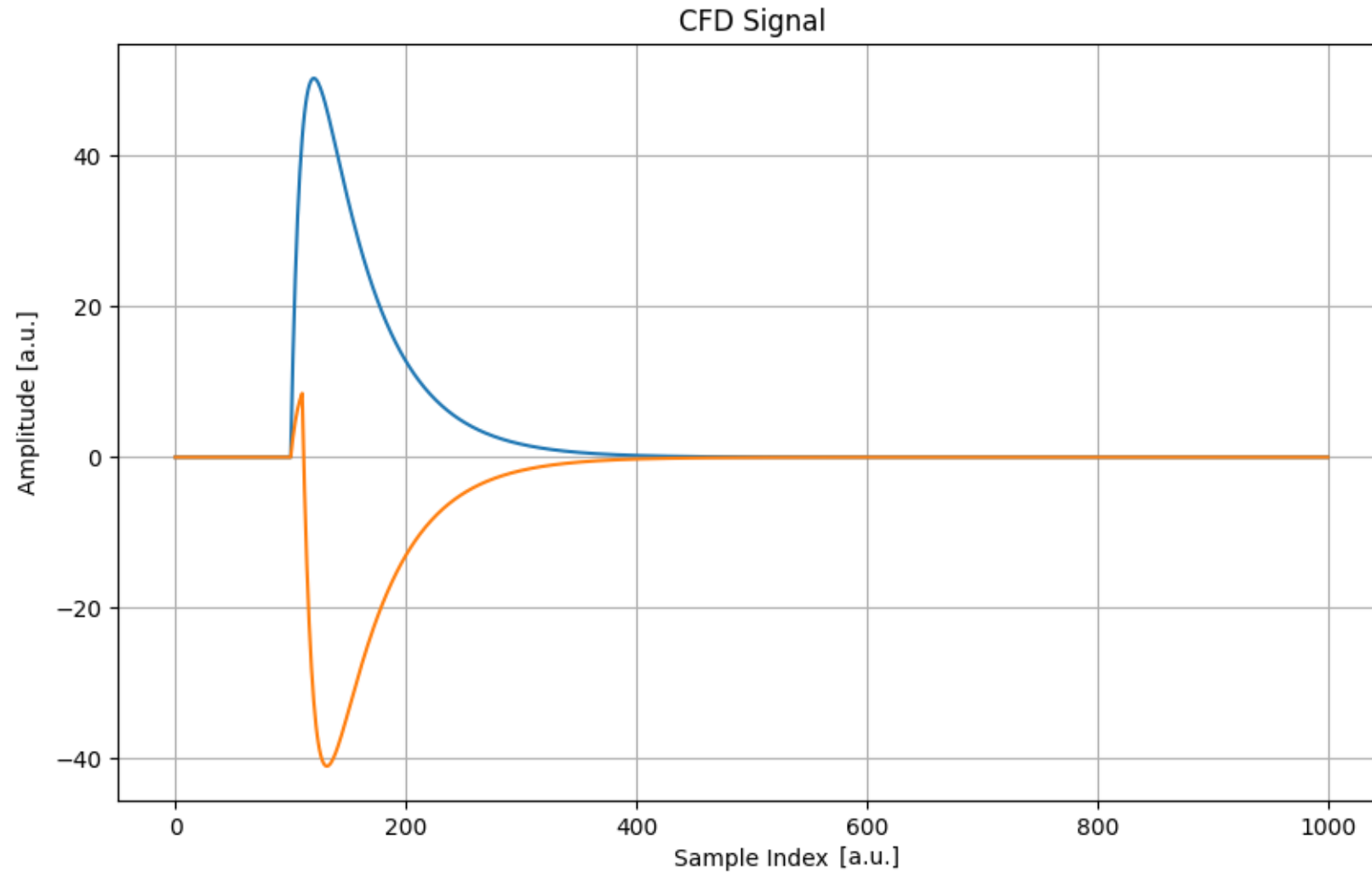
$$y[n] = k \cdot x[n] - x[n-N]$$

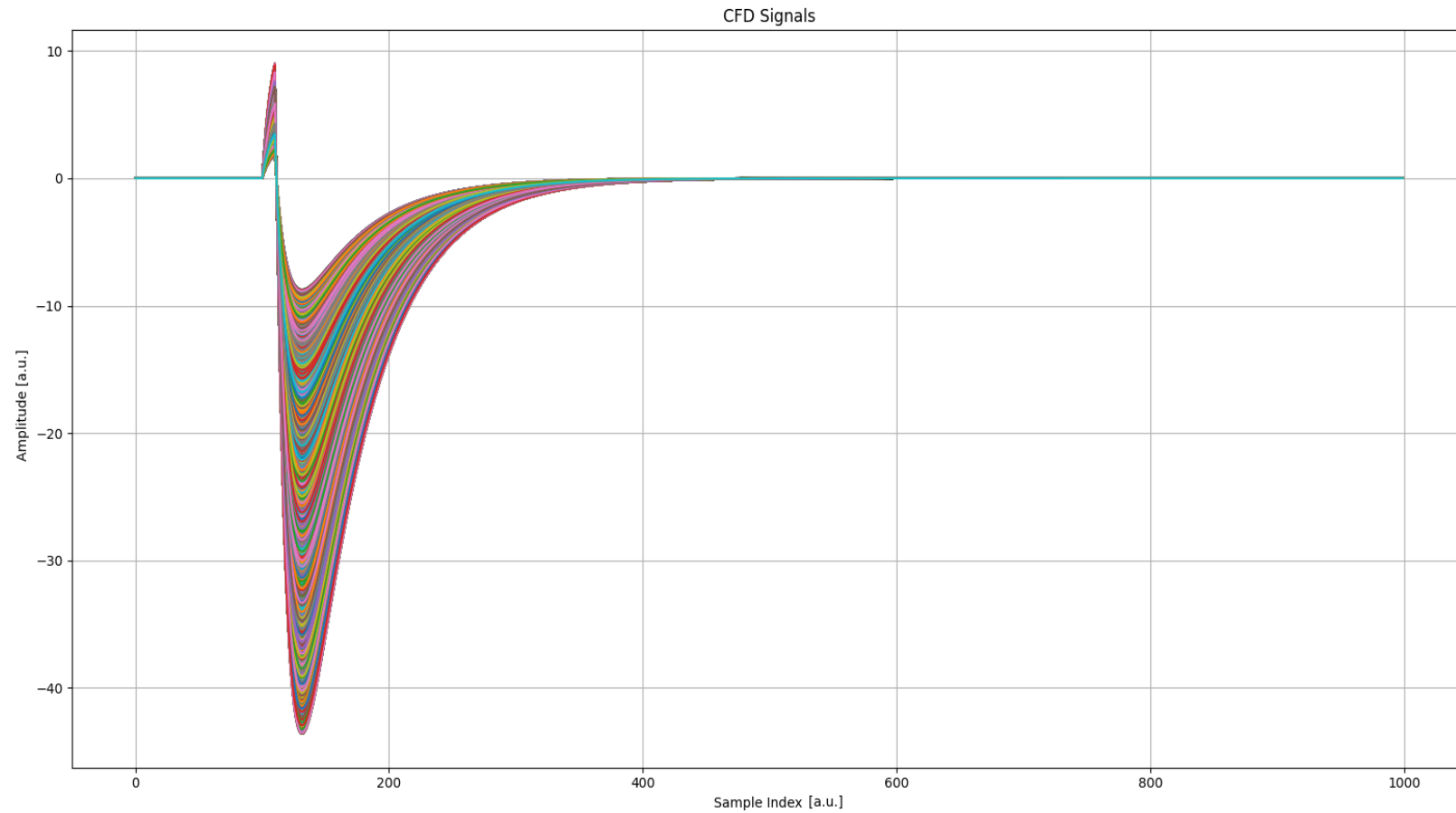
Constant Fraction Discriminator (CFD)

