

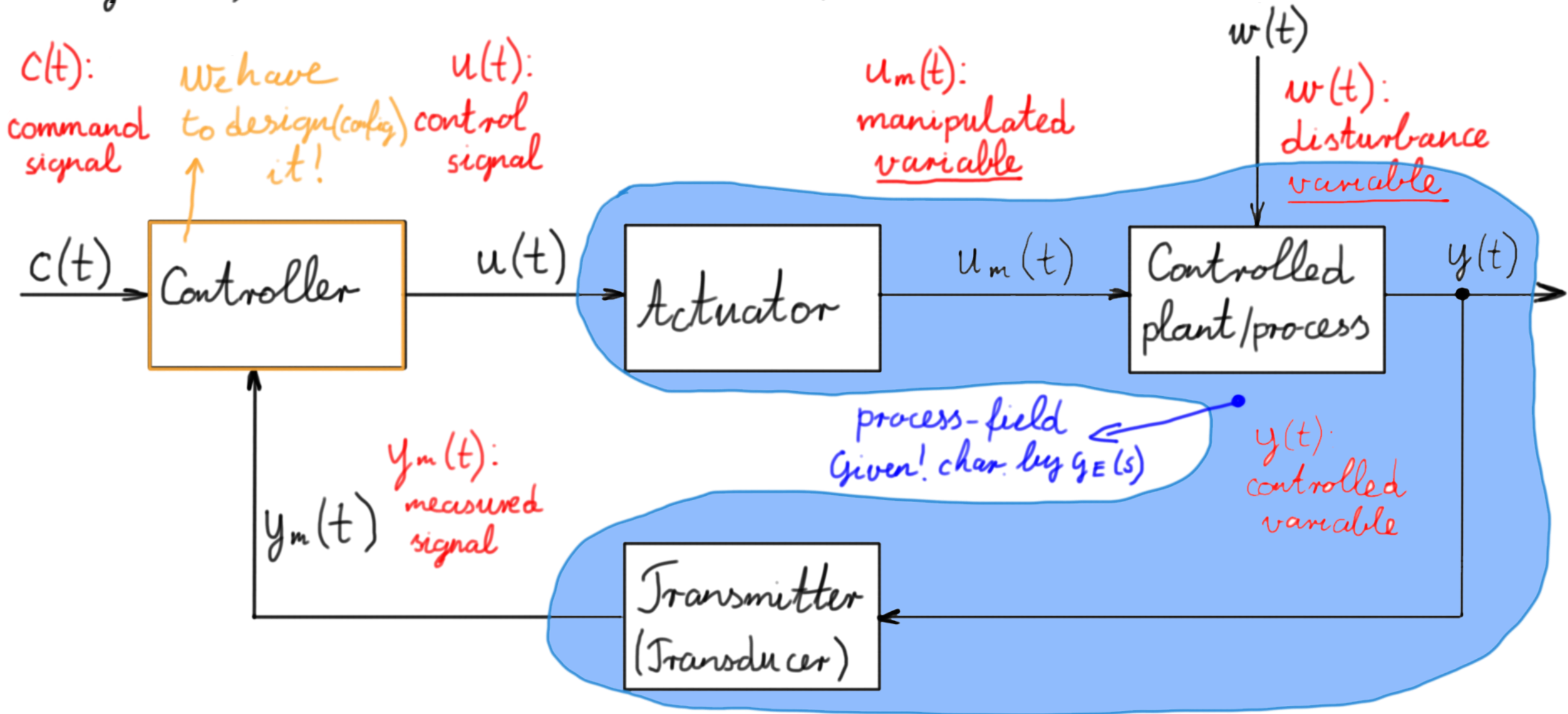
Programming a PID controller and a closed-loop control simulation using a PLC