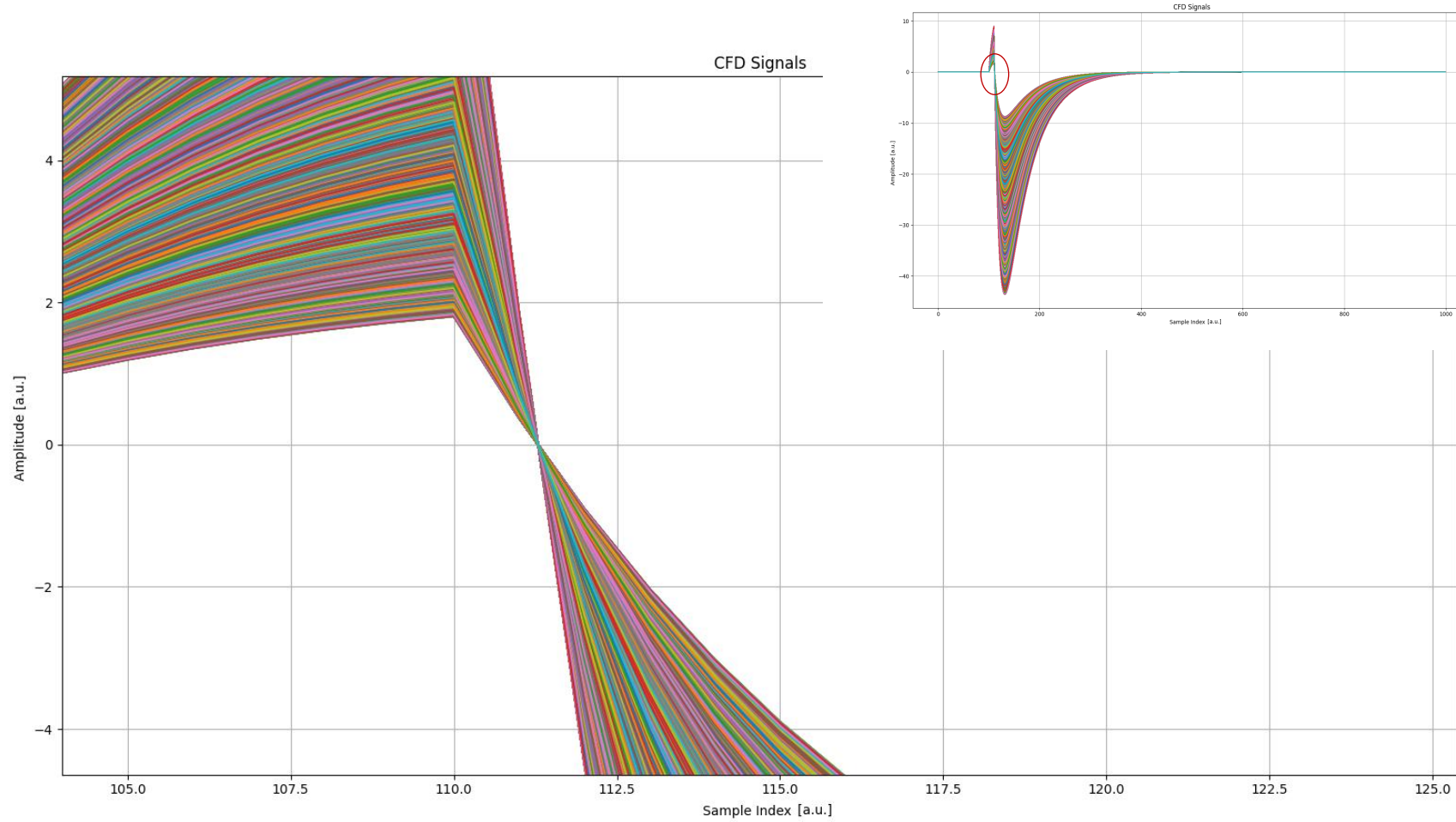


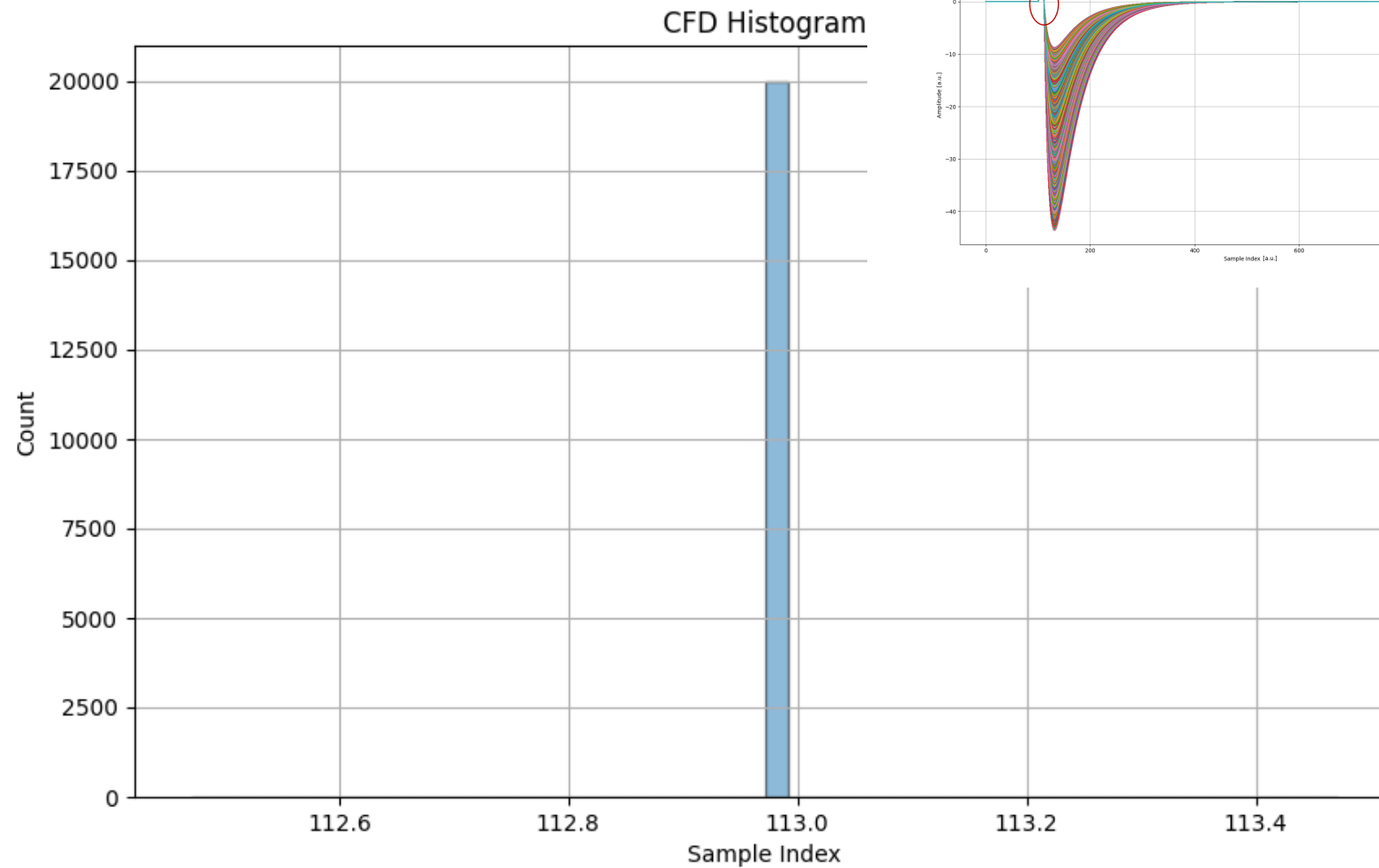
Constant Fraction Discriminator



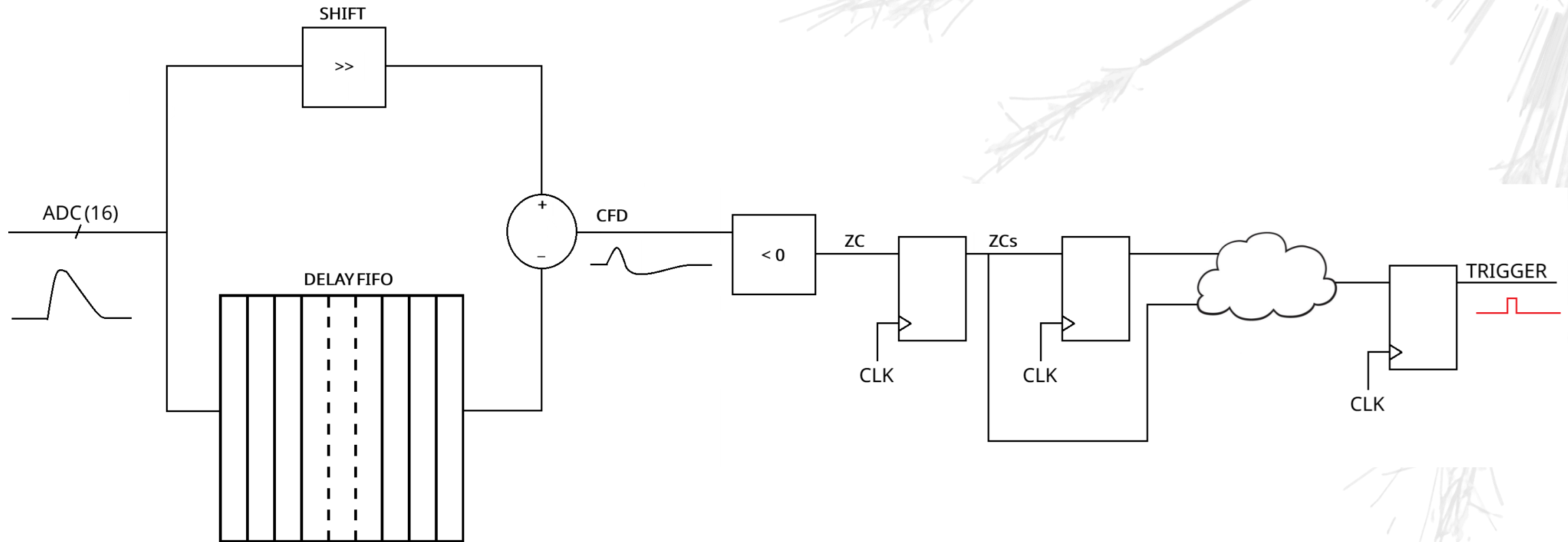




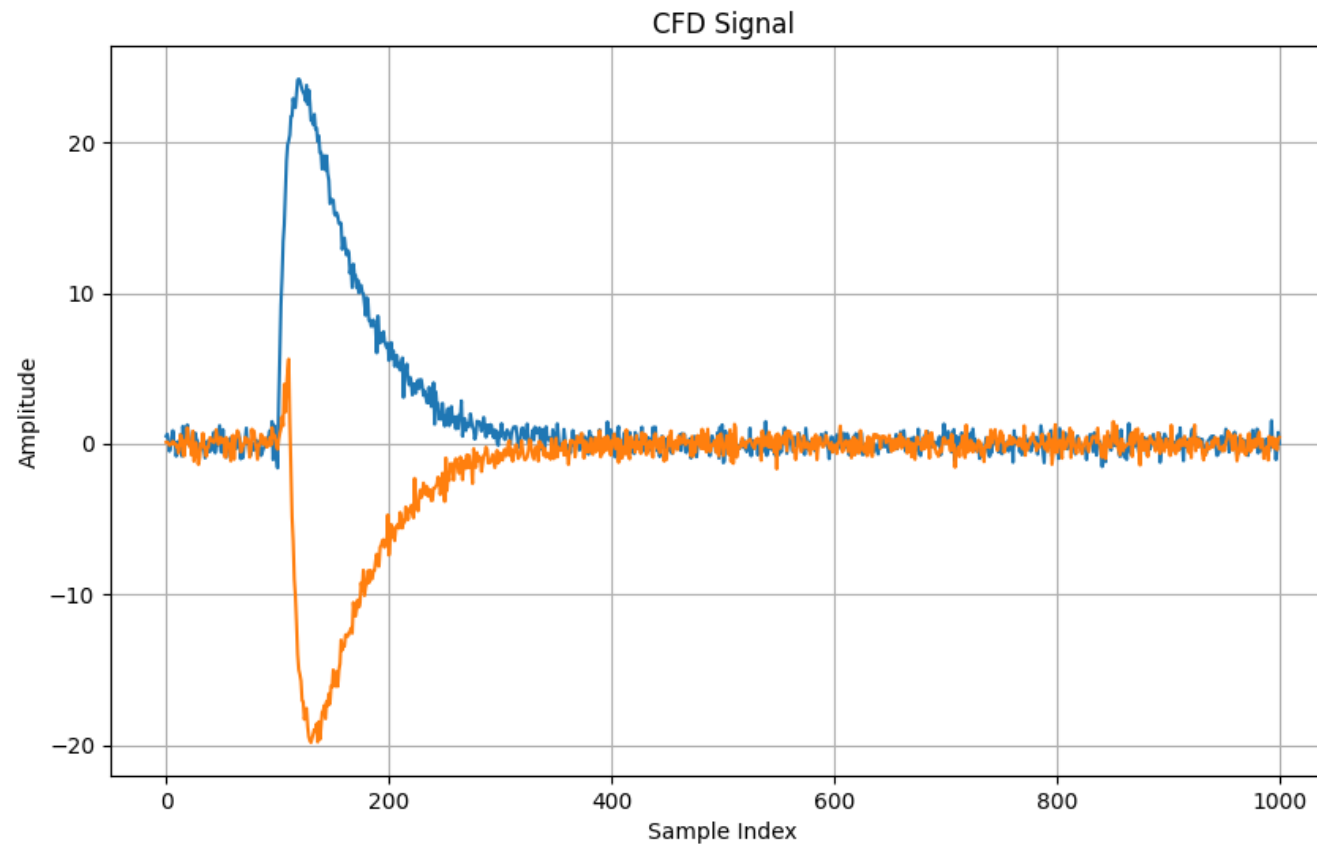




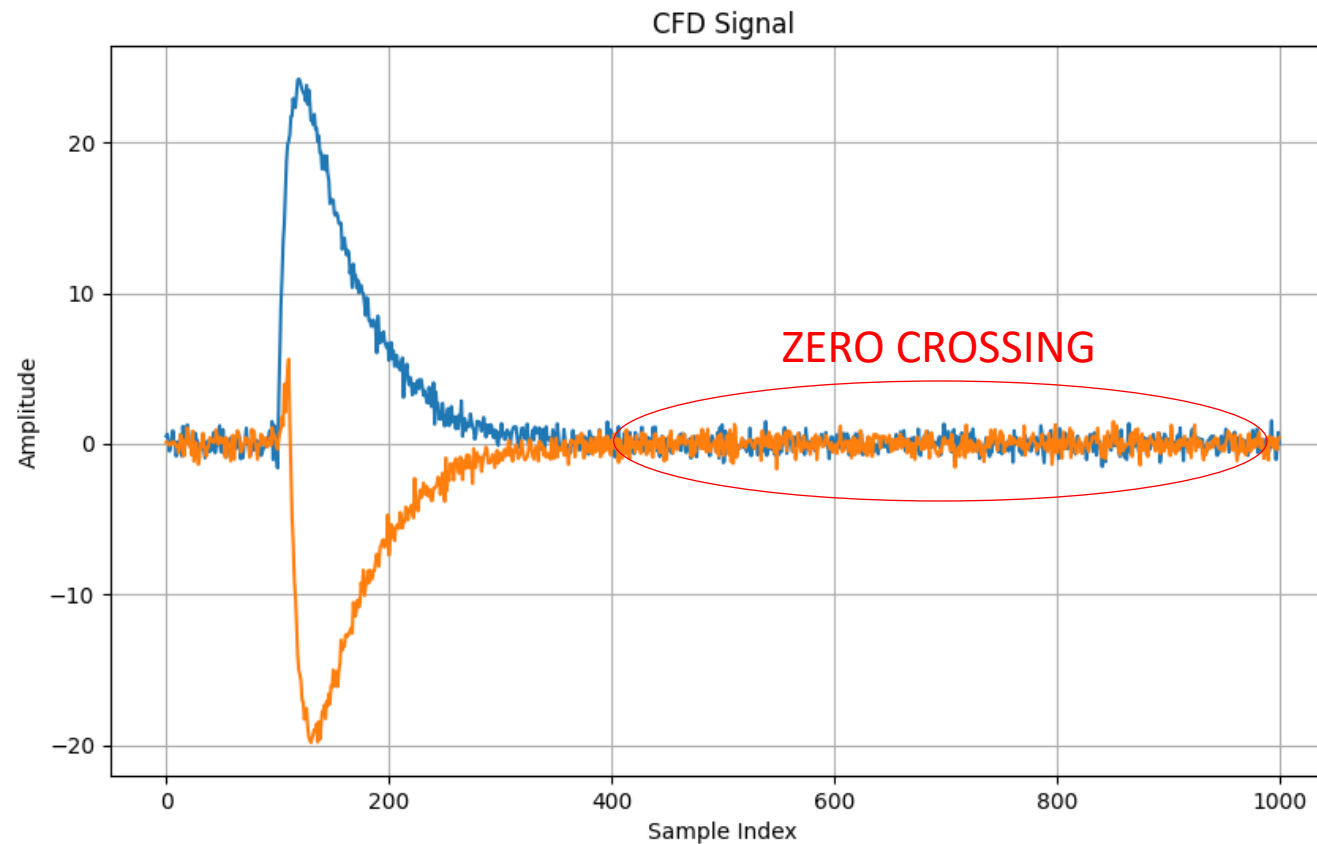
Constant Fraction Discriminator



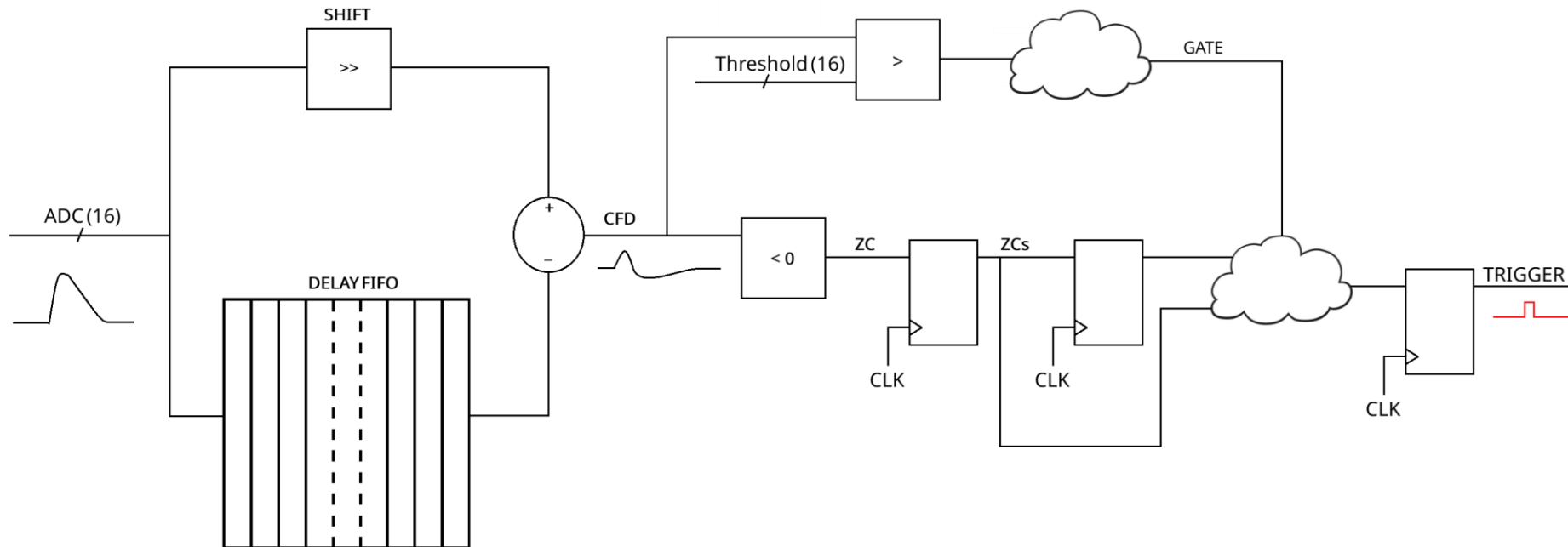
Noise Effects



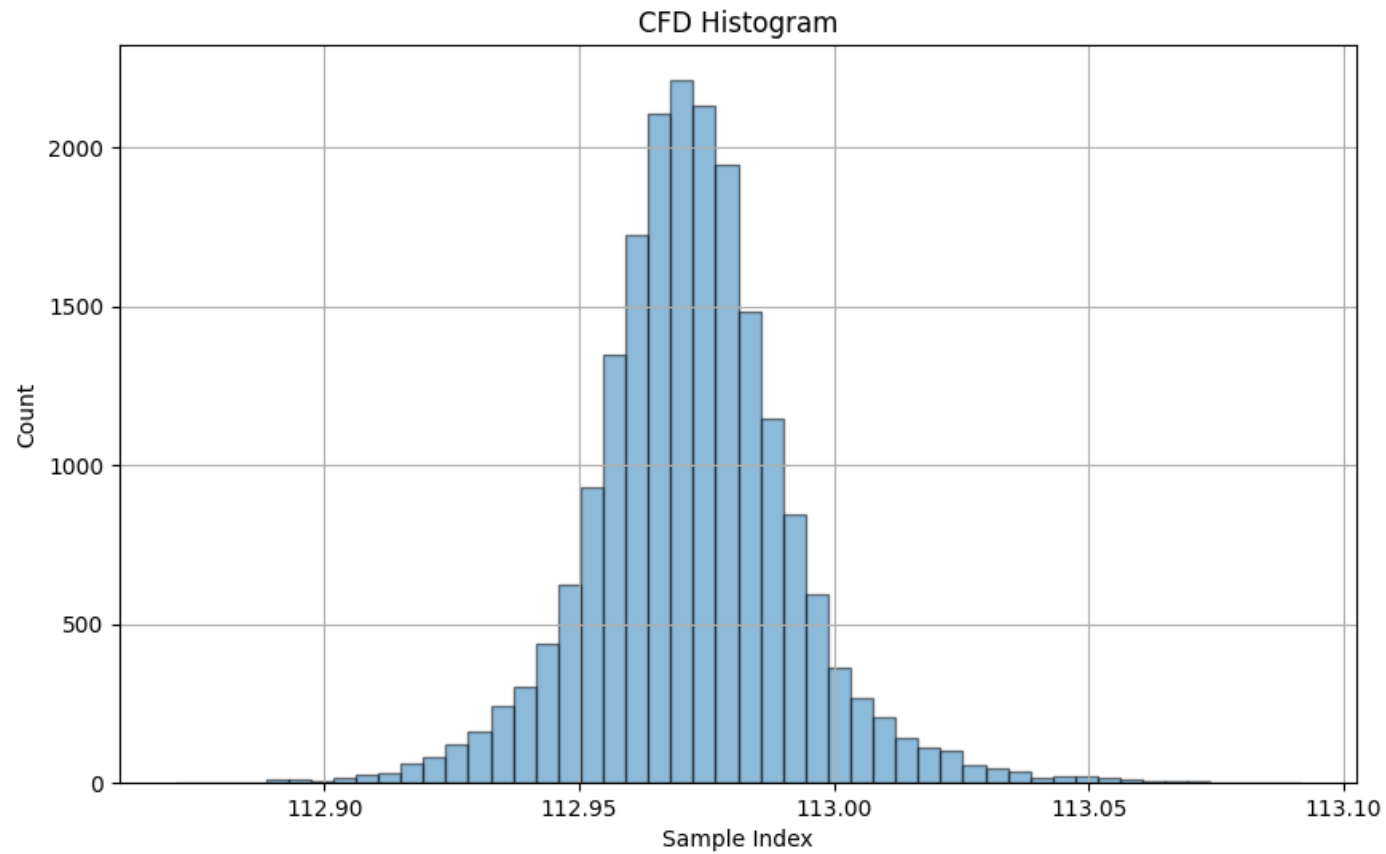
Noise Effects



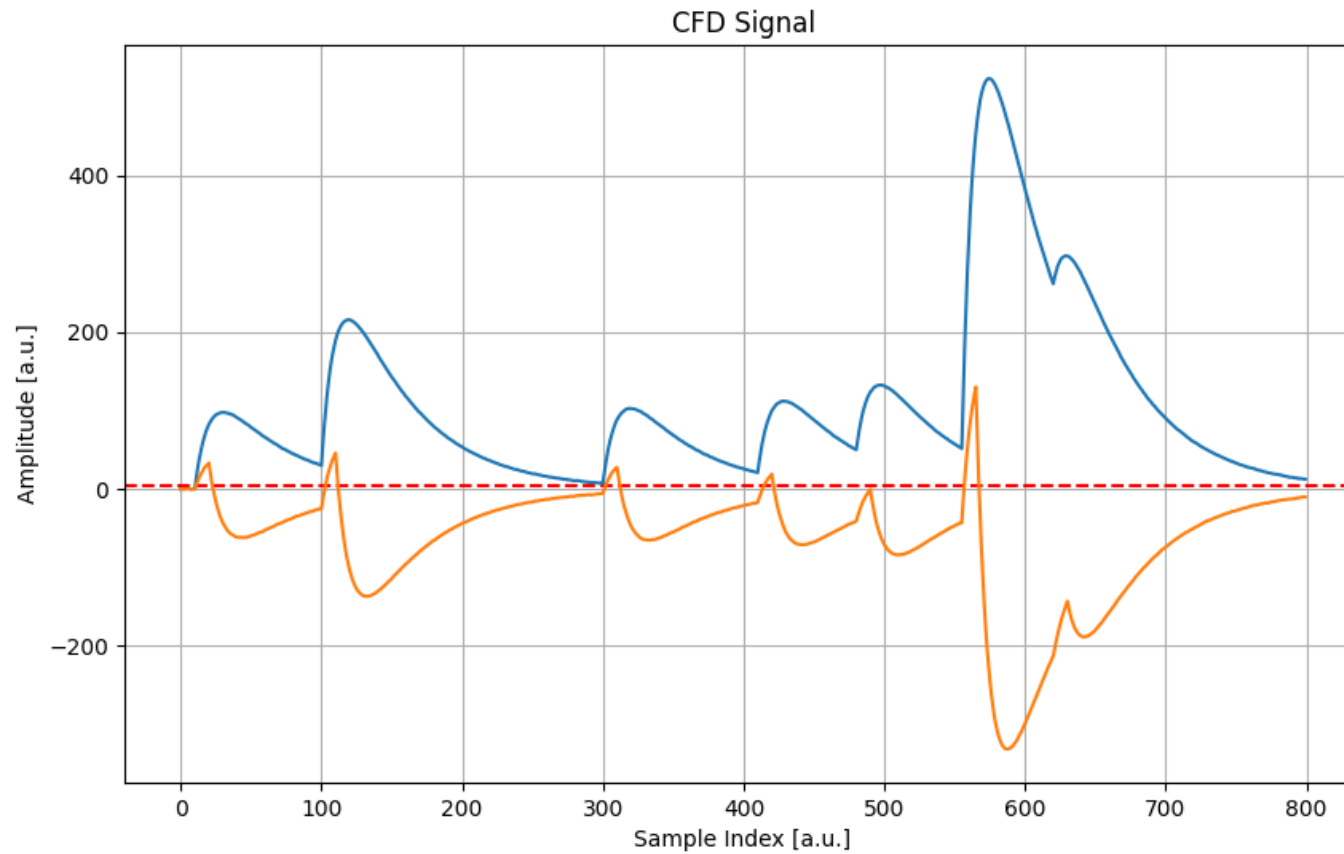
Noise Effects



Noise Effects

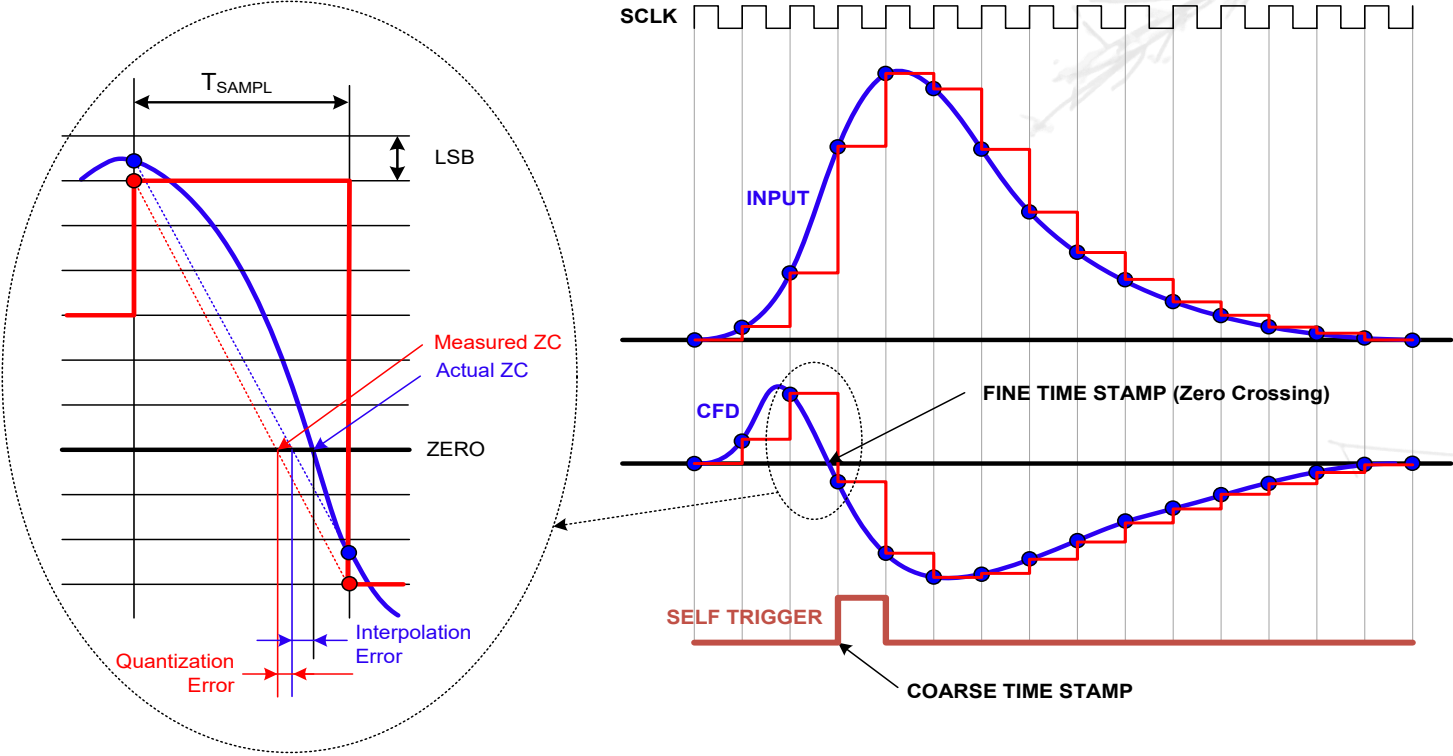


Pile-up Problem



	Leading Edge Discriminator	Constant Fraction Discriminator
Trigger Criterion	Absolute Threshold	Fixed Fraction
Time Walk	Significant	Eliminated
Complexity	Minimal	Medium
Rate Sensitivity	High Rate Pile Up	High Rate Pile Up
Application	First Level Trigger, Coarse Timing	ToF, Coincidence, Fine Timing

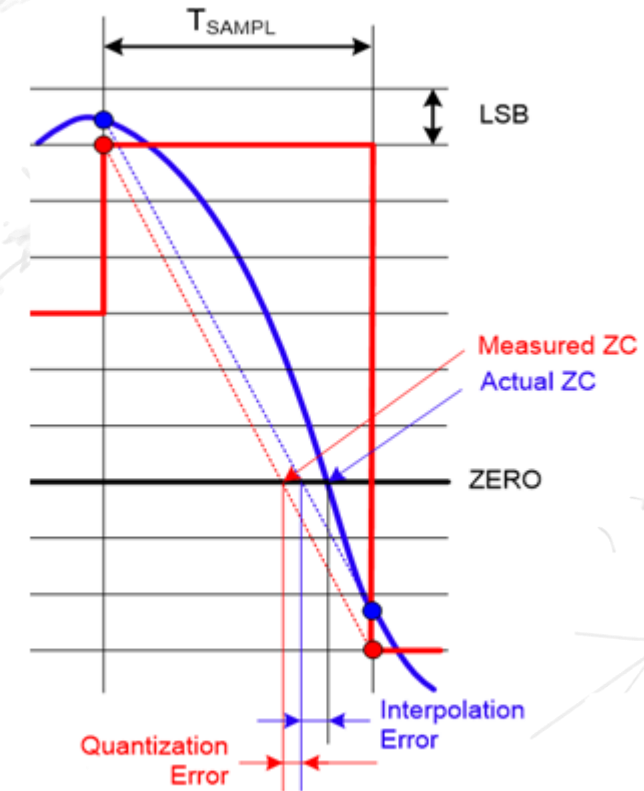
What about a Coarse Time Stamp improvement?



- The ADC gives you integers at fixed intervals
- The zero-crossing lives somewhere in between two integers
- Linear interpolation gives the fractional offset δ :

$$\delta = \frac{y[n]}{y[n] - y[n + 1]}$$

- Timing resolution then becomes $\delta \cdot T_s$, well below the sampling period
- **On an FPGA: no floating-point division**



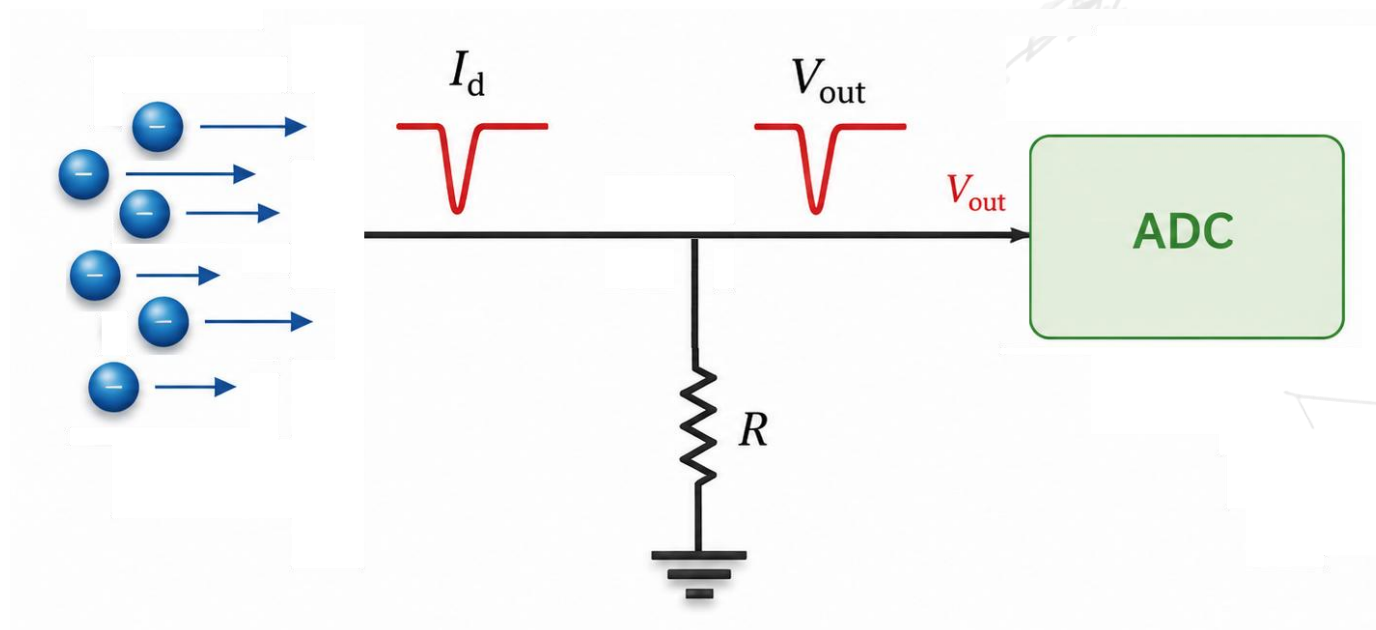
Idea: Binary search between two samples — no divider, no float, deterministic latency.

1. Evaluate the midpoint m (sum + shift)
2. If $m > 0$, zero crossing is in the second half
3. If $m < 0$, zero crossing is in the first half
4. Repeat for k -iterations

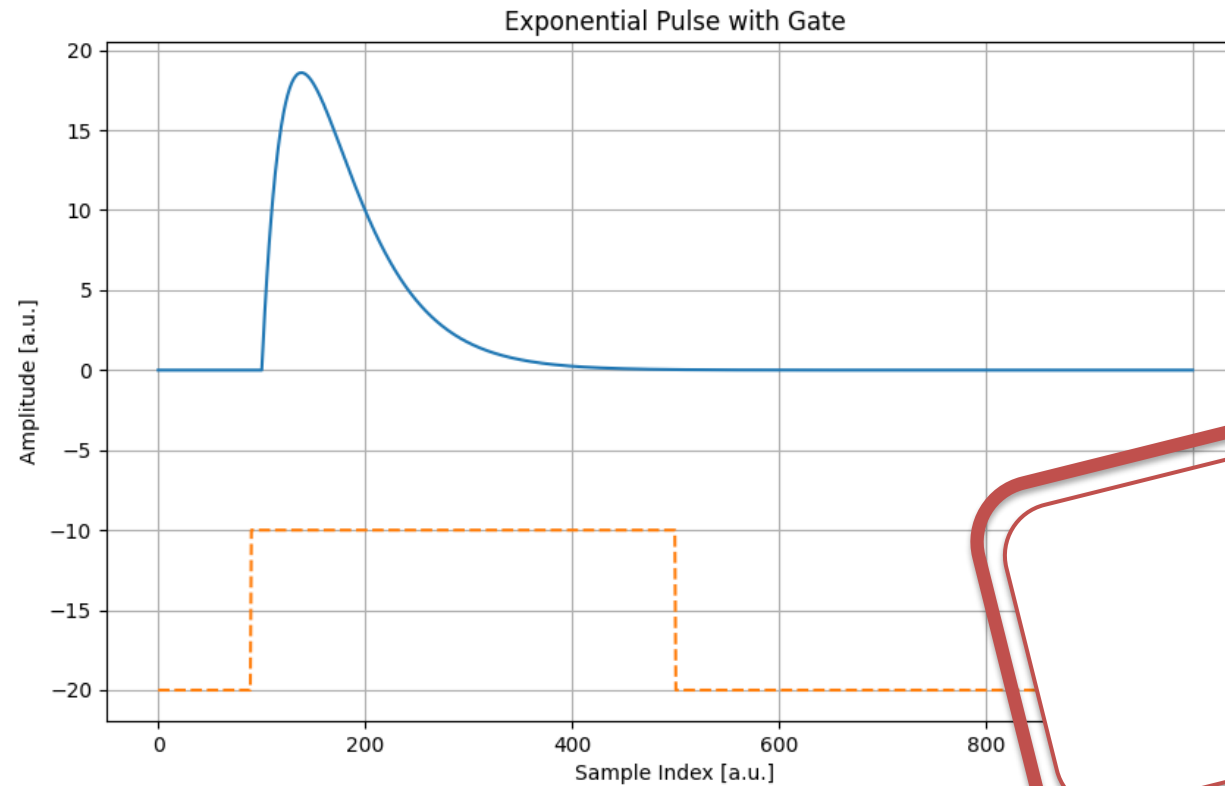
SUM and SHIFT are easy on FPGA

Energy Filters

$$Q = \sum_{n=a}^{a+N} x[n]$$

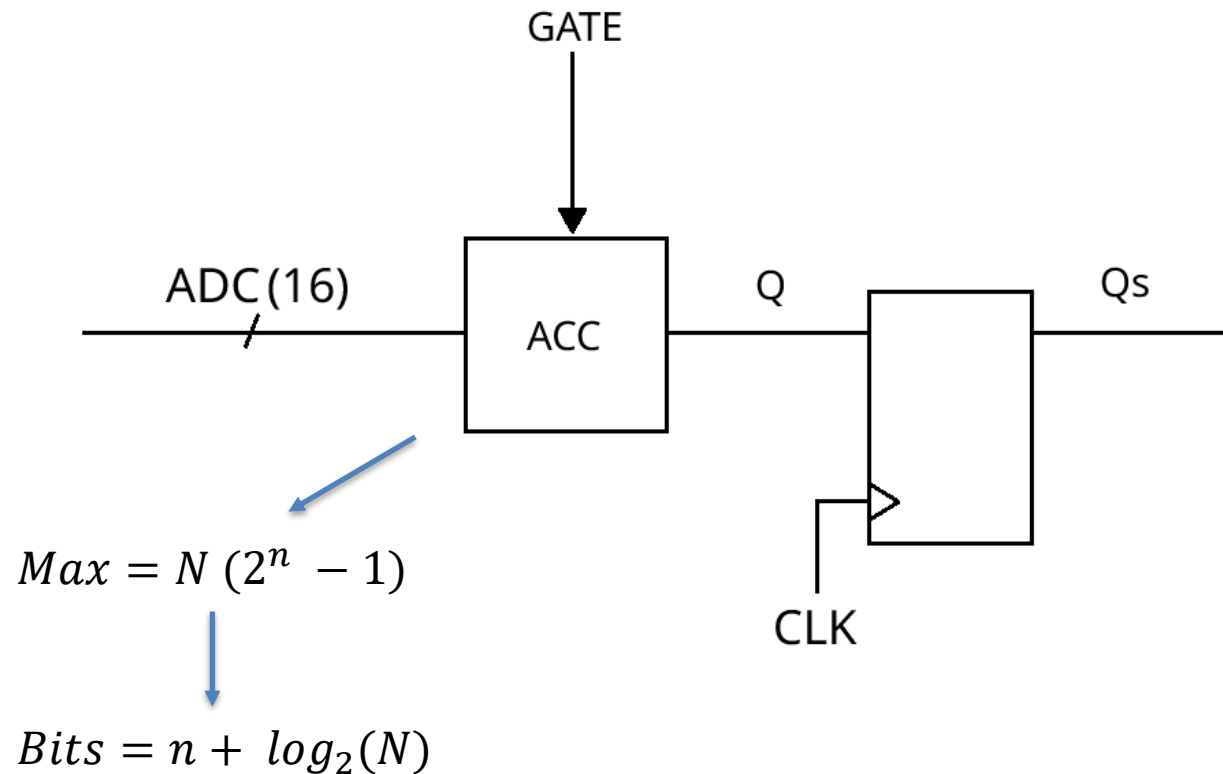


$$Q = \sum_{n=a}^{a+N} x[n]$$

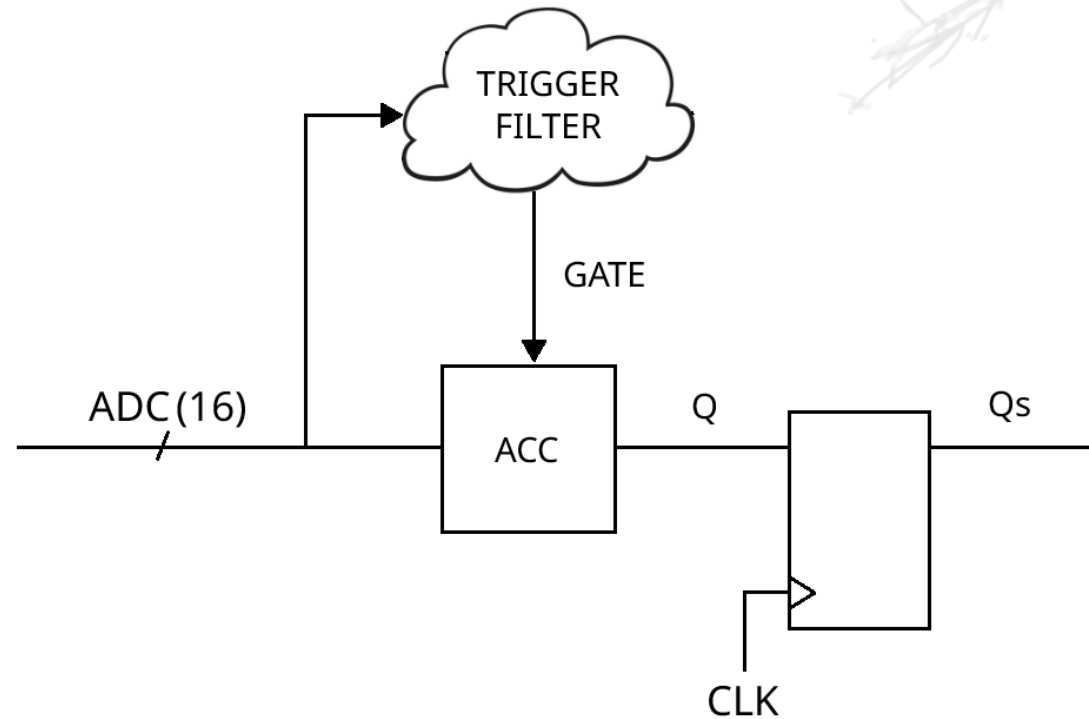


SIMPLE ?!?