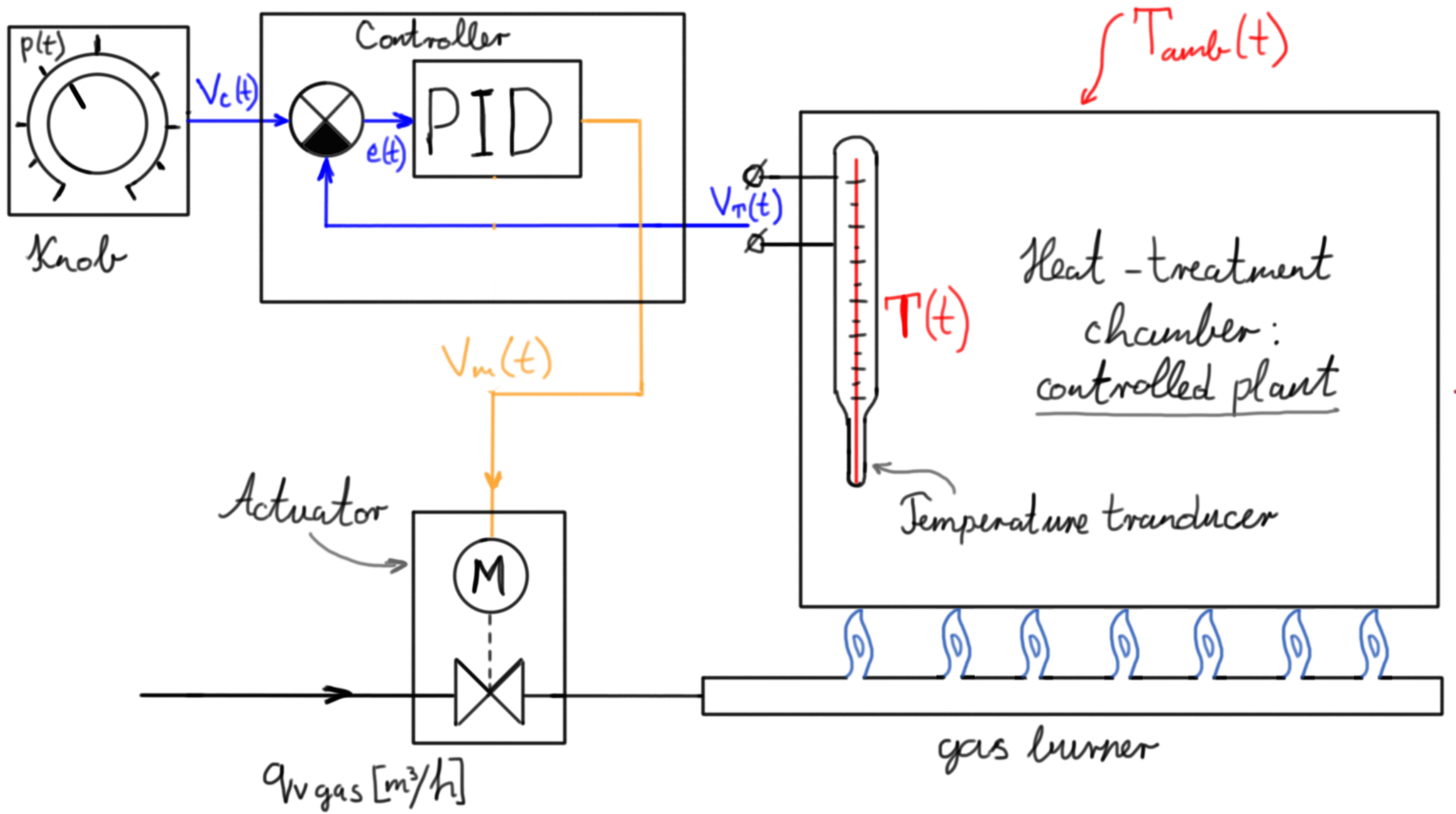
Mini course in Embedded Systems
2023.11.7-28.

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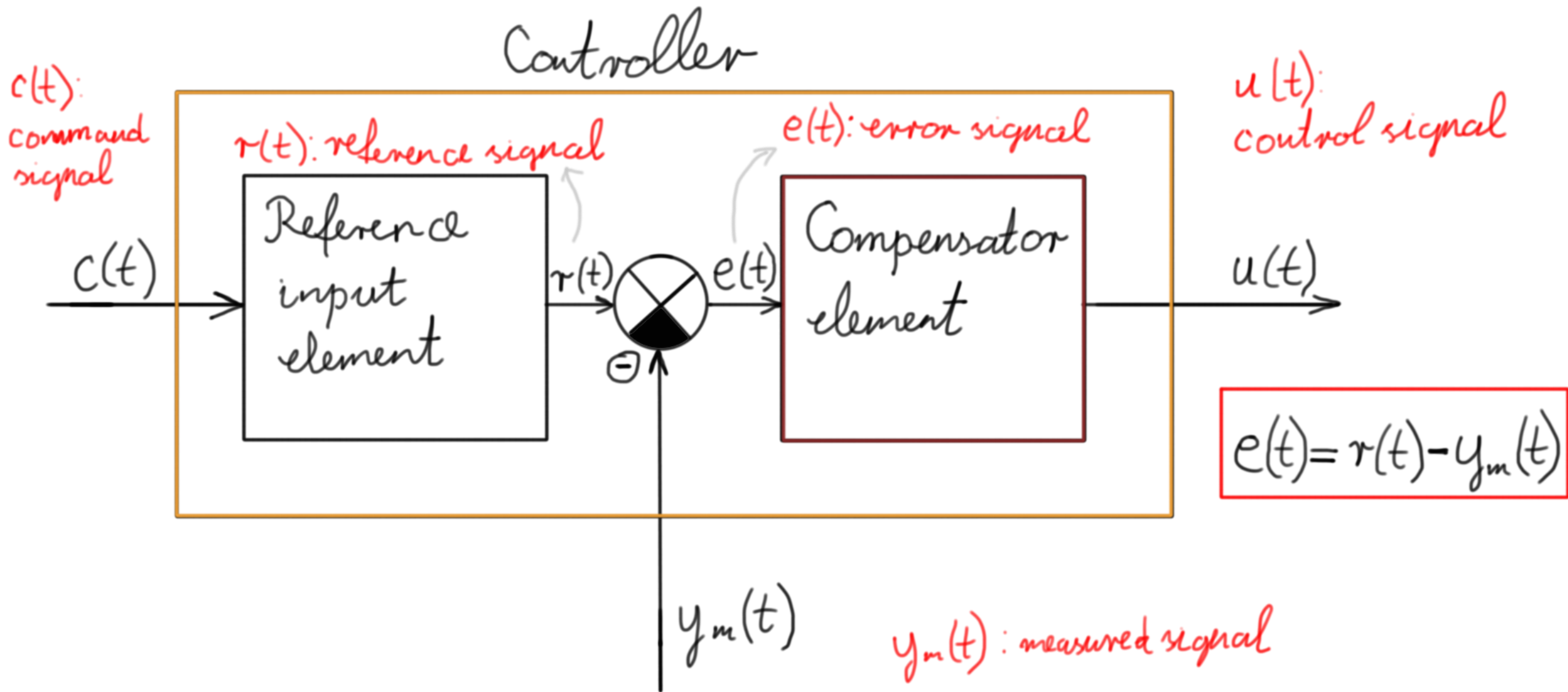