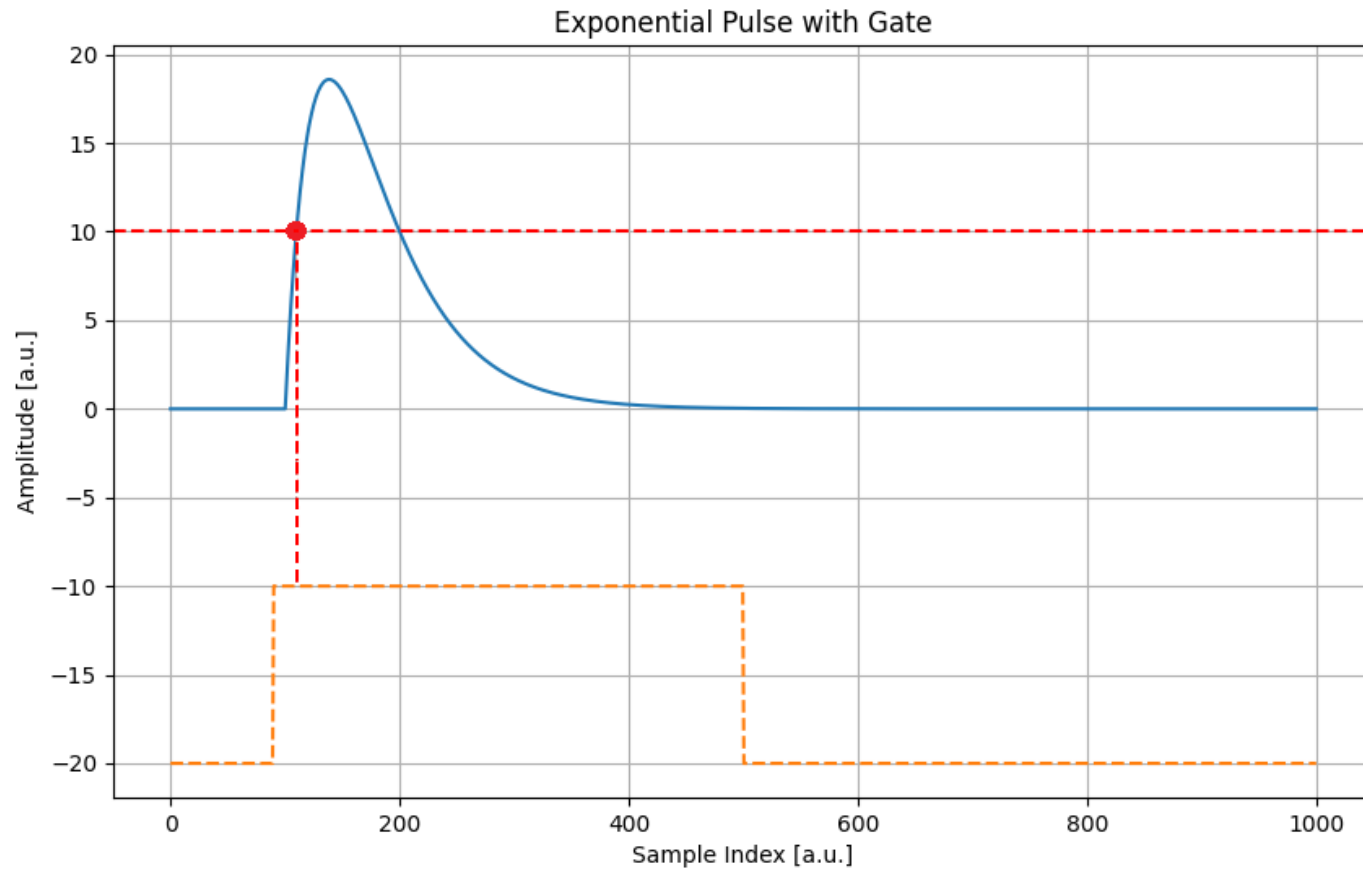
$$Q = \sum_{n=a}^{a+N} x[n]$$



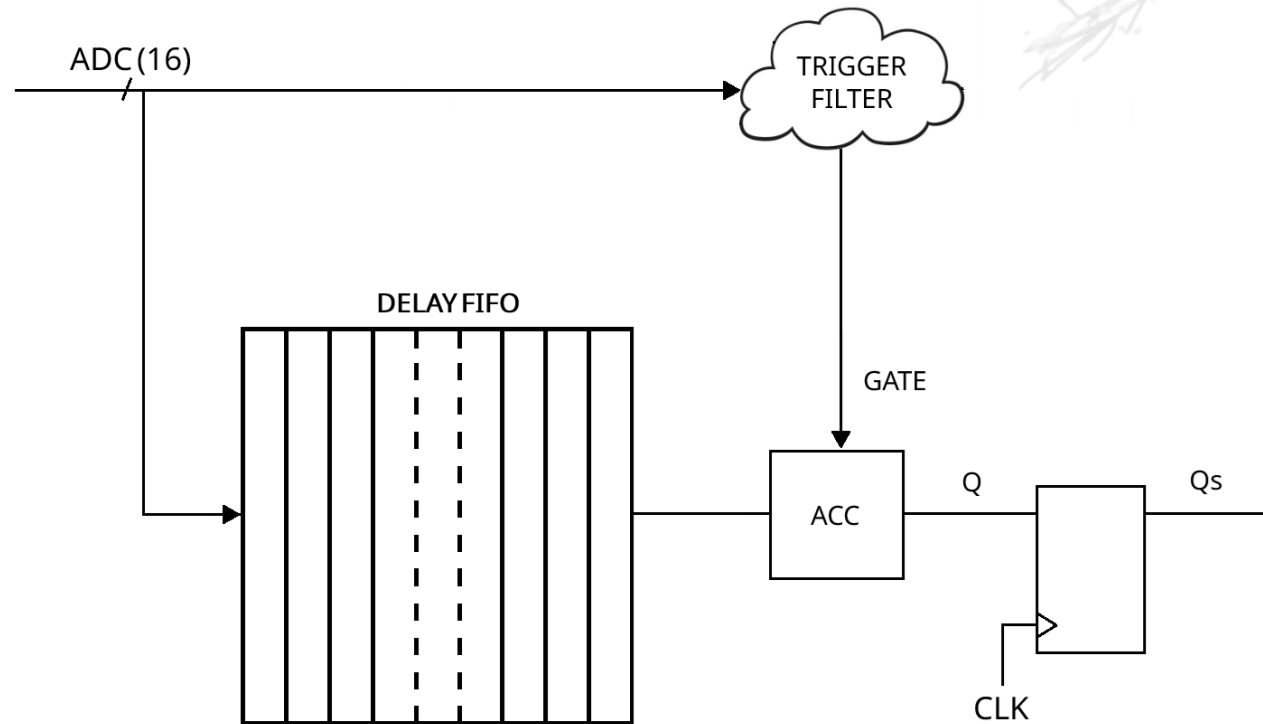
$$Q = \sum_{n=a}^{a+N} x[n]$$



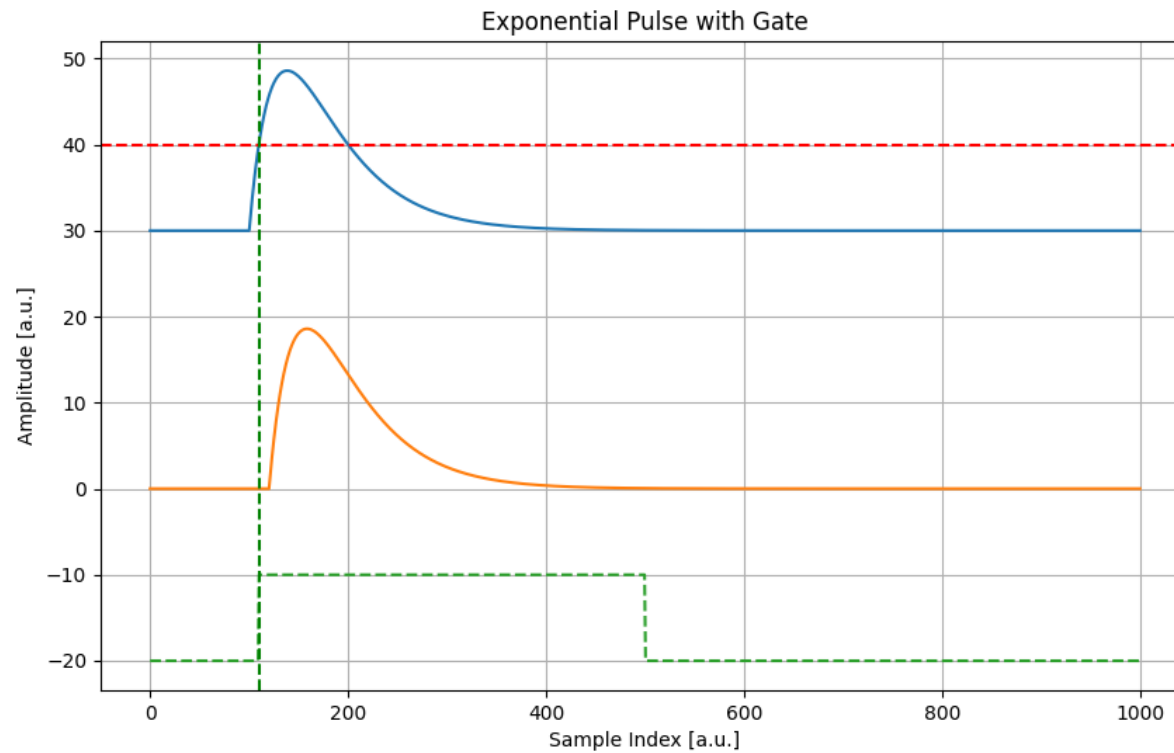
$$Q = \sum_{n=a}^{a+N} x[n]$$



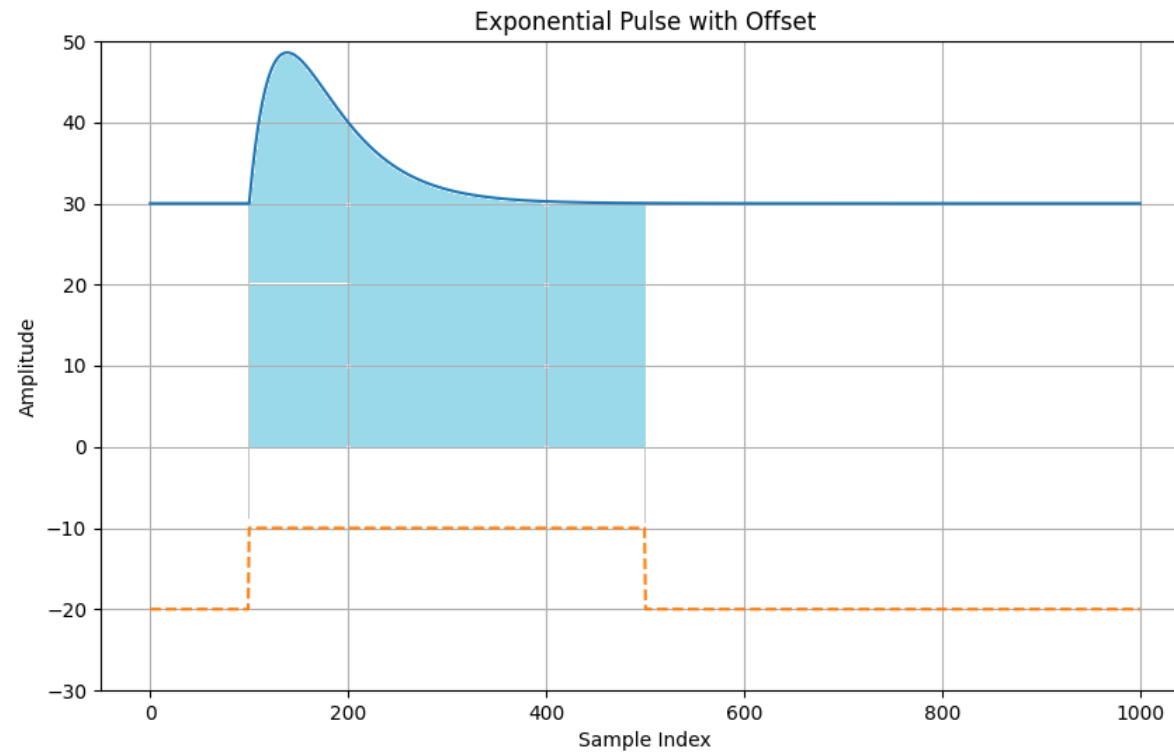
$$Q = \sum_{n=a}^{a+N} x[n]$$



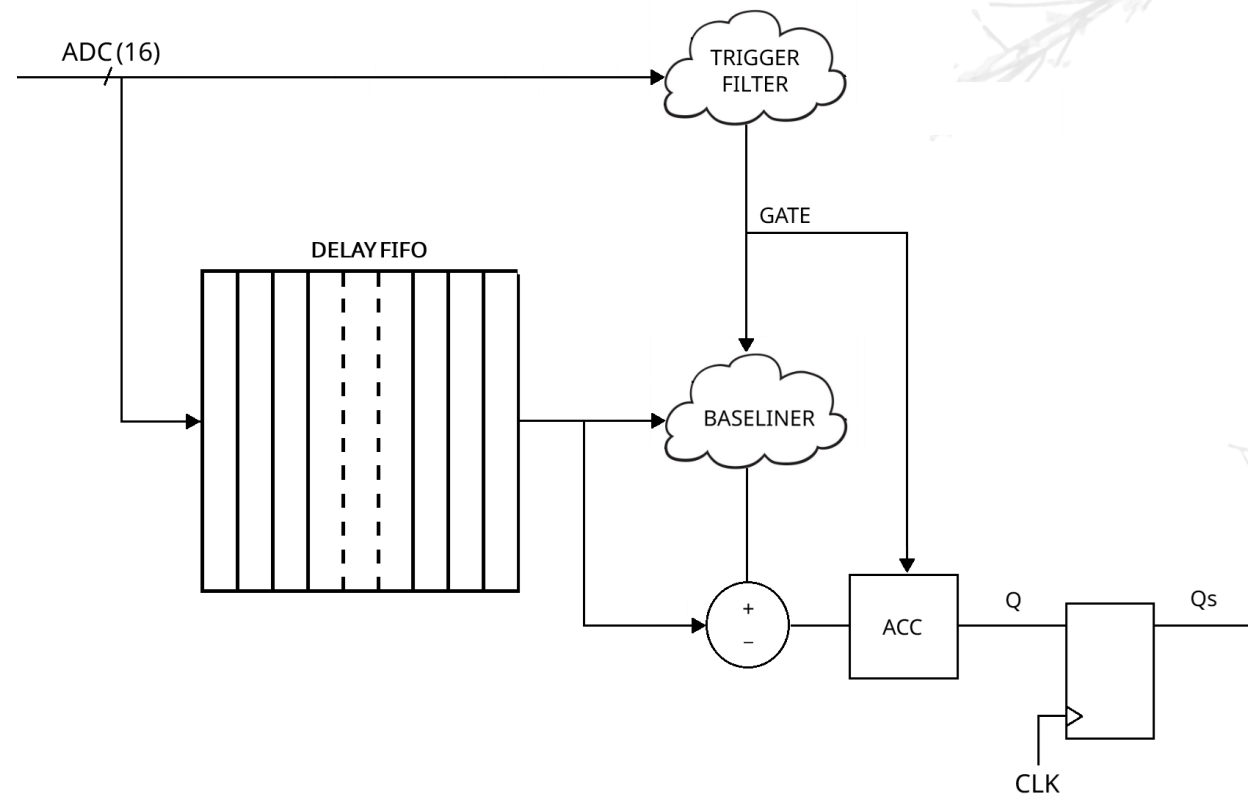
$$Q = \sum_{n=a}^{a+N} x[n]$$



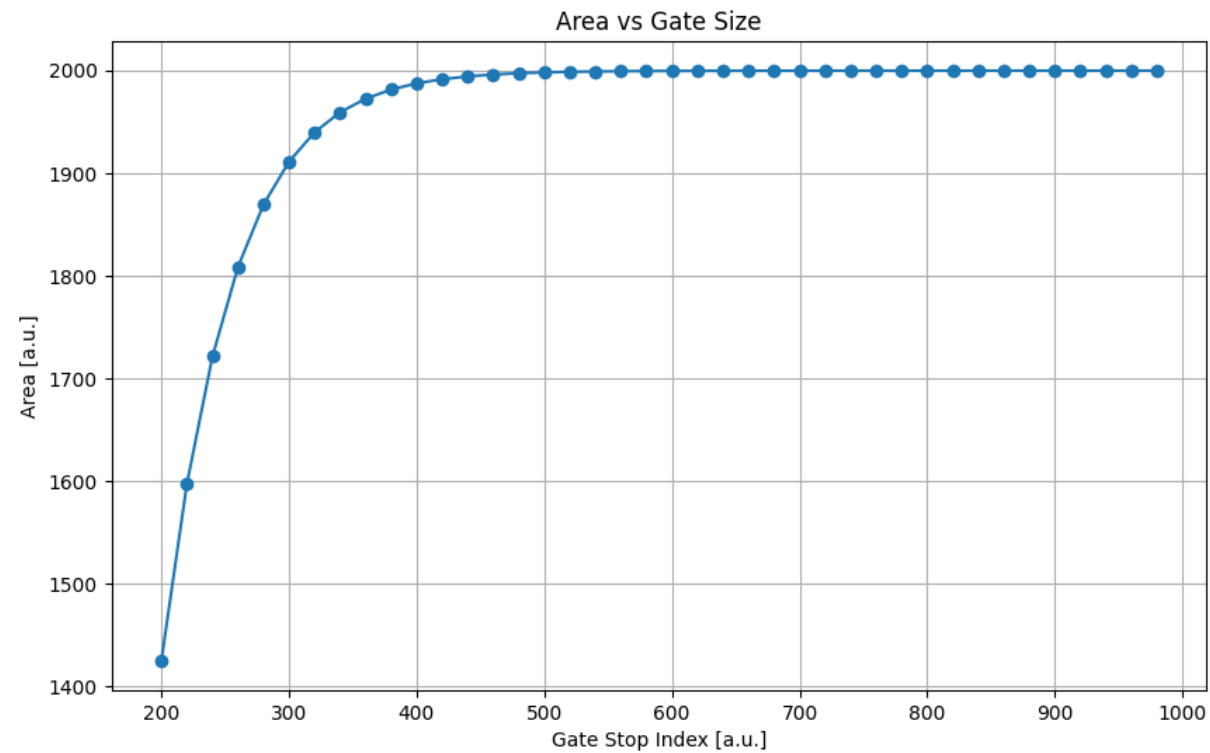
$$Q = \sum_{n=a}^{a+N} x[n]$$



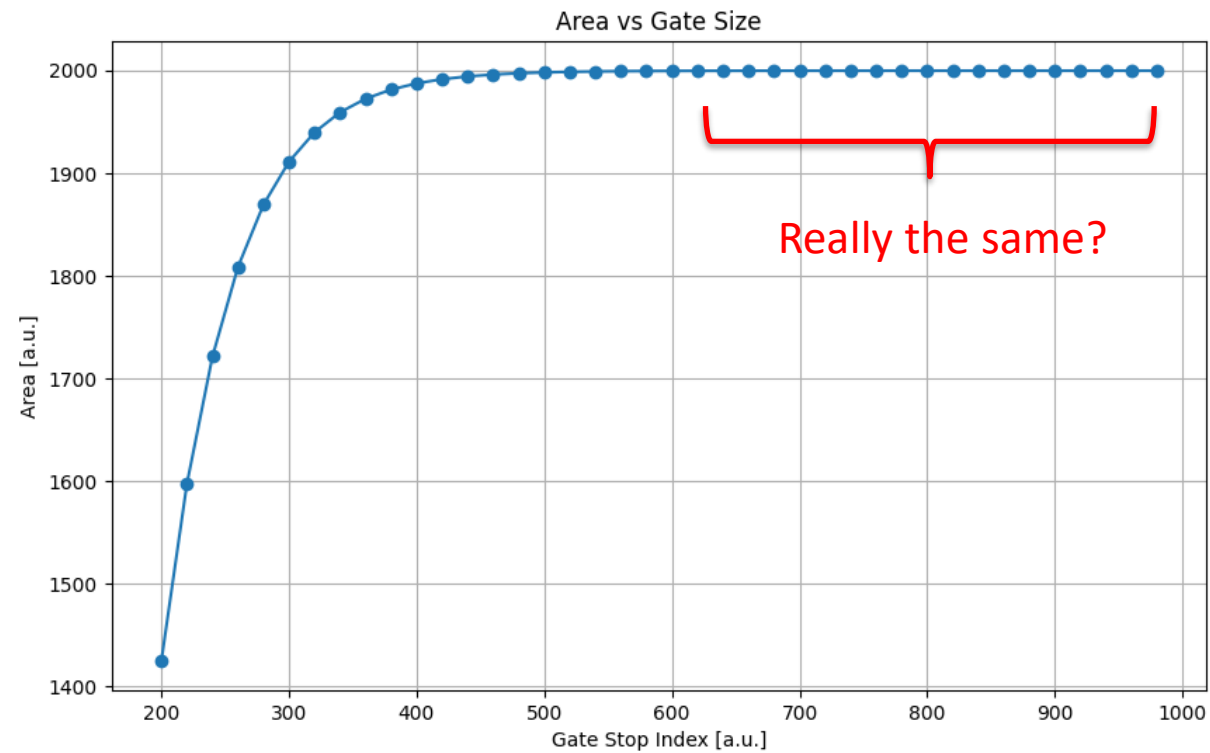
$$Q = \sum_{n=a}^{a+N} x[n]$$



$$Q = \sum_{n=a}^{a+N} x[n]$$

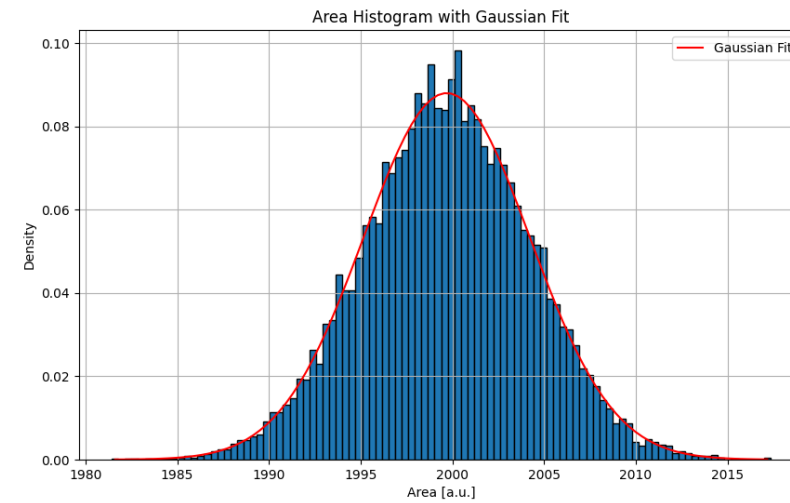
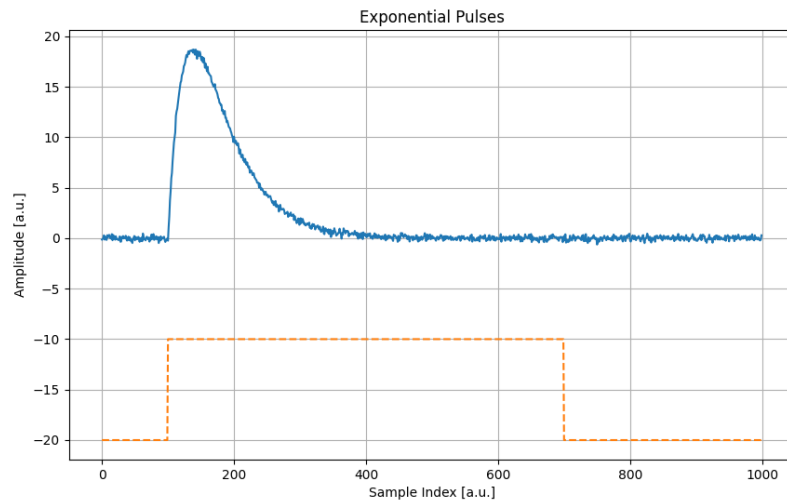


$$Q = \sum_{n=a}^{a+N} x[n]$$

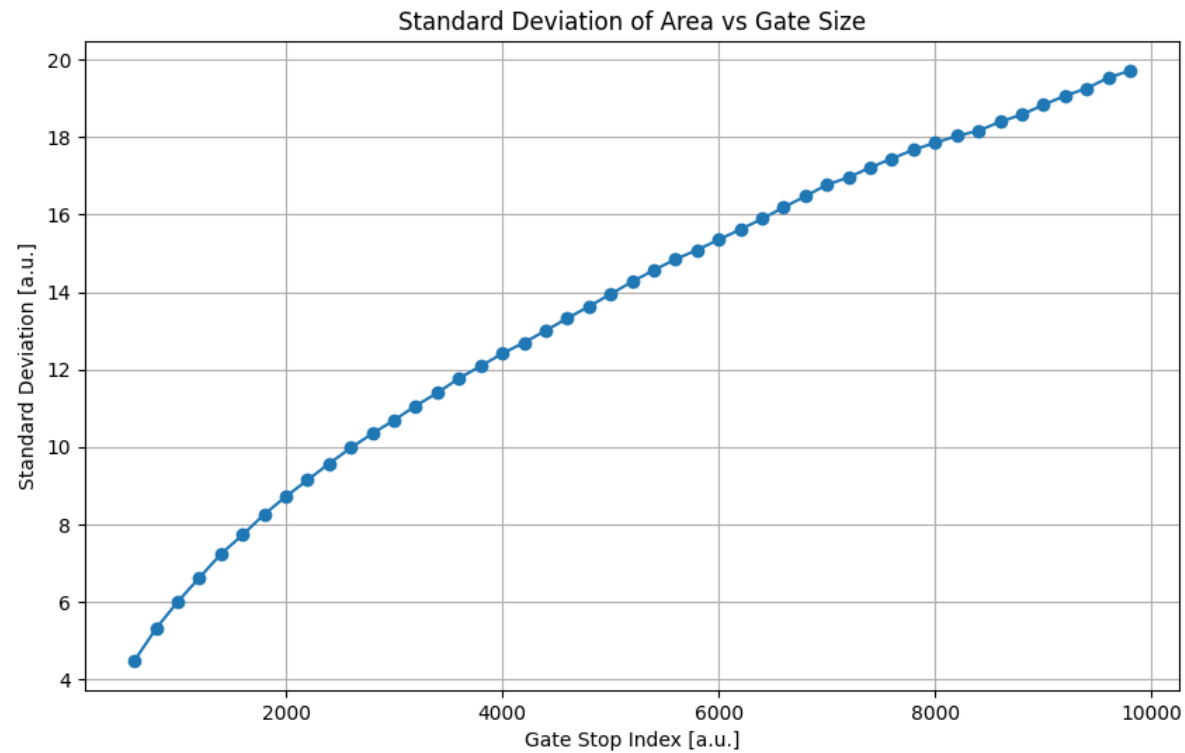


$$Q = \sum_{n=a}^{a+N} x[n]$$

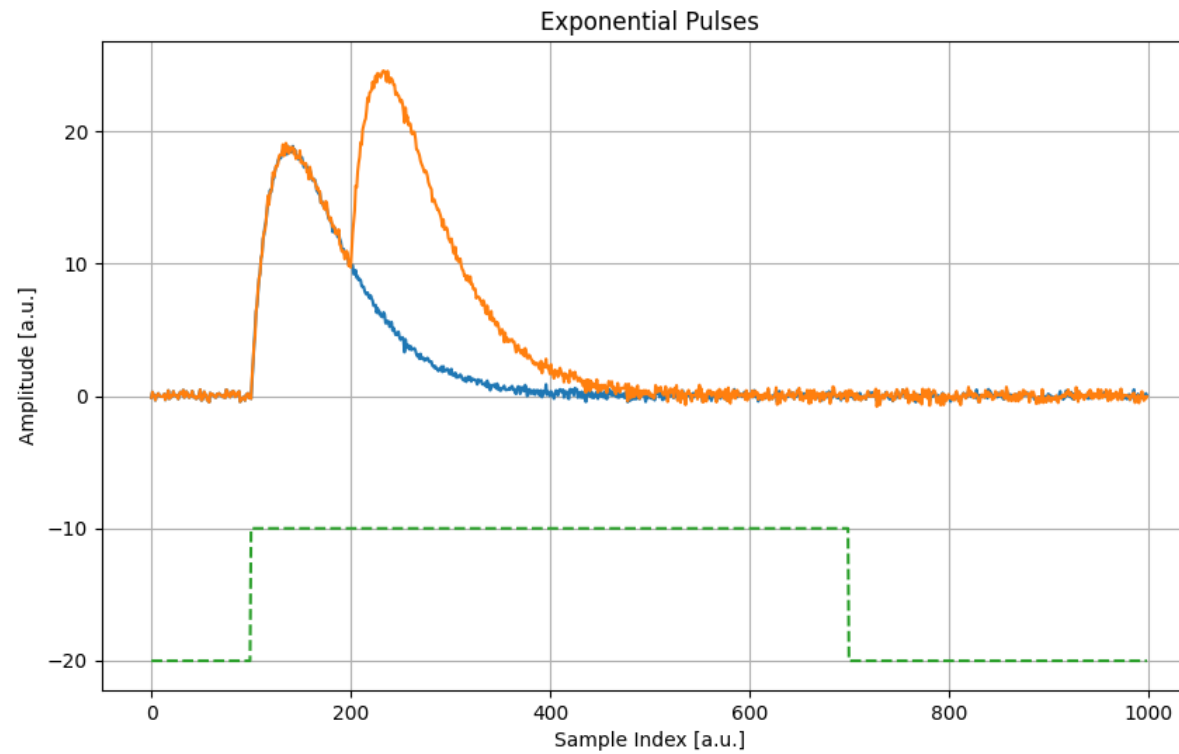
$$\sigma_{tot}^2 = \sum_{n=a}^{a+N} \sigma^2 = N\sigma^2$$



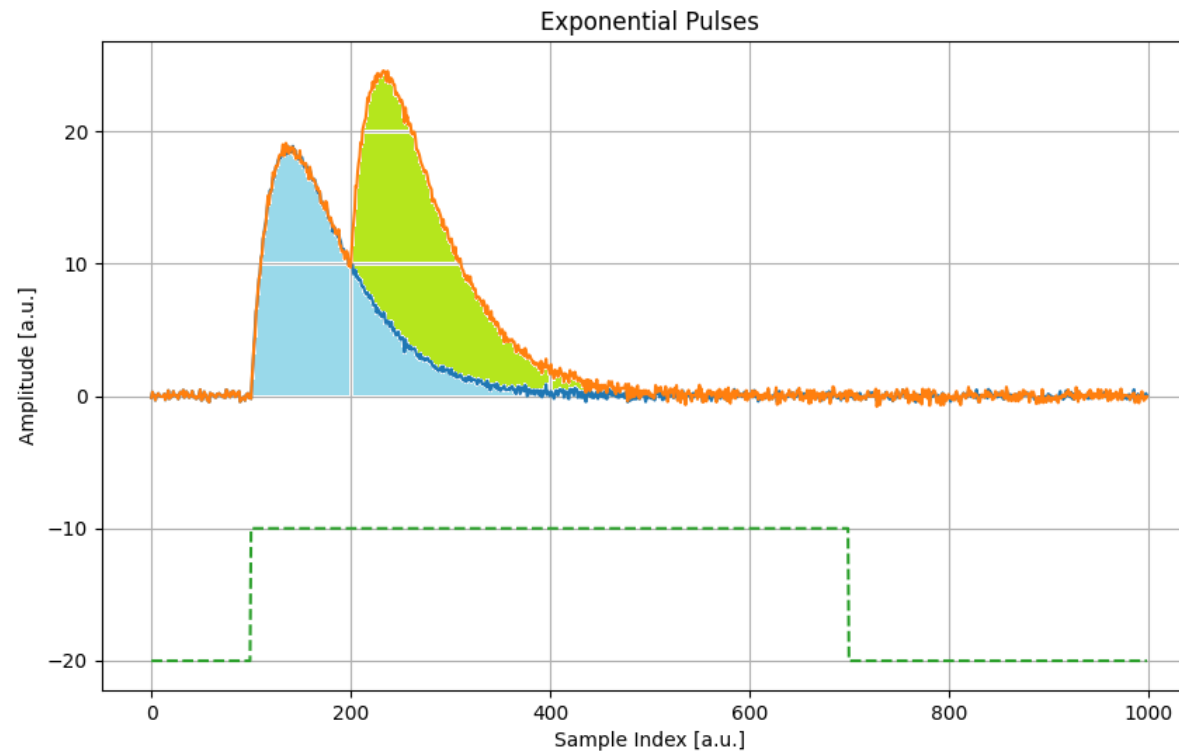
$$\sigma_{tot} = \sigma\sqrt{N}$$



Pile-up Limit



Pile-up Limit



PROs

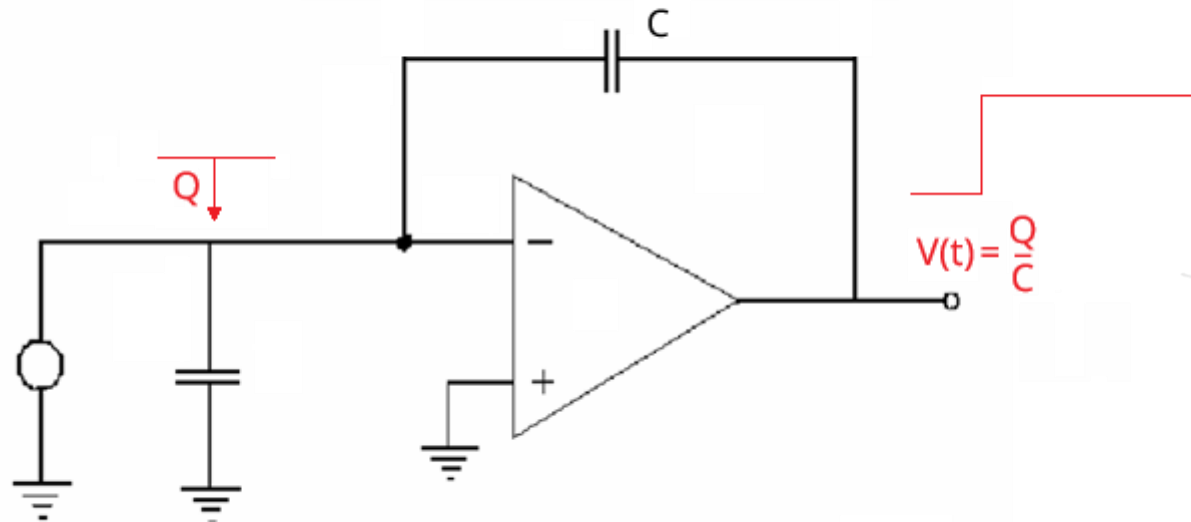
- Hardware Efficiency
- Minimal Parametric Option
- Shape Independence

CONs

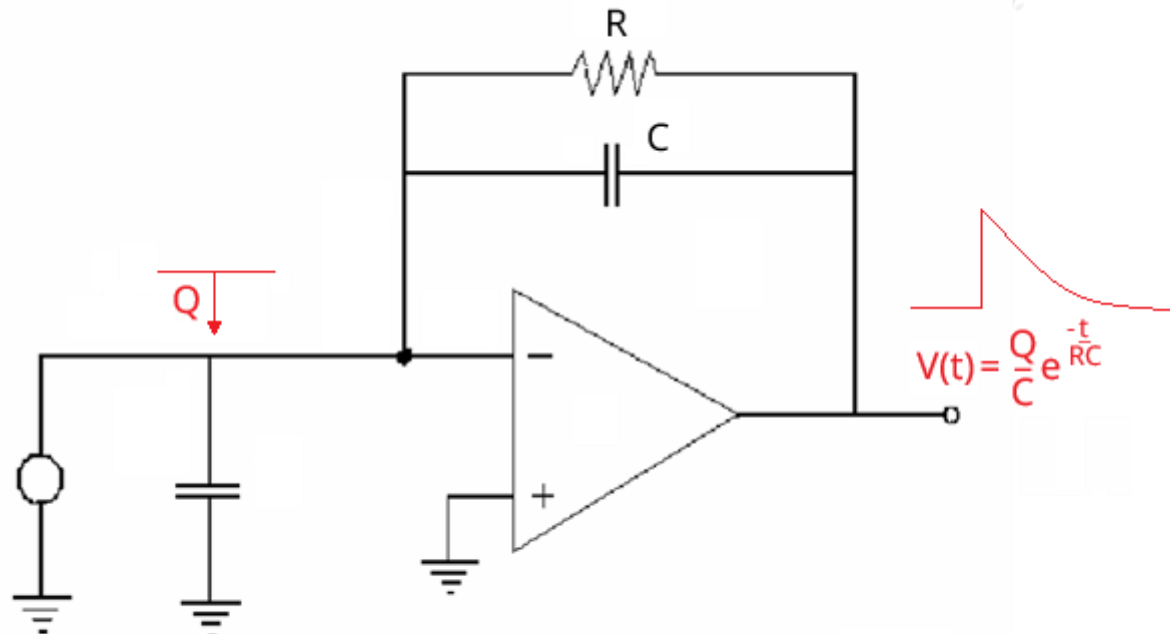
- No bandpass filter
- Noise Penalty
- Pile-up Vulnerability

... but what if the charge integration is done by a charge preamplifier ?!?

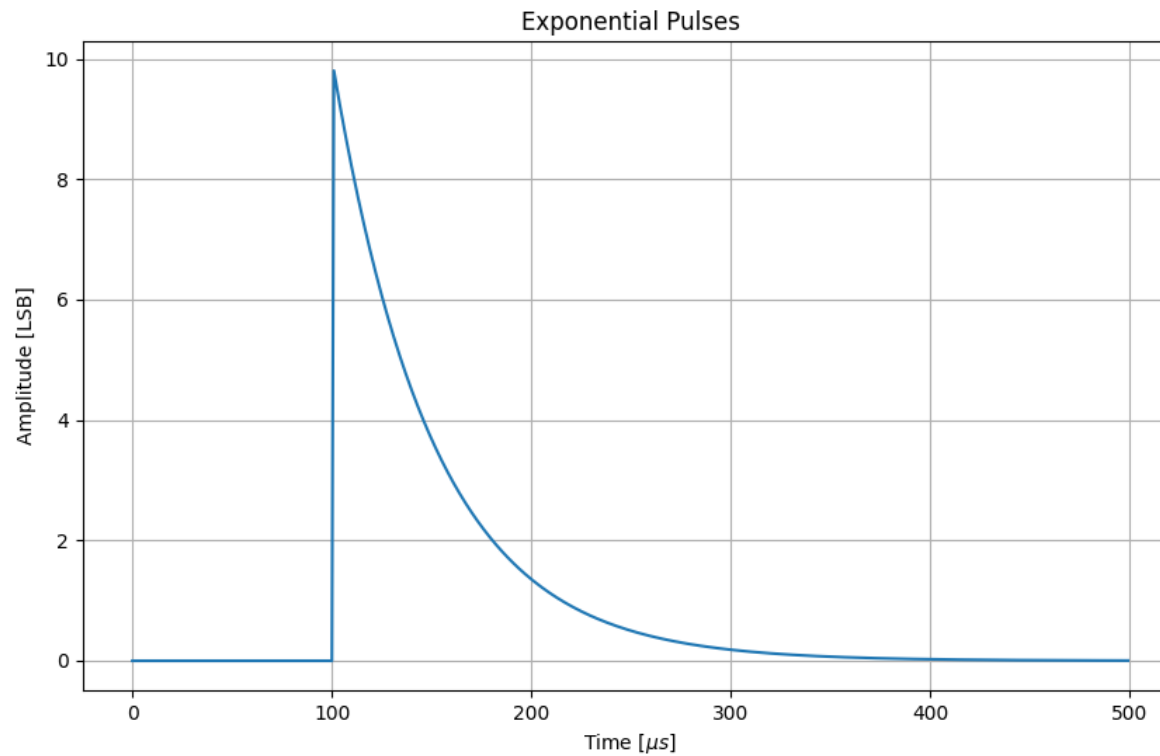
$$V(t) = \begin{cases} 0, & t < 0 \\ \frac{Q}{C}, & t \geq 0 \end{cases}$$



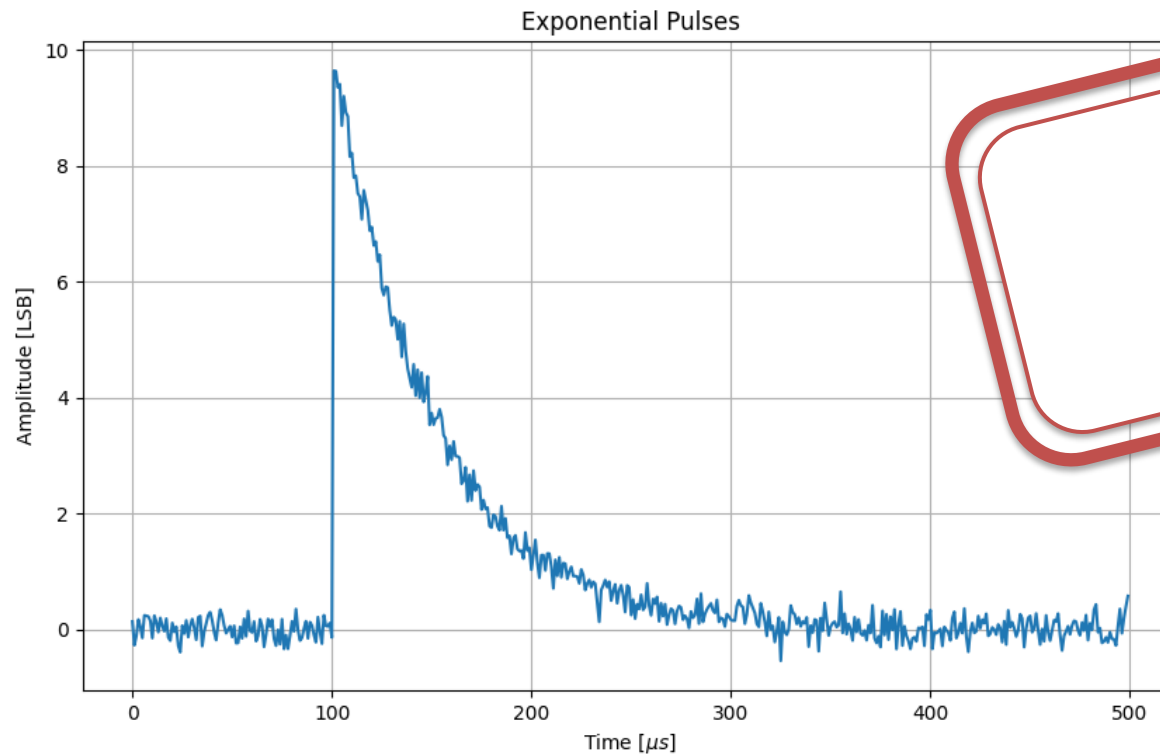
$$V(t) = \begin{cases} 0, & t < 0 \\ \frac{Q}{C} e^{-\frac{t}{RC}}, & t \geq 0 \end{cases}$$



$$Q \propto Vma_x$$

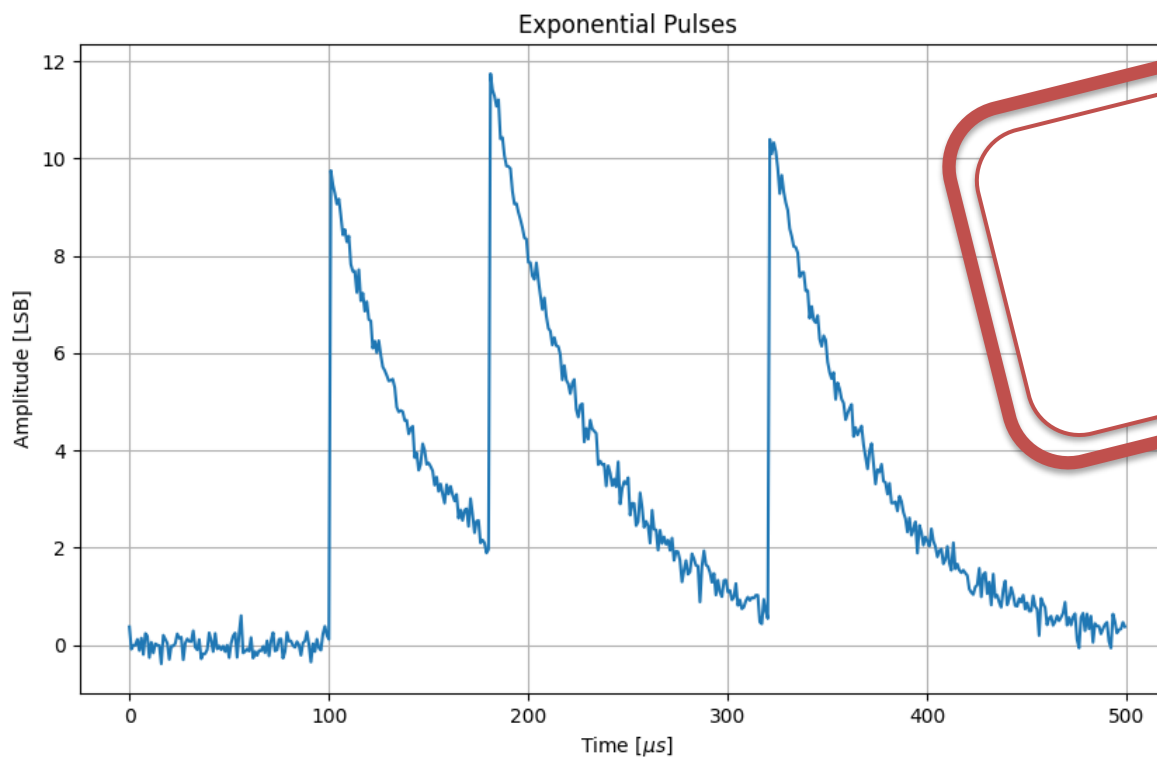


$$Q \propto V_{max}$$

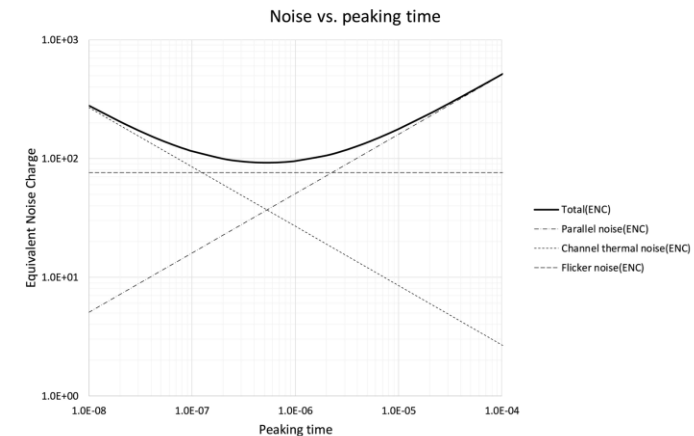
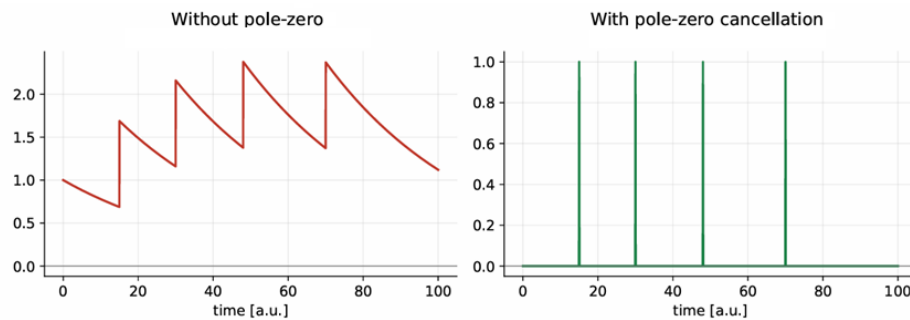
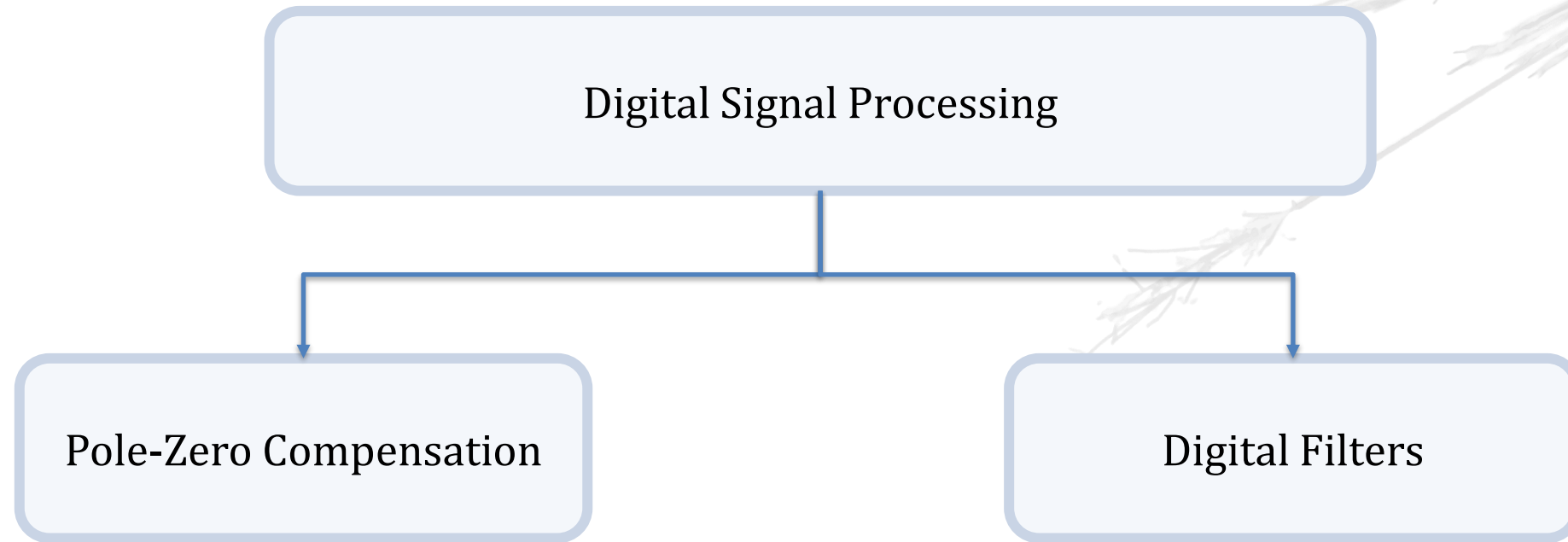


NOISE

$$Q \propto V_{max}$$



PILE-UP



Jordanov's Trapezoidal Filter (1994)

- Combines discrete-time Pole-Zero cancellation and band-pass shaping into a single, real-time recursive algorithm
- Provides efficient filtering for white series, white parallel and flicker noise
- Implements a lightweight recursive equation
- Usually adopted as the de-facto baseline for modern digital gamma-ray spectroscopy



Nuclear Instruments and Methods in Physics Research A 353 (1994) 261–264



Digital techniques for real-time pulse shaping in radiation measurements

Valentin T. Jordanov ^{a,*}, Glenn F. Knoll ^b, Alan C. Huber ^a, John A. Pantazis ^a

^a Amptek Inc., 6 De Angelo Dr., Bedford, MA 01730, USA

^b The University of Michigan, Department of Nuclear Engineering, Ann Arbor, MI 48109, USA

Abstract

Recursive algorithms for real-time digital pulse shaping in pulse height measurements have been developed. The differentiated signal from the preamplifier (exponential pulse) is amplified and then digitized. Digital data are deconvolved so that the response of the high-pass network is eliminated. The deconvolved pulse is processed by a time-invariant digital filter which allows trapezoidal/triangular or cusp-like shapes to be synthesized. A prototype of a digital trapezoidal processor was built which is capable of sampling and processing digital data in real time at clock rates up to 50 MHz.

1. Introduction

In our previous work [1] we described recursive algorithms for real time pulse processing in high resolution spectroscopy. In that paper we also proposed a hardware configuration for a trapezoidal/triangular pulse shaper. Although we presented some initial results obtained using a quasi-real time system, our further work has now resulted in the assembly and testing of a prototype that operates in true real time.

In the discussion that follows, it is assumed that an exponential pulse is digitized. This signal can be obtained by CR differentiation of the signal from a reset type charge sensitive preamplifier or by differentiation with a pole-zero cancellation network of the signal from a resistive feedback preamplifier. This approach allows elimination of the dc offset of the preamplifier signal and sufficient amplification of the short exponential pulses so that maximum utilization of the resolution of the sampling ADC can be

second algorithm transforms an exponential or step pulse to a symmetrical pulse shape with the leading edge proportional to $t^2 + t$. We call this shape "cusp-like."

2. Trapezoidal digital pulse-shaper

The recursive algorithm [1] that converts a digitized exponential pulse $v(n)$ into a symmetrical trapezoidal pulse $s(n)$ is given as

$$d^{k,l}(n) = v(n) - v(n-k) - v(n-l) + v(n-k-l), \quad (1)$$

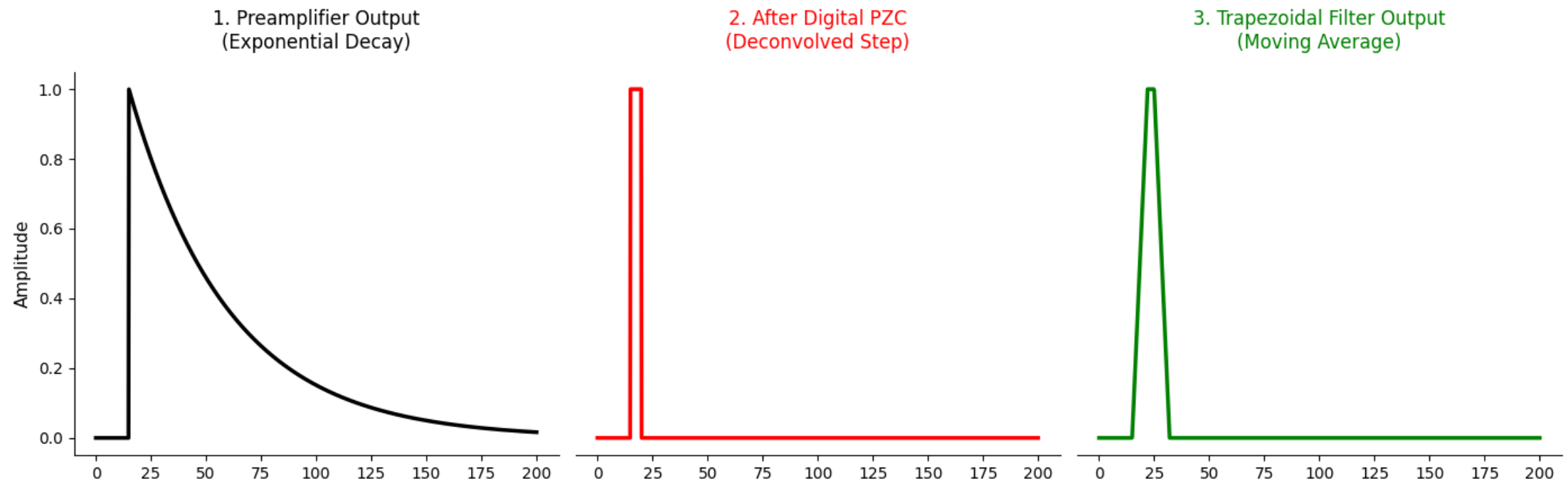
$$p(n) = p(n-1) + d^{k,l}(n), \quad n \geq 0, \quad (2)$$

$$r(n) = p(n) + Md^{k,l}(n), \quad (3)$$

$$s(n) = s(n-1) + r(n), \quad n \geq 0, \quad (4)$$

where $v(n)$, $p(n)$, and $s(n)$ are equal to zero for $n < 0$.

Algorithm Steps



$$\int_0^T e^{-\frac{t}{\tau}} dt = -\tau (e^{-\frac{t}{\tau}})_0^T = \tau - \tau e^{-\frac{T}{\tau}}$$

$$(\int_0^T e^{-\frac{t}{\tau}} dt) + \tau e^{-\frac{T}{\tau}} = \tau - \tau e^{-\frac{T}{\tau}} + \tau e^{-\frac{T}{\tau}} = \tau$$

Riemann Sum

$$\sum_{i=0}^N e^{-\frac{i \Delta t}{\tau}} \Delta t + \tau e^{-\frac{N}{\tau}} = \tau$$

$$\int_0^T e^{-\frac{t}{\tau}} dt = -\tau (e^{-\frac{t}{\tau}})_0^T = \tau - \tau e^{-\frac{T}{\tau}}$$

$$(\int_0^T e^{-\frac{t}{\tau}} dt) + \tau e^{-\frac{T}{\tau}} = \tau - \tau e^{-\frac{T}{\tau}} + \tau e^{-\frac{T}{\tau}} = \tau$$

Riemann Sum

$$\sum_{i=0}^N e^{-\frac{i \Delta t}{\tau}} \cancel{\Delta t} + \tau e^{-\frac{N}{\tau}} = \tau$$

$$\Delta t = \text{clock cycle} = 1$$

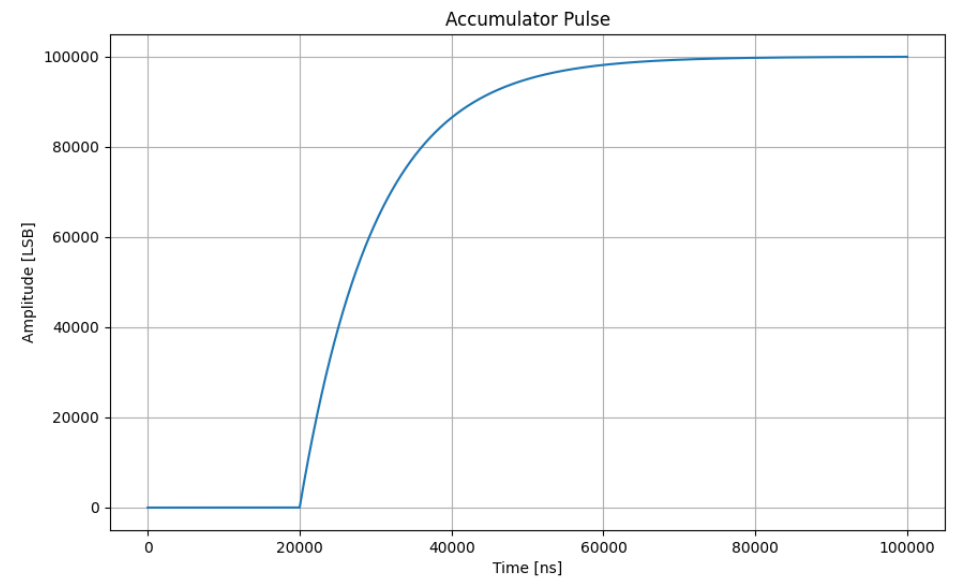
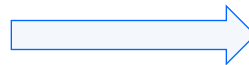
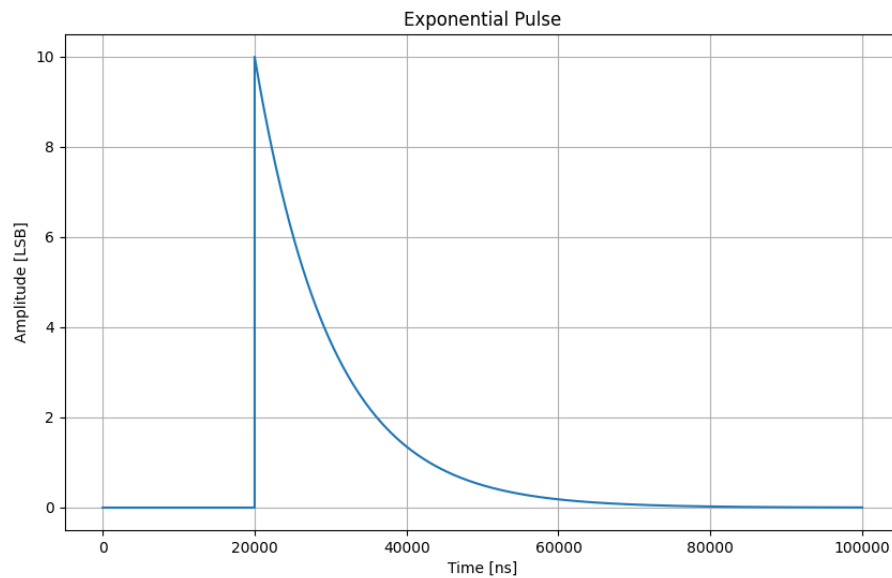
$$\int_0^T e^{-\frac{t}{\tau}} dt = -\tau (e^{-\frac{t}{\tau}})_0^T = \tau - \tau e^{-\frac{T}{\tau}}$$

$$(\int_0^T e^{-\frac{t}{\tau}} dt) + \tau e^{-\frac{T}{\tau}} = \tau - \tau e^{-\frac{T}{\tau}} + \tau e^{-\frac{T}{\tau}} = \tau$$

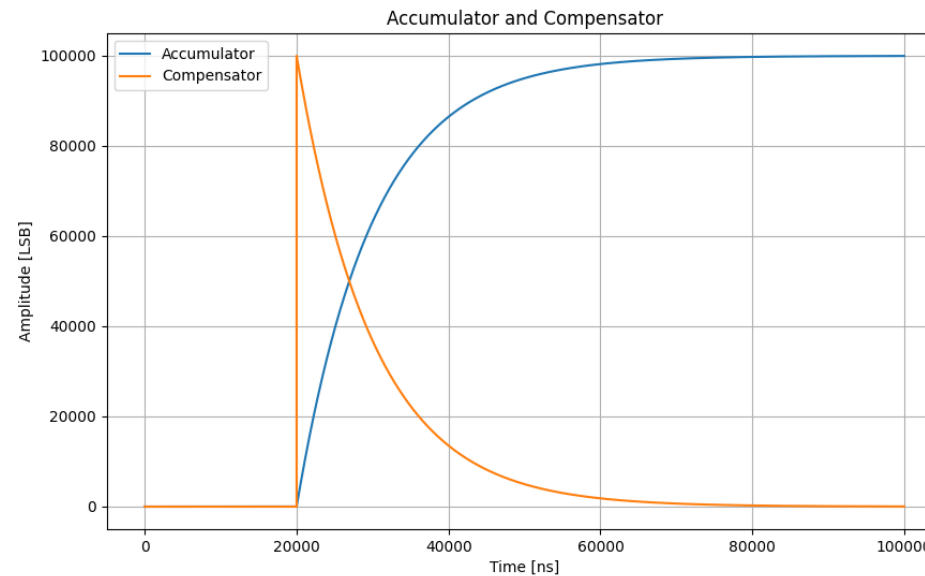
Riemann Sum

$$\sum_{i=0}^N e^{-\frac{i}{\tau}} + \tau e^{-\frac{N}{\tau}} = \tau$$

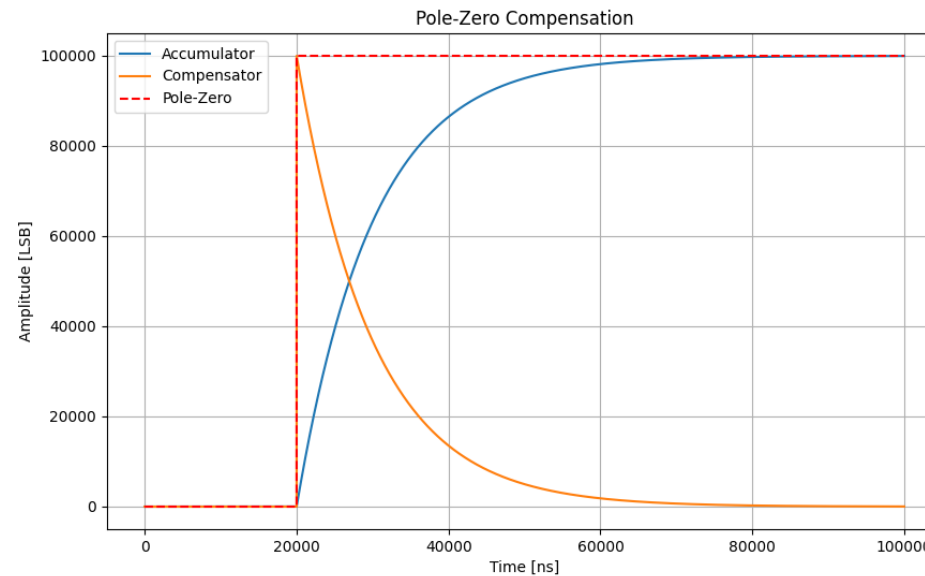
$$\sum_{i=0}^N e^{-\frac{i}{\tau}} + \tau e^{-\frac{N}{\tau}} = \tau$$



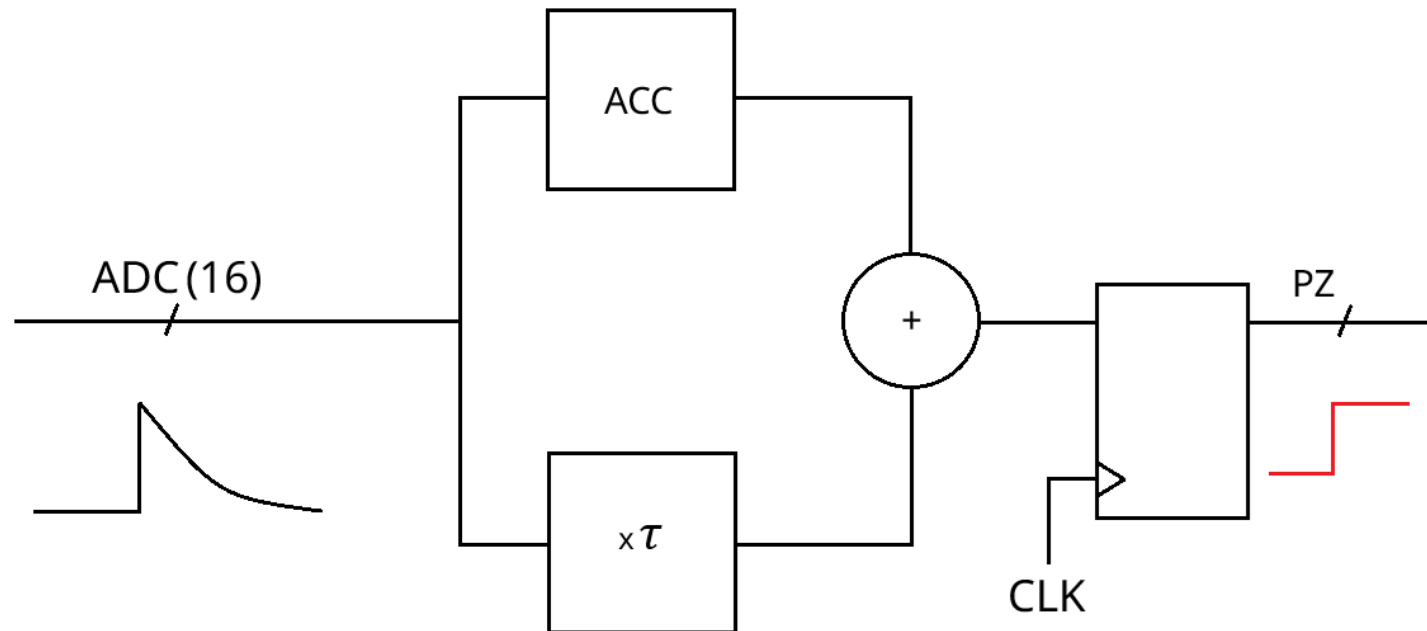
$$\sum_{i=0}^N e^{-\frac{i}{\tau}} + \tau e^{-\frac{N}{\tau}} = \tau$$



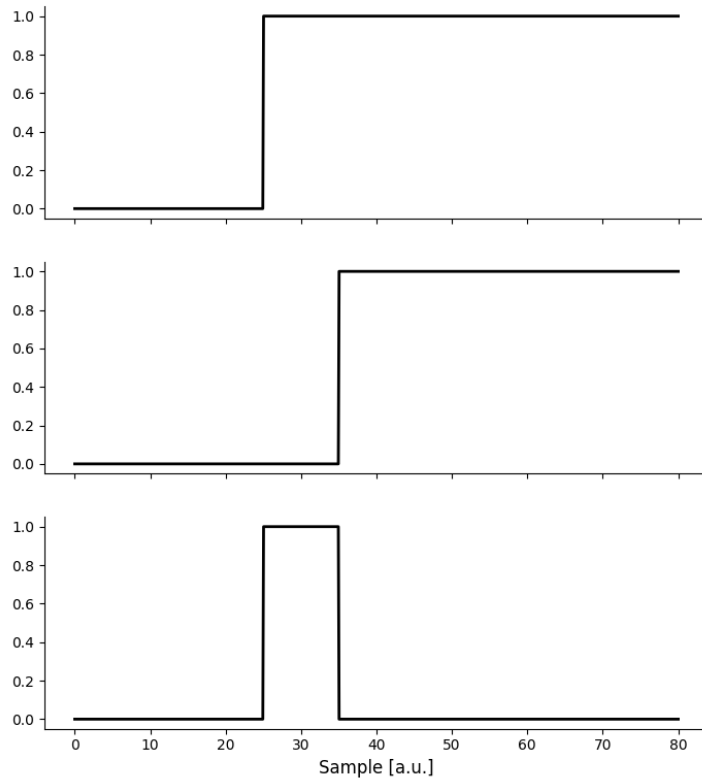
$$\sum_{i=0}^N e^{-\frac{i}{\tau}} + \tau e^{-\frac{N}{\tau}} = \tau$$



$$\sum_{i=0}^N e^{-\frac{i}{\tau}} + \tau e^{-\frac{N}{\tau}} = \tau$$



$$d[n] = x[n] - x[n-r]$$

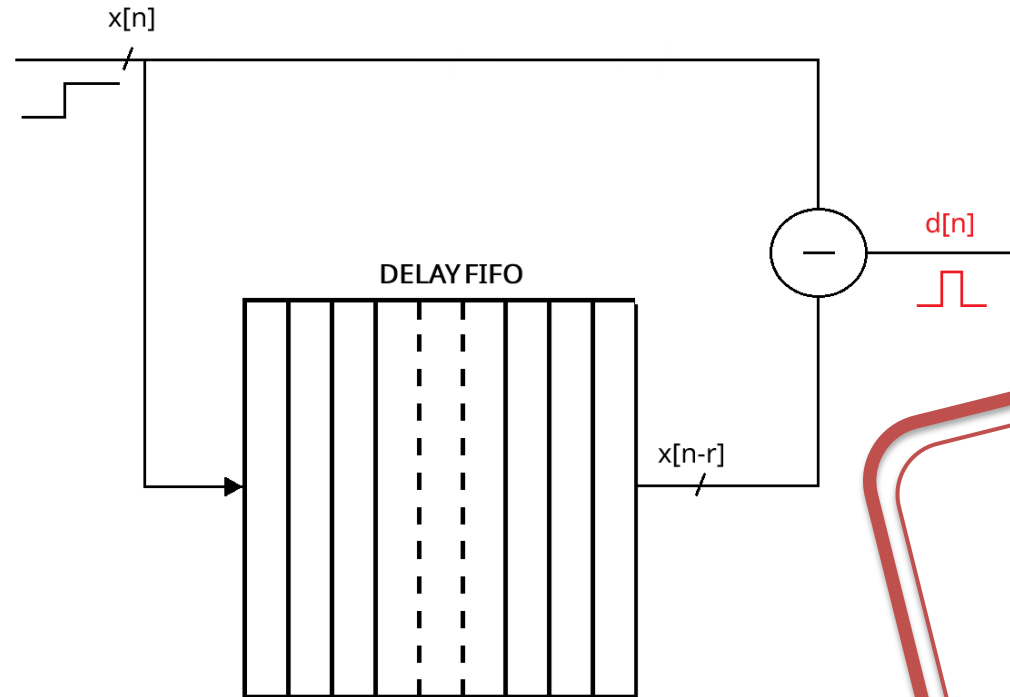


Step Function

Delayed Step Function

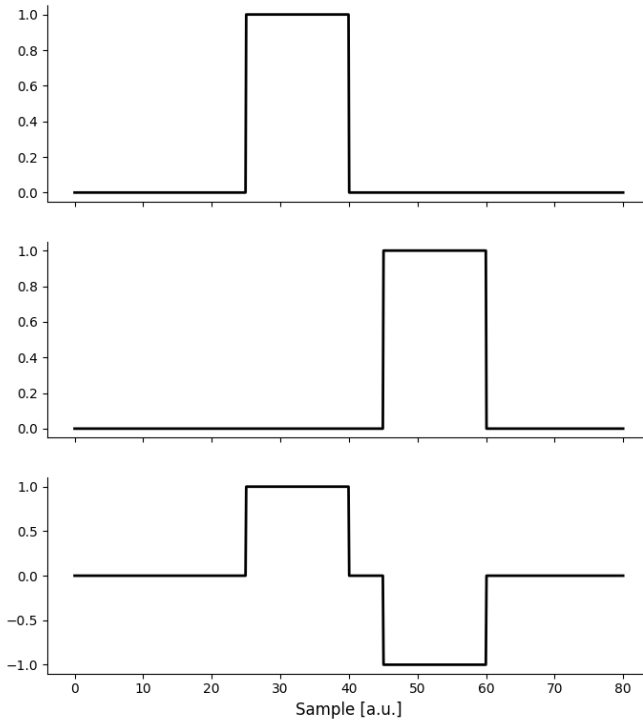
Rectangular Wave

$$d[n] = x[n] - x[n-r]$$



High Pass Filter

$$p[n] = d[n] - d[n-m]$$

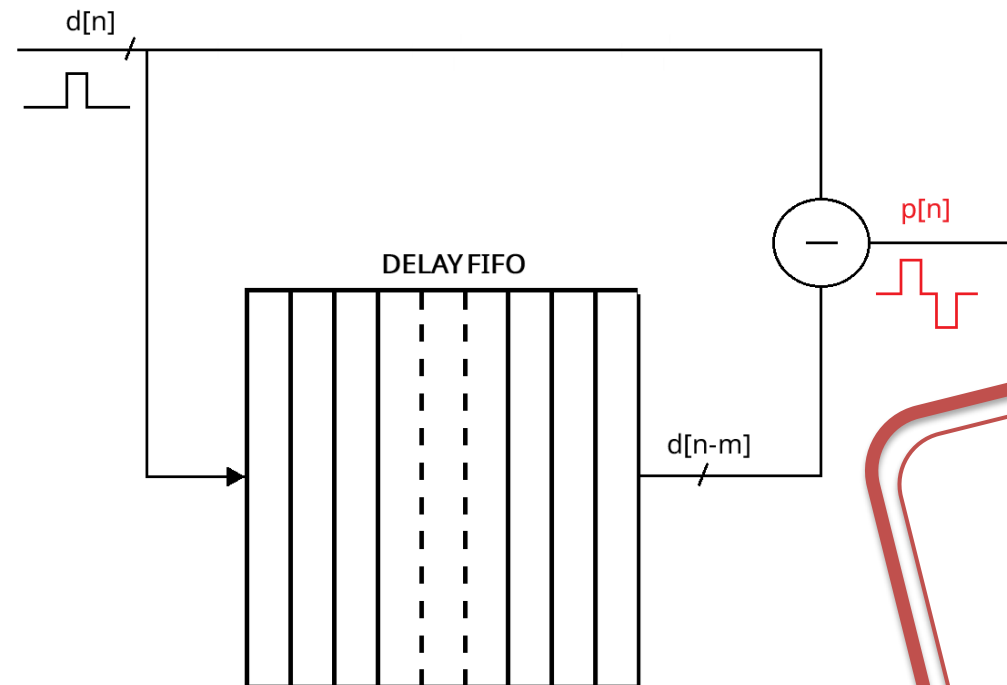


Rectangular Pulse

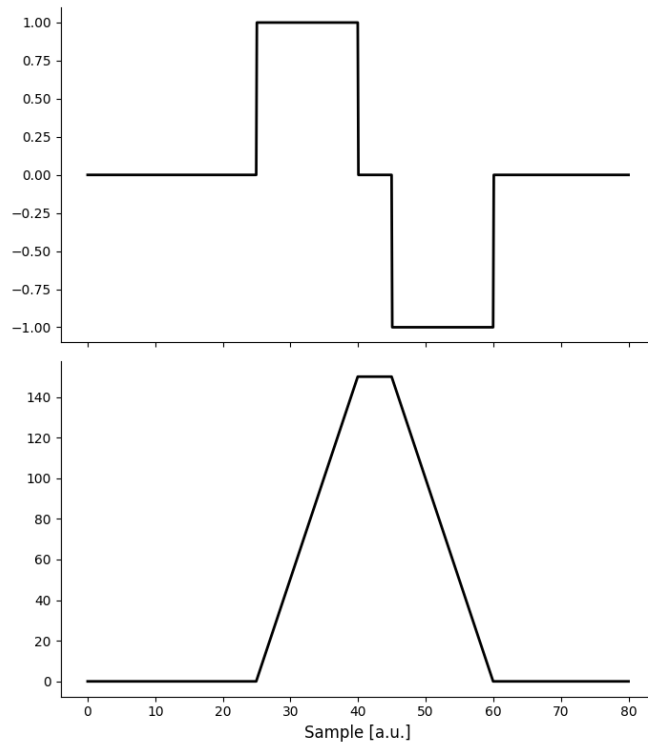
Delayed Rectangular Pulse

Bipolar Rectangular Pulse

$$p[n] = d[n] - d[n-m]$$



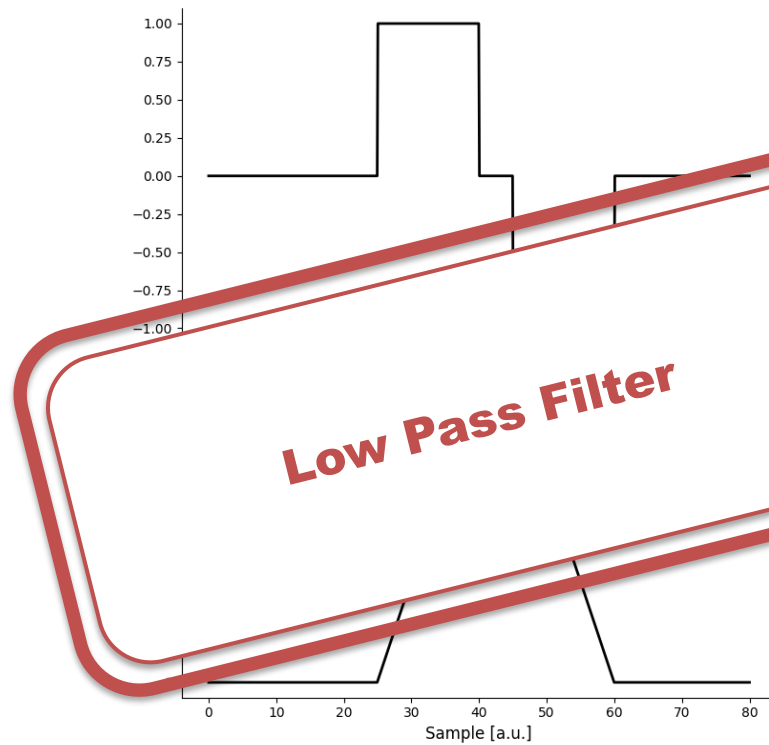
$$g[n] = g[n - 1] - d[n]$$



Bipolar Rectangular Pulse

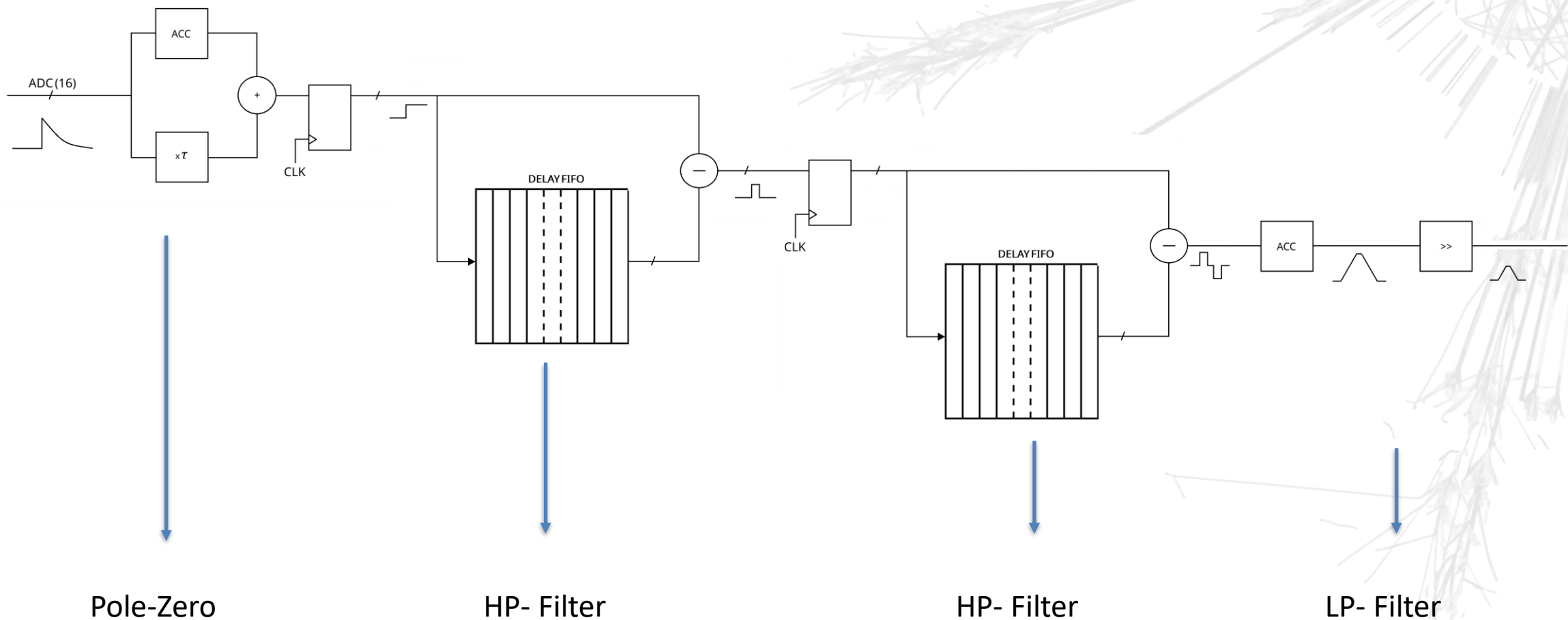
Trapezoid

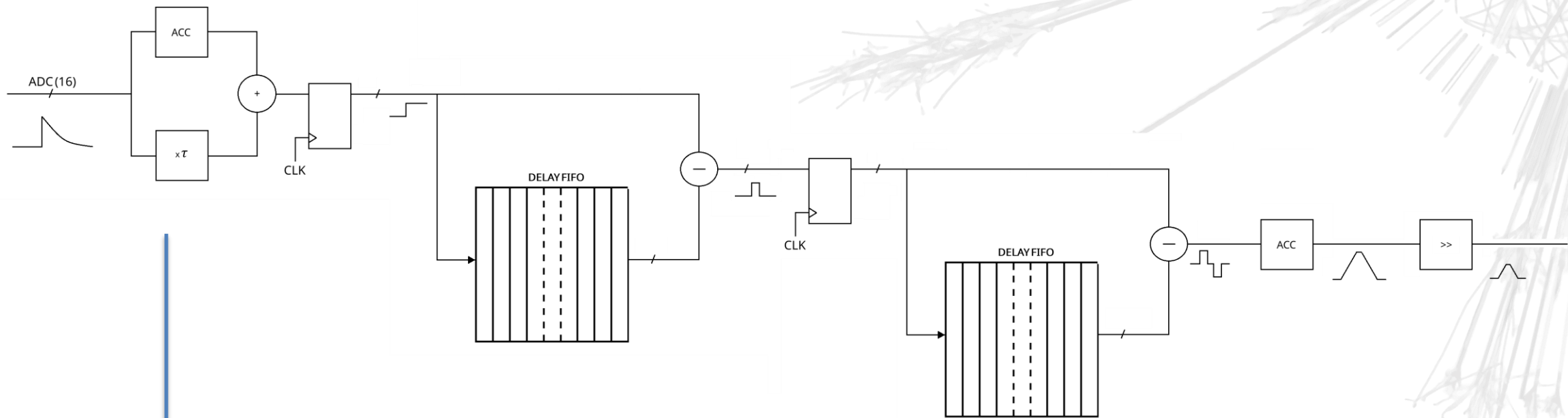
$$g[n] = g[n - 1] - d[n]$$



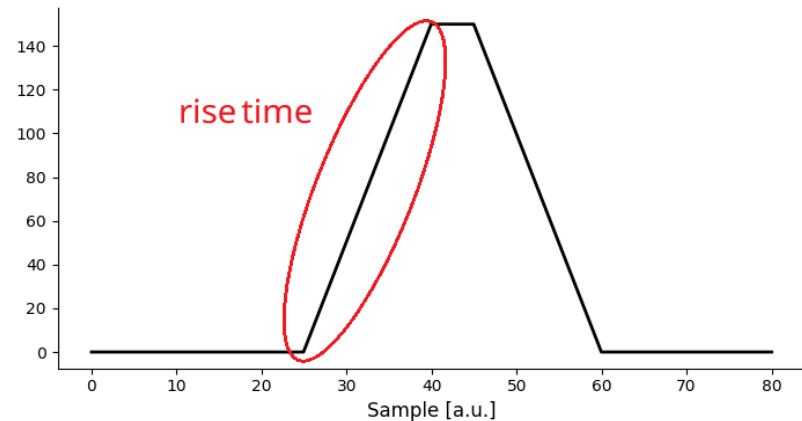
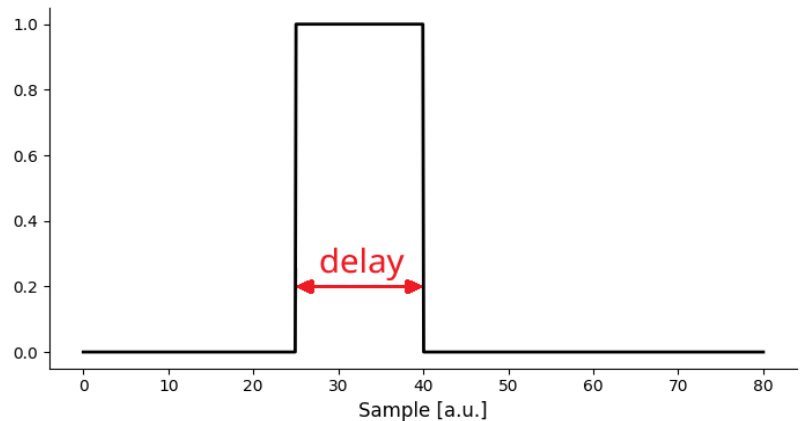
Bipolar Rectangular Pulse

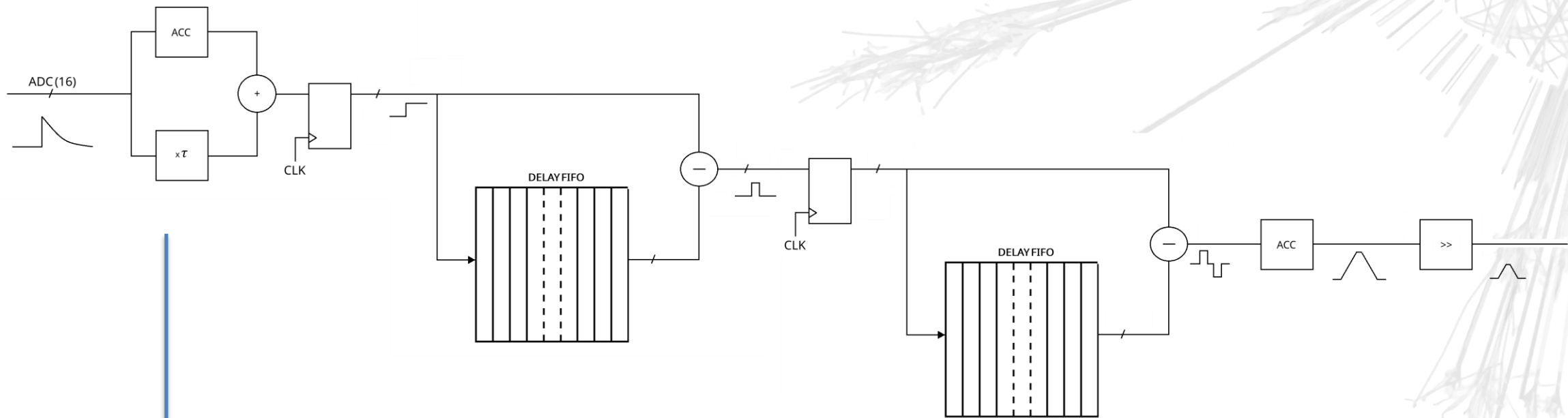
Trapezoid + Shift



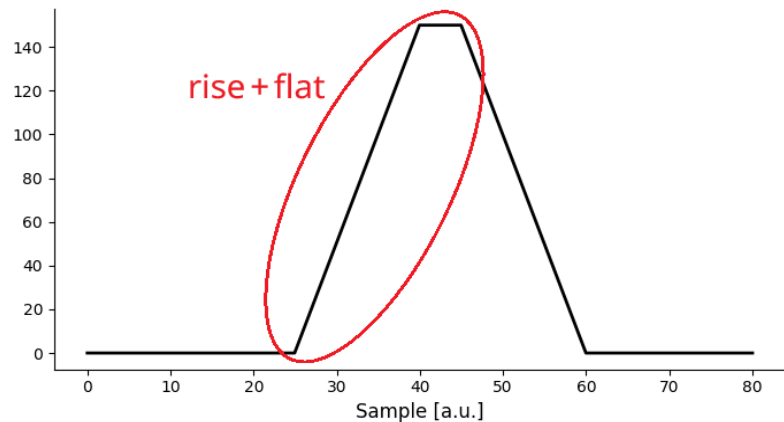
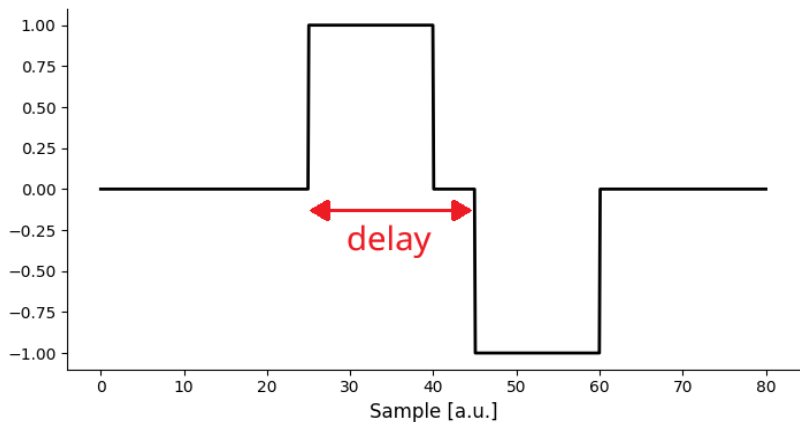


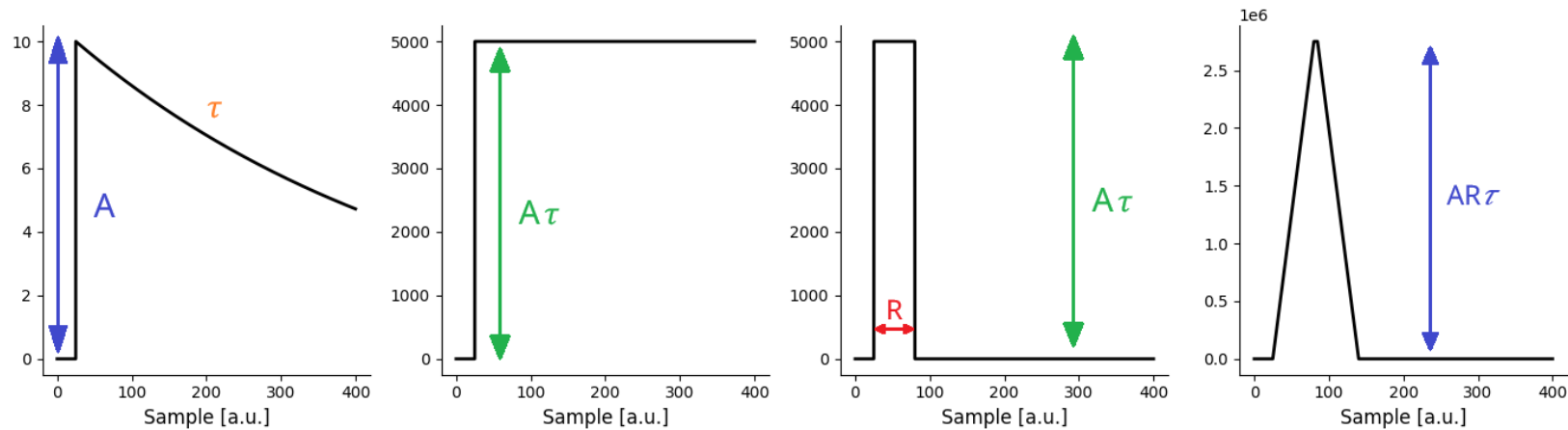
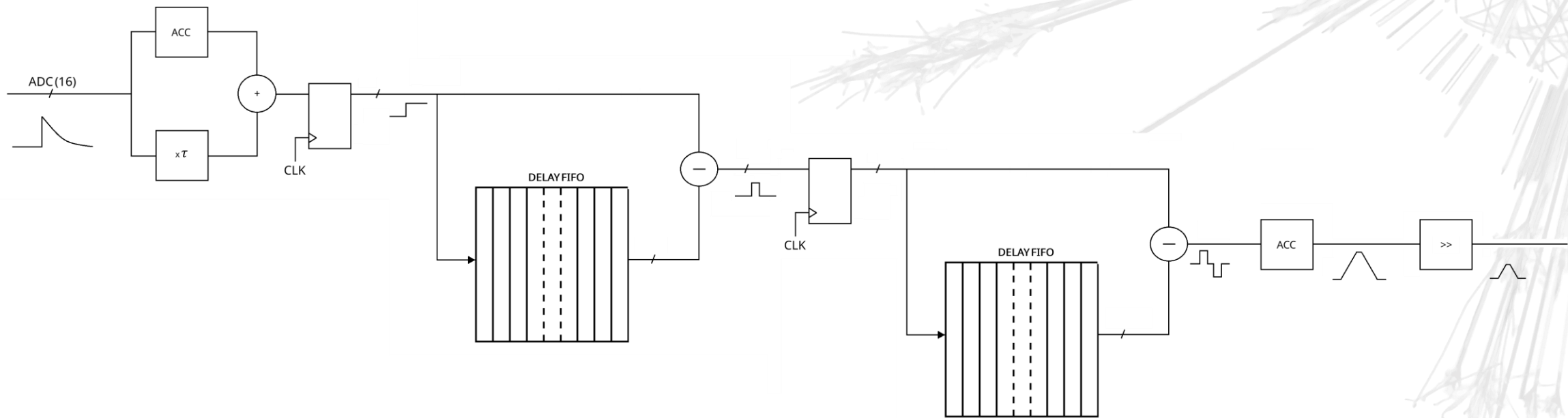
Pole-Zero

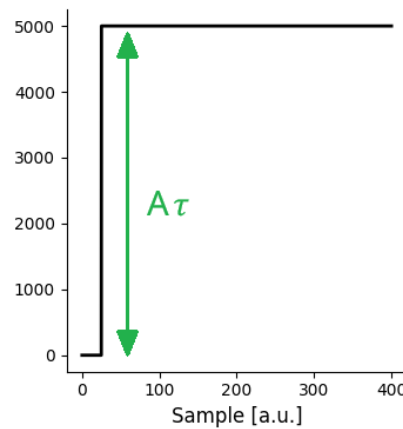
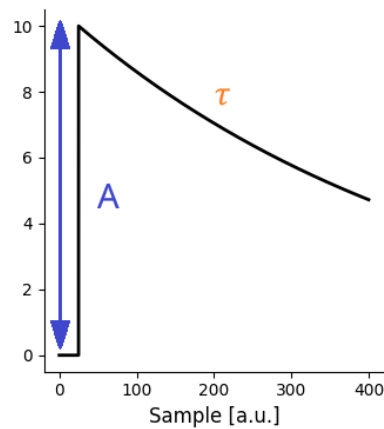
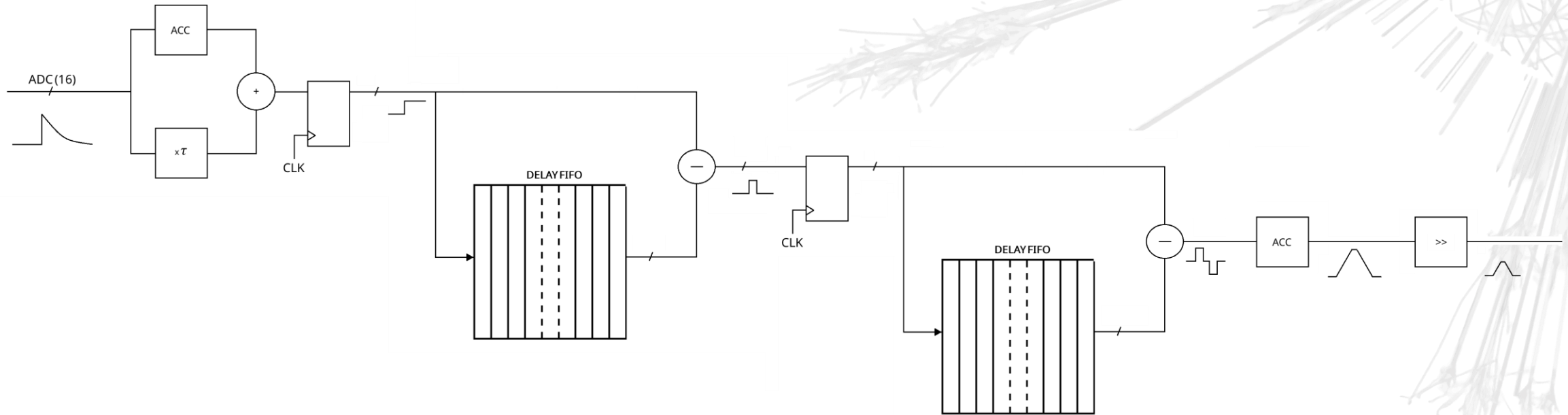




Pole-Zero





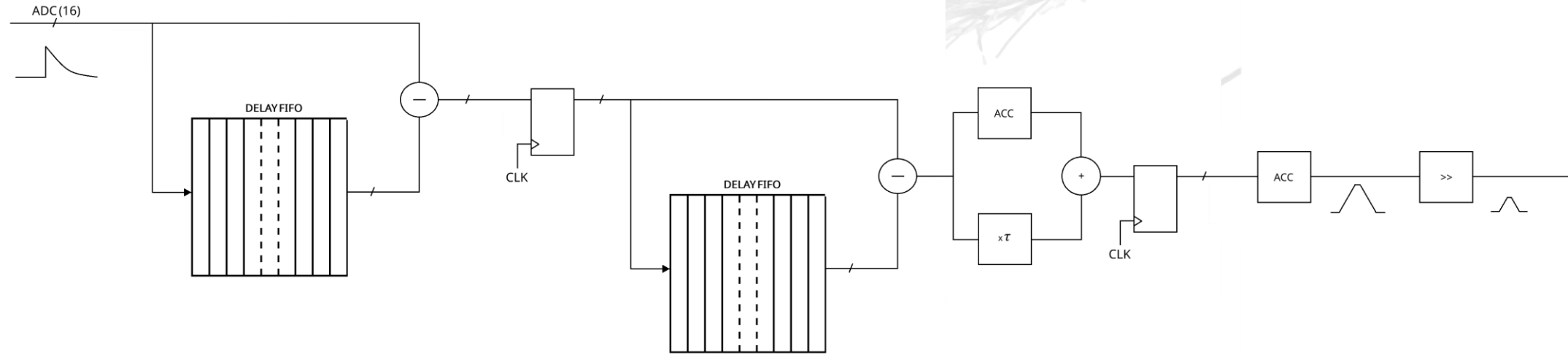


ADC = 16 bit

τ = 13 bit

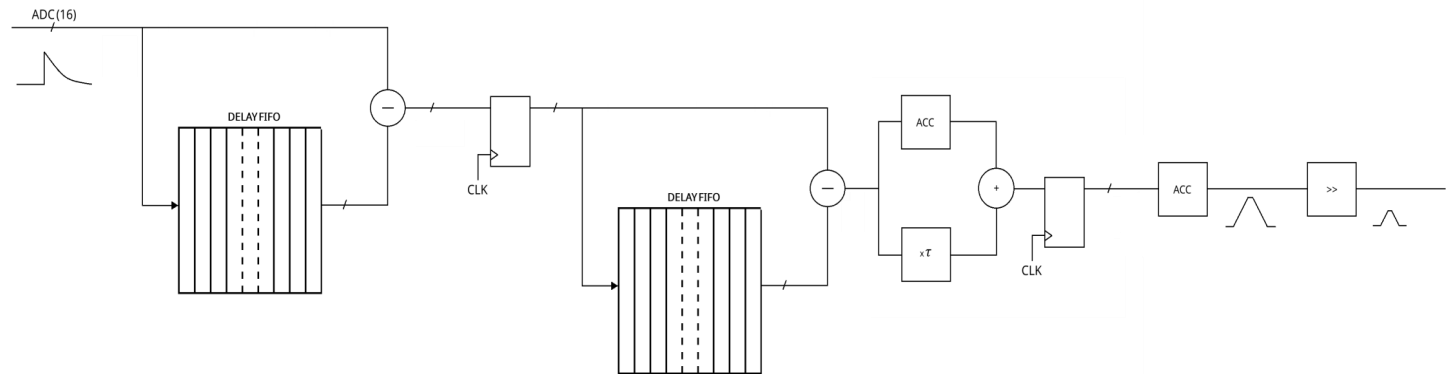


$A \times \tau = 29 \text{ bit}$

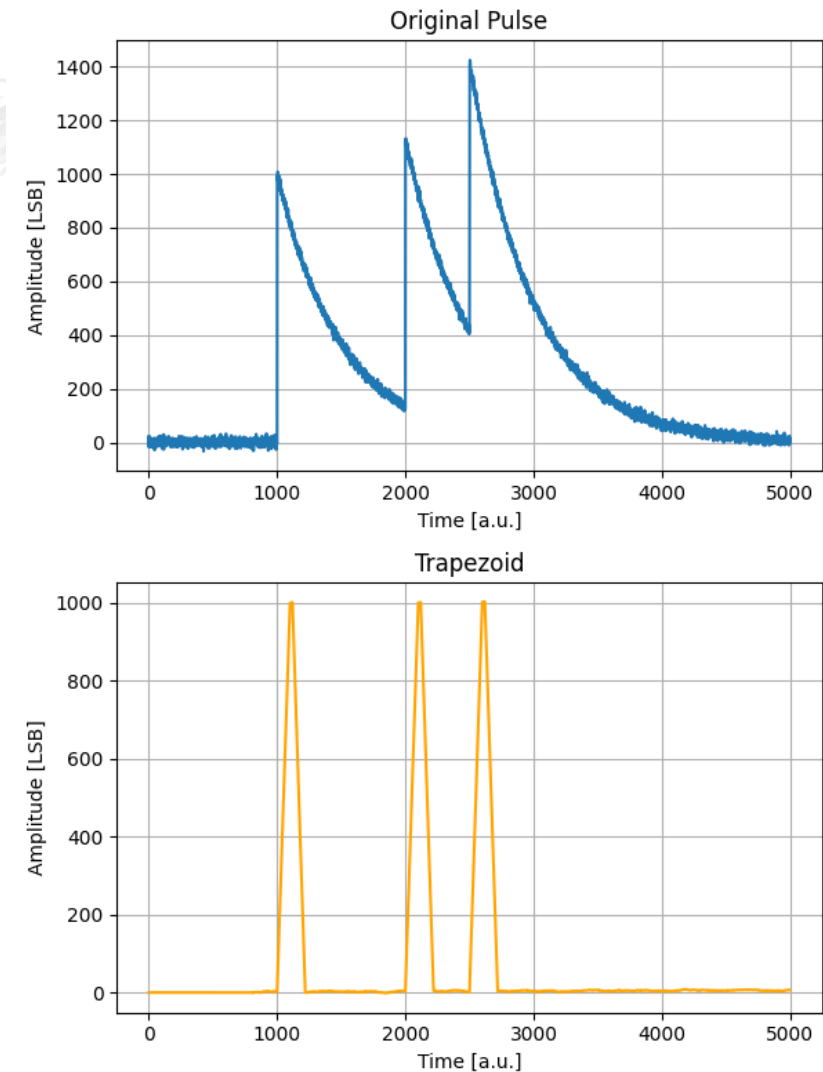


Pole-zero cancellation and double-differencing are both LTI operations :
cascaded LTI systems commute, so their order is interchangeable without affecting the output.

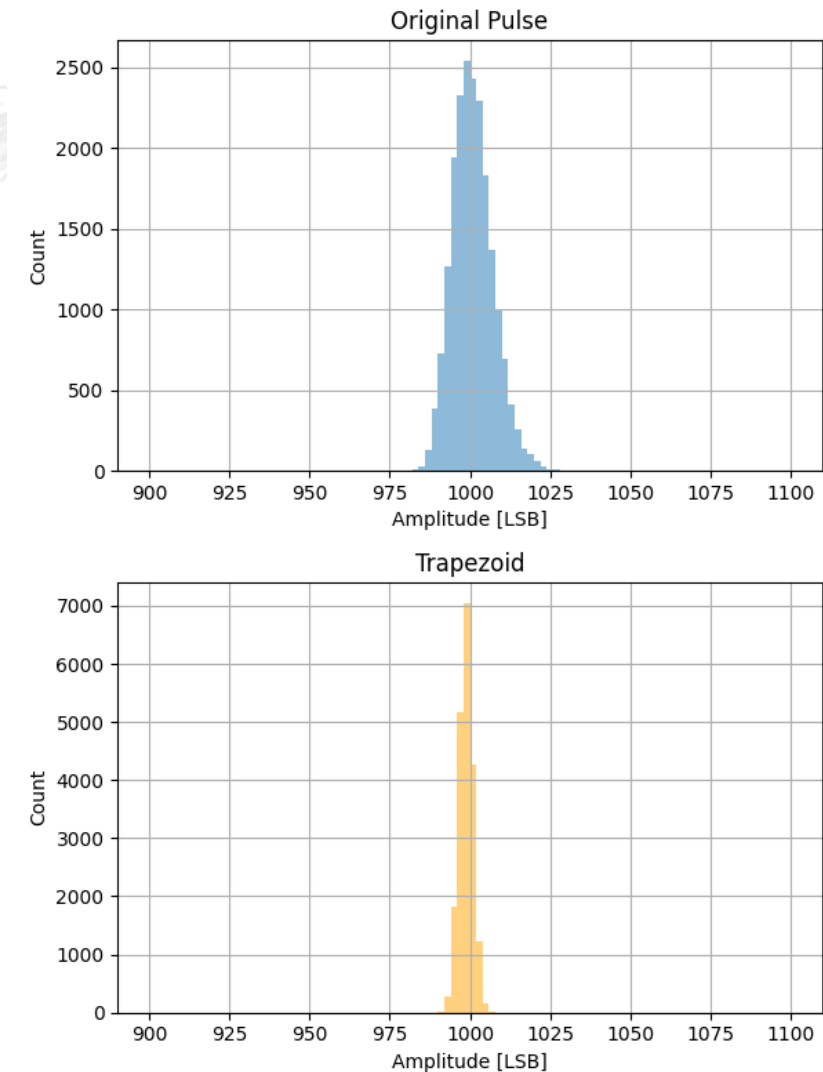
- Real Time Filter



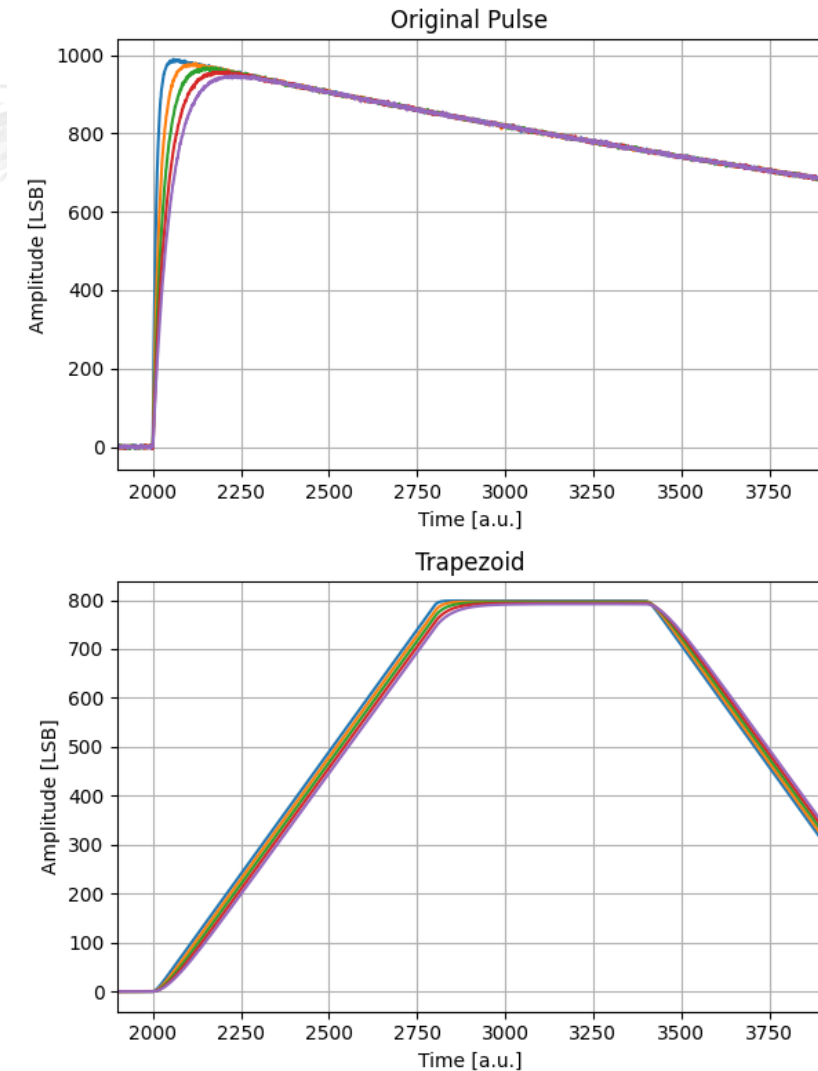
- Real Time Filter
- Improved Pile-up management due to the Pole-Zero Compensation



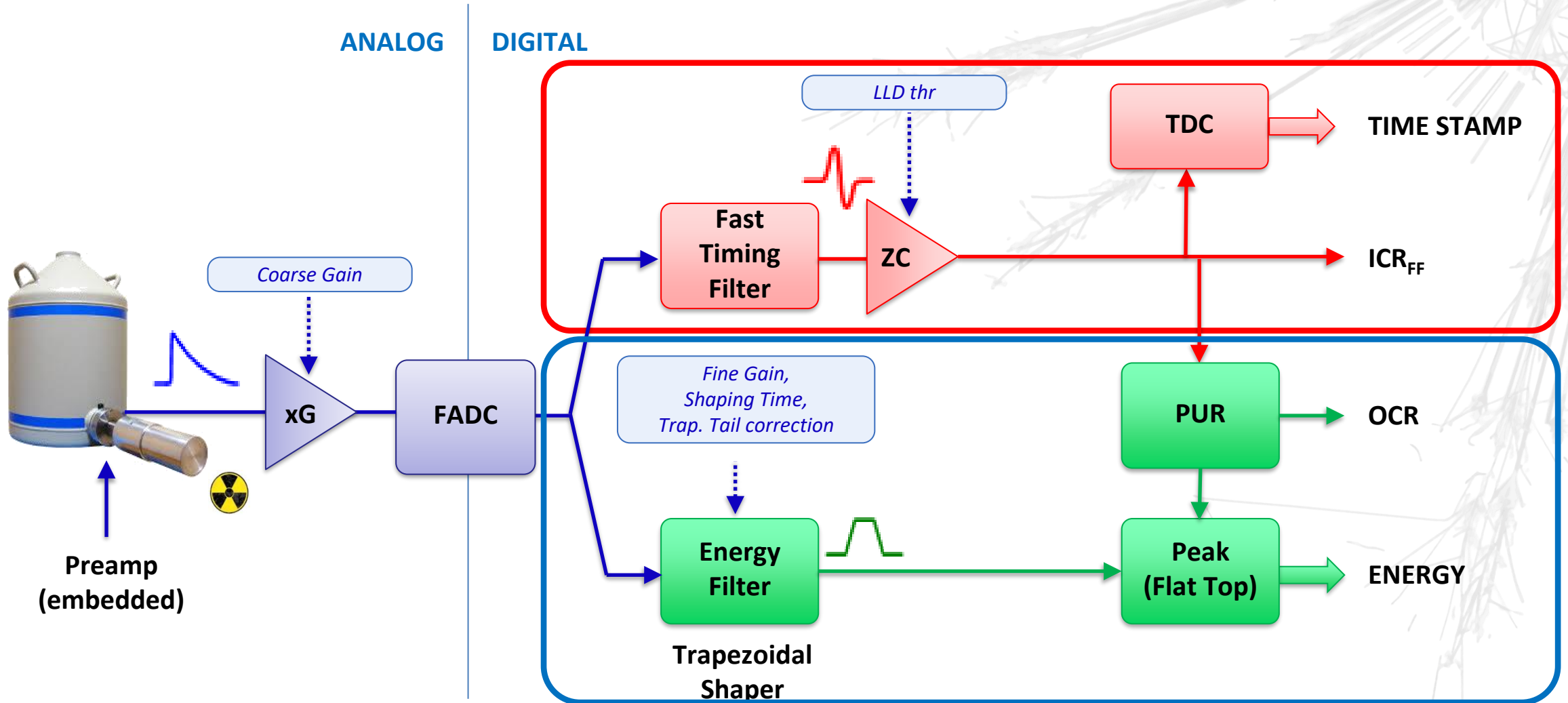
- Real Time Filter
- Improved Pile-up management due to the Pole-Zero Compensation
- Better Noise Performance due to the Pass-Band Filter

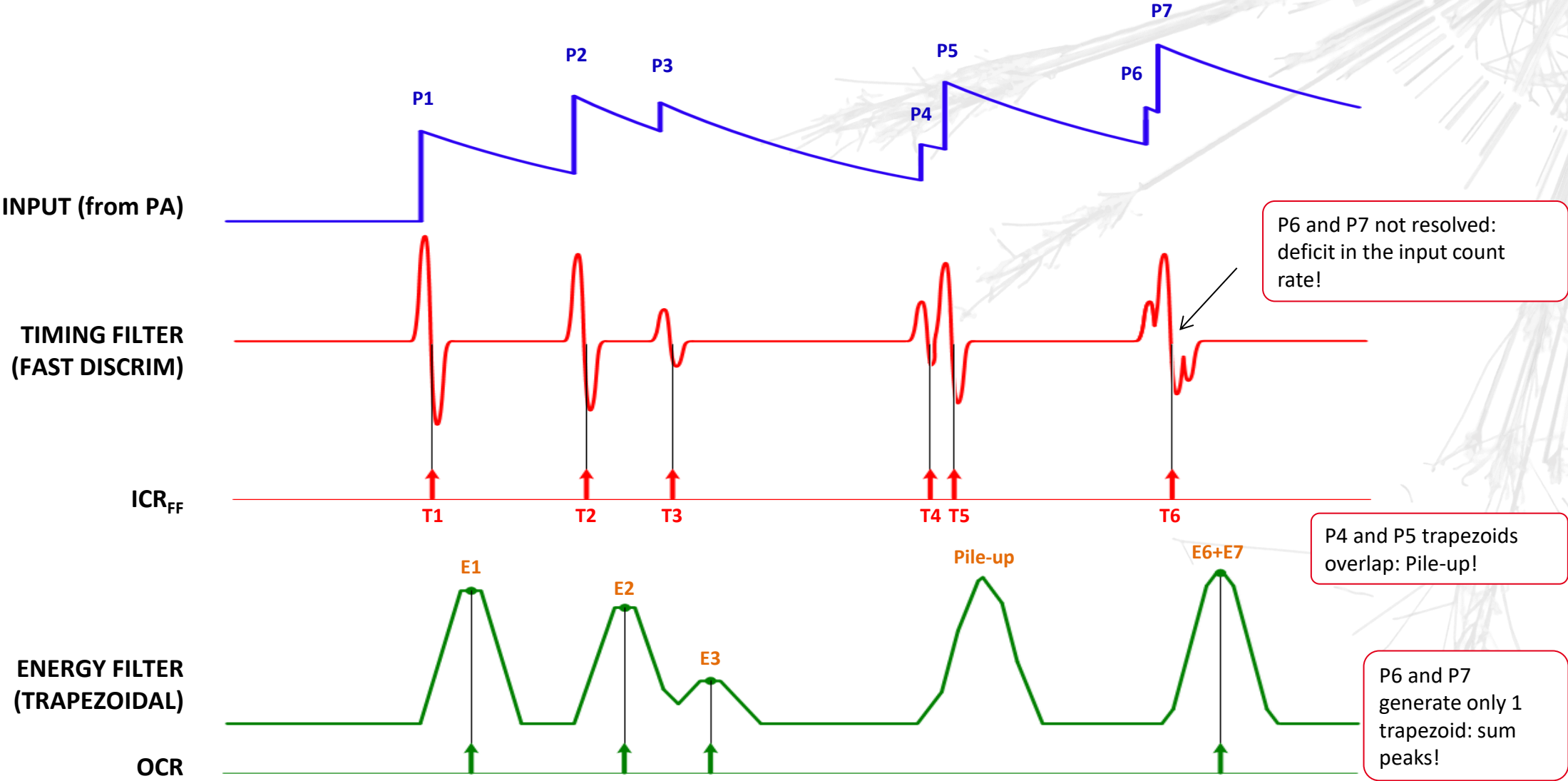


- Real Time Filter
- Improved Pile-up management due to the Pole-Zero Compensation
- Better Noise Performance due to the Pass-Band Filter
- Ballistic Deficit Mitigation due to the Flat-Top



The DPP-PHA Algorithms and Block Diagram

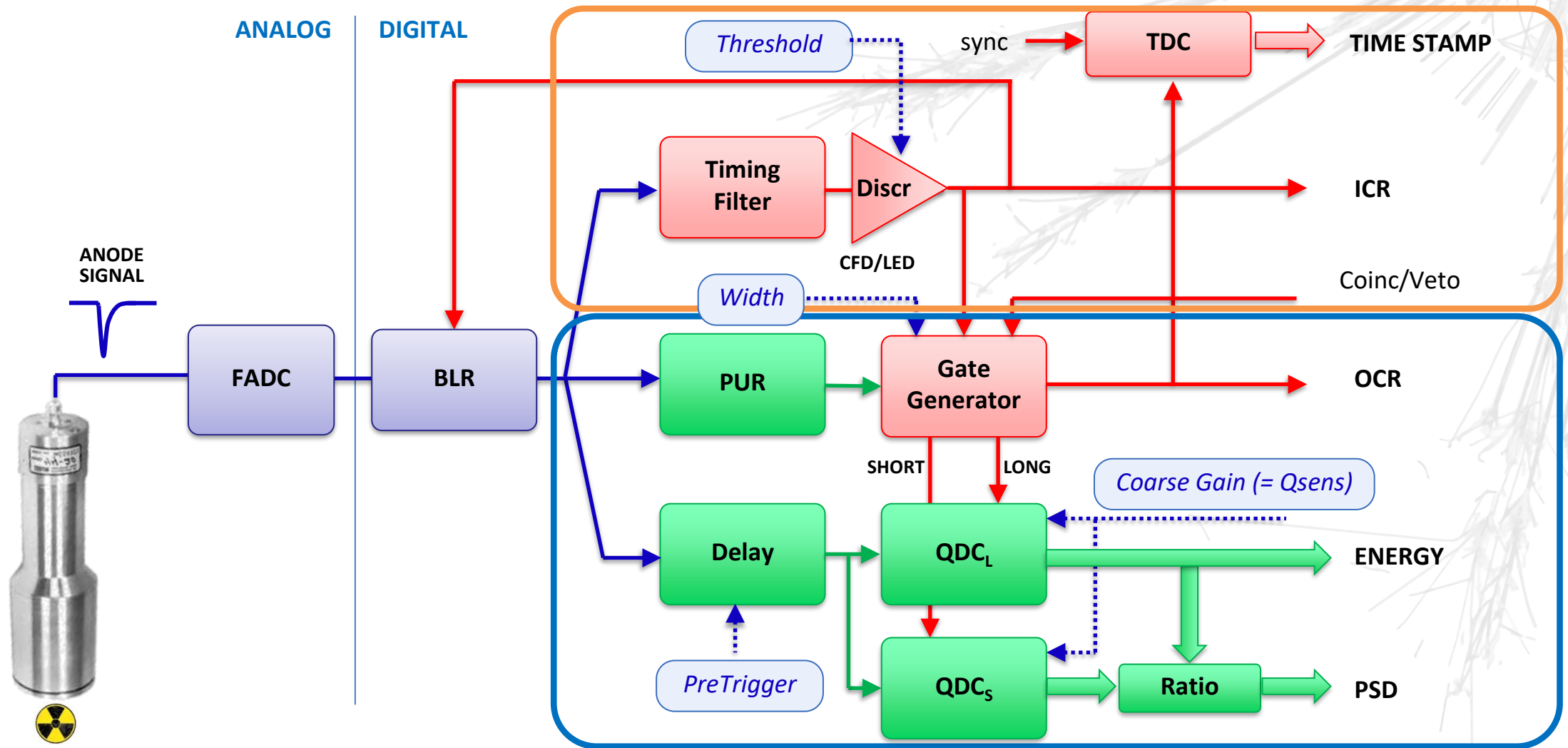




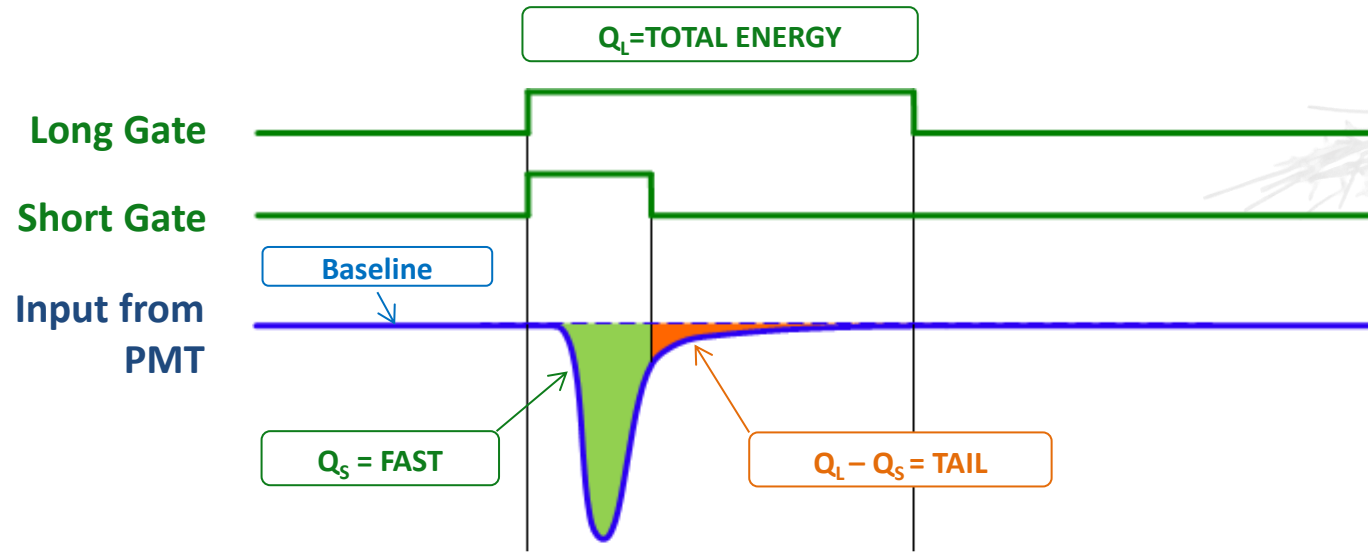
PSD

Pulse Shape Discrimination

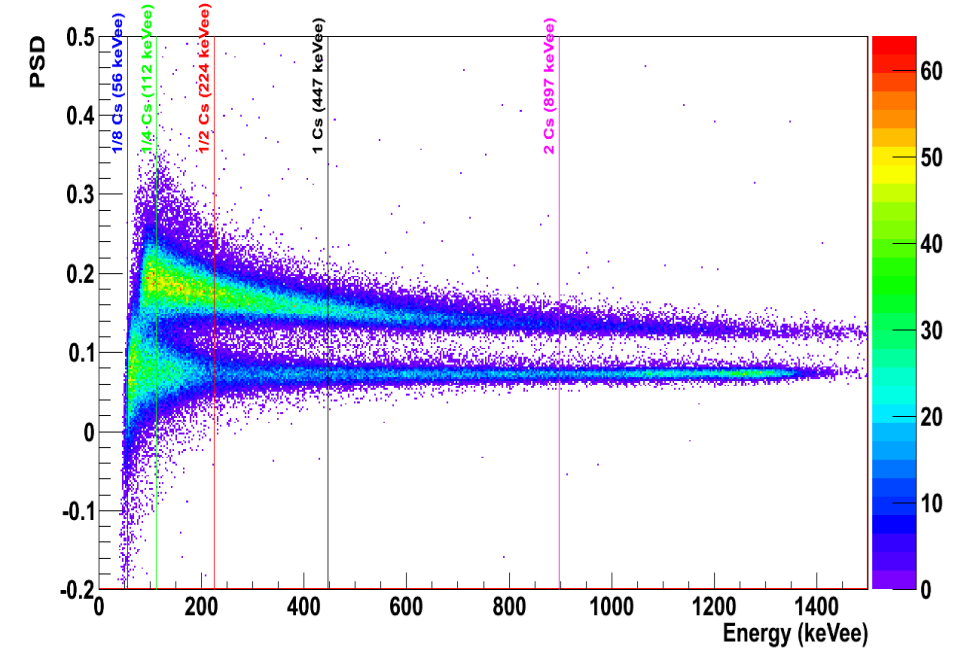
The DPP-PSD/QDC Algorithm and Block Diagram



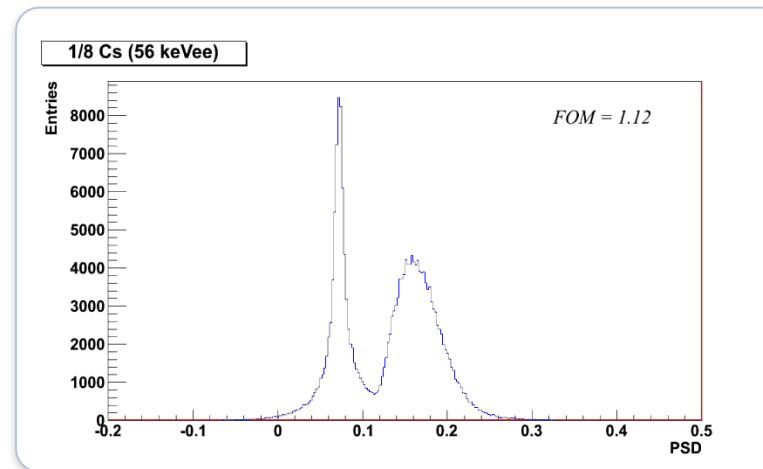
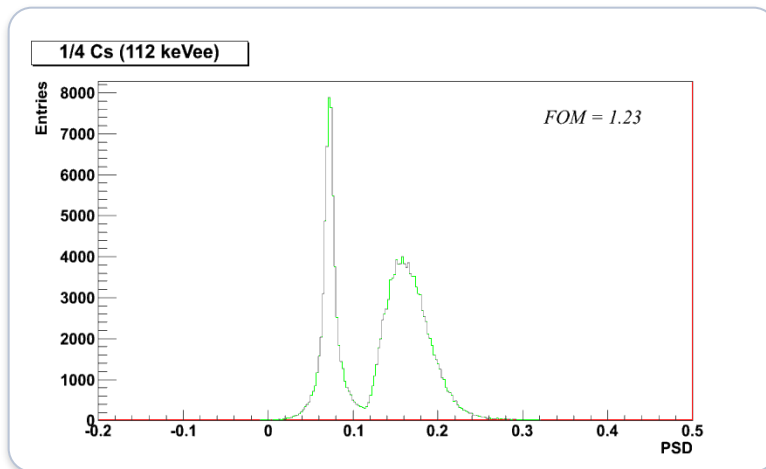
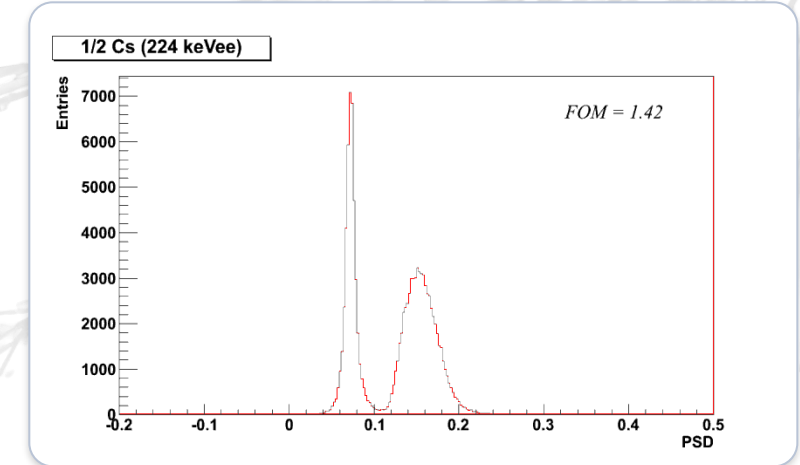
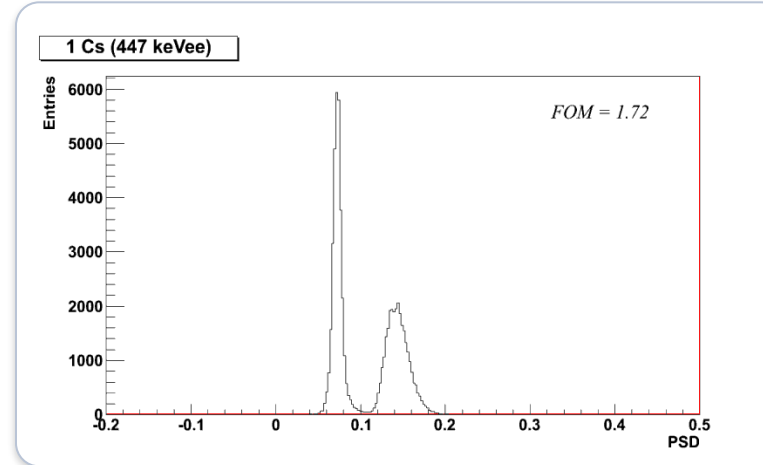
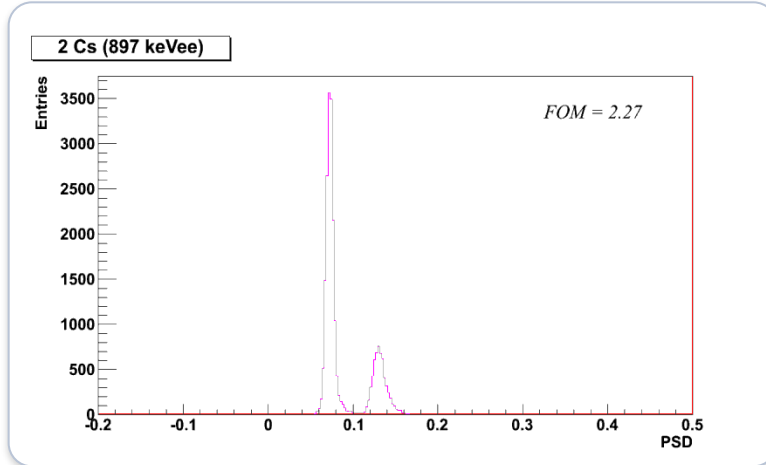
The DPP-PSD/QDC Signals



$$\text{PSD} = \frac{\text{TAIL}}{\text{TOTAL}}$$



The DPP-PSD/QDC Results



$$FOM = \frac{\Delta PEAK}{FWHM_{\gamma} + FWHM_n}$$



Thank you for your attention!

Alberto Potenza



a.potenza@caen.it

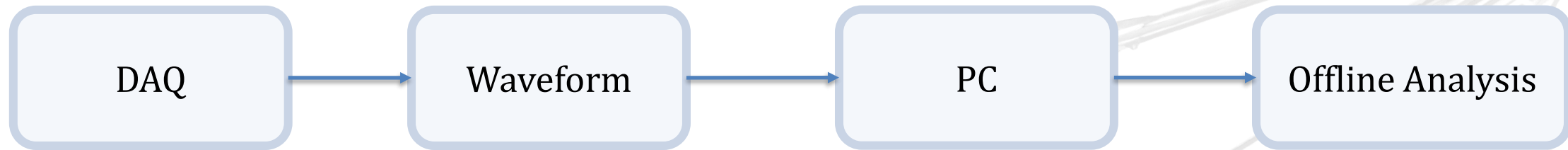


Reach us at educational@caen.it

Visit www.edu.caen.it

www.caen.it

Backup



Data Throughput Continuous Acquisition (e.g.)

- ADC = 16 bit
- Sampling Rate = 100 Msps
- N. Channels = 64

$$\begin{aligned} \text{Data Throughput Continuous Acquisition} \\ 2[\text{B}] * 100 [\text{Msps}] * 64 [\text{ch}] = \\ = 12 [\text{GB/s}] \end{aligned}$$

Data Throughput Triggered Acquisition (e.g.)

- ADC = 16 bit
- Sampling Rate = 100 Msps
- Trigger Rate = 5 kevents/s
- Waveform Stored = 100 μs = 10000 samples
- N. Channels = 64

$$\begin{aligned} \text{Data Throughput Triggered Acquisition} \\ 2[\text{B}] * 5000 [\text{events/s}] * 10000 [\text{sample/event}] * 64 [\text{ch}] = \\ = 6.4 [\text{GB/s}] \end{aligned}$$

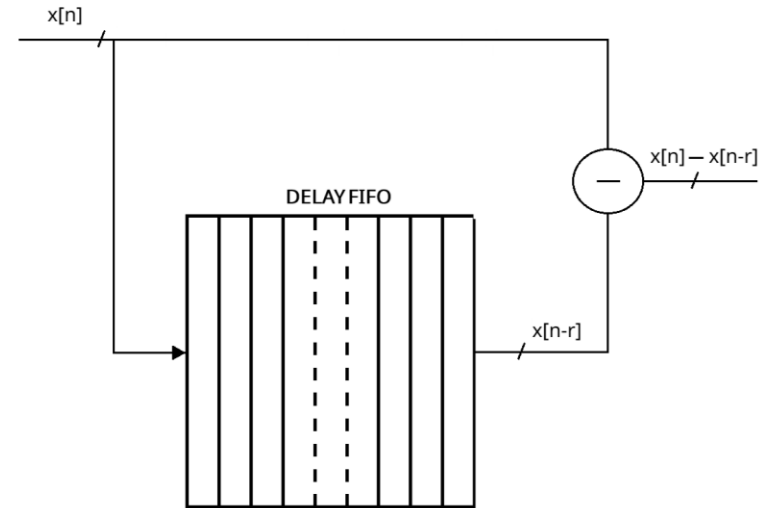
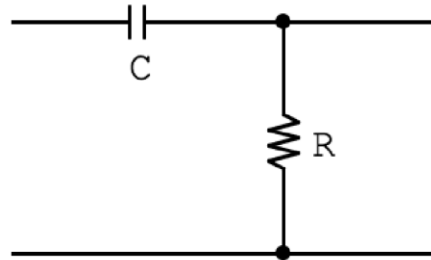


Data Throughput Online Analysis (e.g.)

- ADC = 16 bit
- Sampling Rate = 100 Msps
- N. Channels = 64
- Trigger Rate = 5 kevents/s
- Timestamp Word = 6B
- Energy Word = 2B
- Extra Word = 4B

Data Throughput Continuous Acquisition
 $(6[B] + 2[B] + 4[B]) * 5000 \text{ [events/s]} * 64 \text{ [ch]} =$
 $= 3.84 \text{ [MB/s]}$

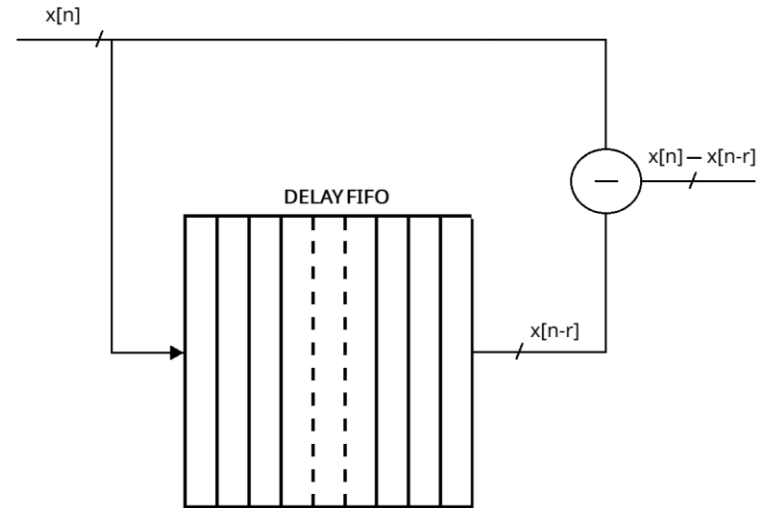
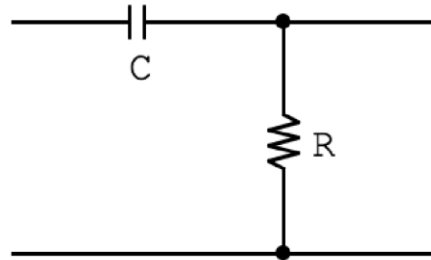
Huge Data Reduction!!



$x[n] = \text{low frequency component}[n] + \text{high frequency component}[n]$
 $x[n - r] = \text{low frequency component}[n - r] + \text{high frequency component}[n - r]$

$\text{low frequency component}[n] \approx \text{low frequency component}[n - r]$

$x[n] - x[n - r] = \text{high frequency component}[n] - \text{high frequency component}[n - r]$



$$\begin{aligned}x[n] &= \text{low frequency component}[n] + \text{high frequency component}[n] \\x[n-r] &= \text{low frequency component}[n-r] + \text{high frequency component}[n-r]\end{aligned}$$

$$\text{low frequency component}[n] \approx \text{low frequency component}[n-r]$$

$$x[n] - x[n-r] = \text{high frequency component}[n] - \text{high frequency component}[n-r]$$

Low frequency component does not change between sample n and sample n - r !!!!!