Signals, variables and elements of a closed-loop control

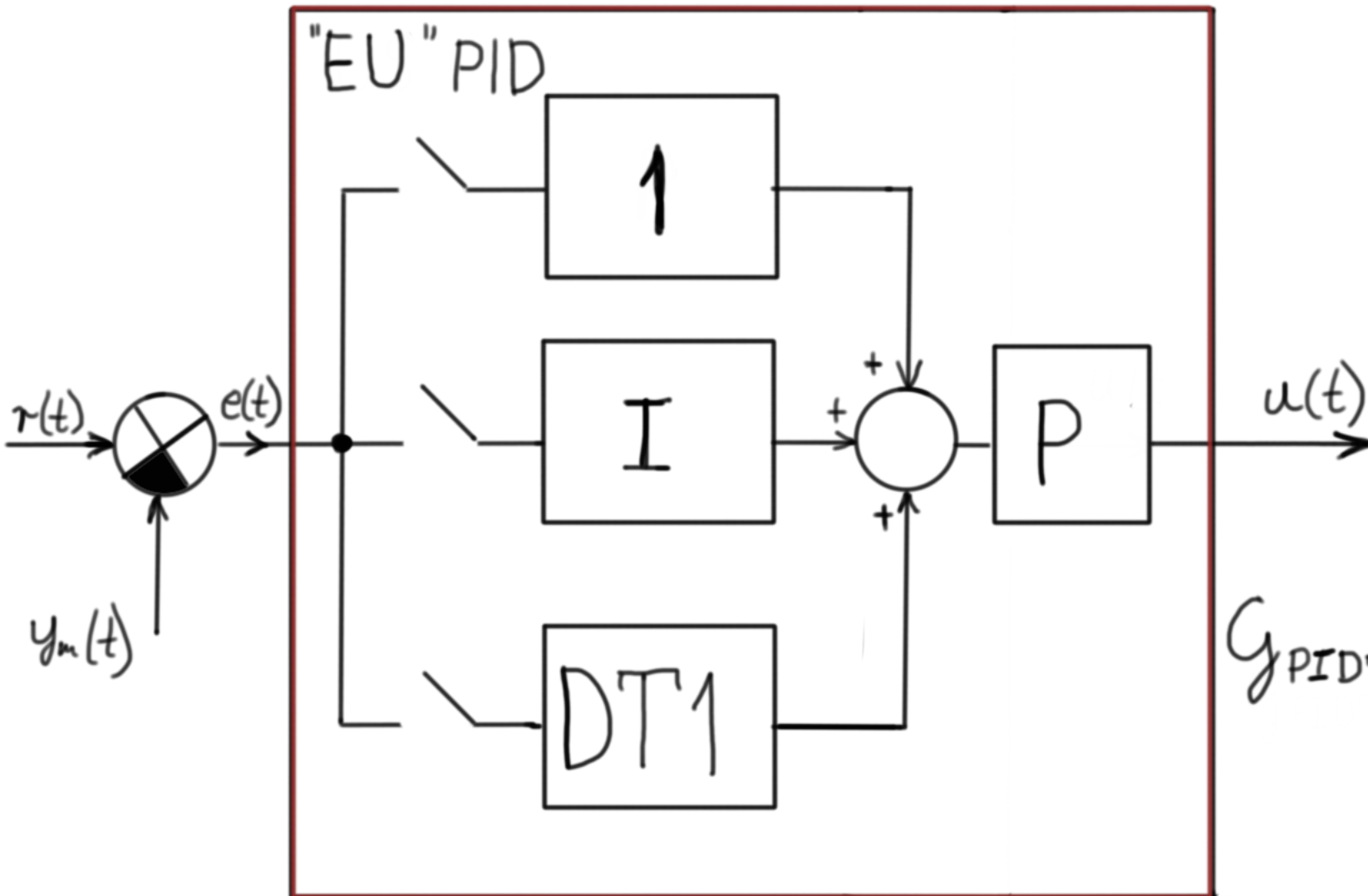




Inside the controller:



Compensator element: 3 switchable (P,I,D) channels, EU/US structure:

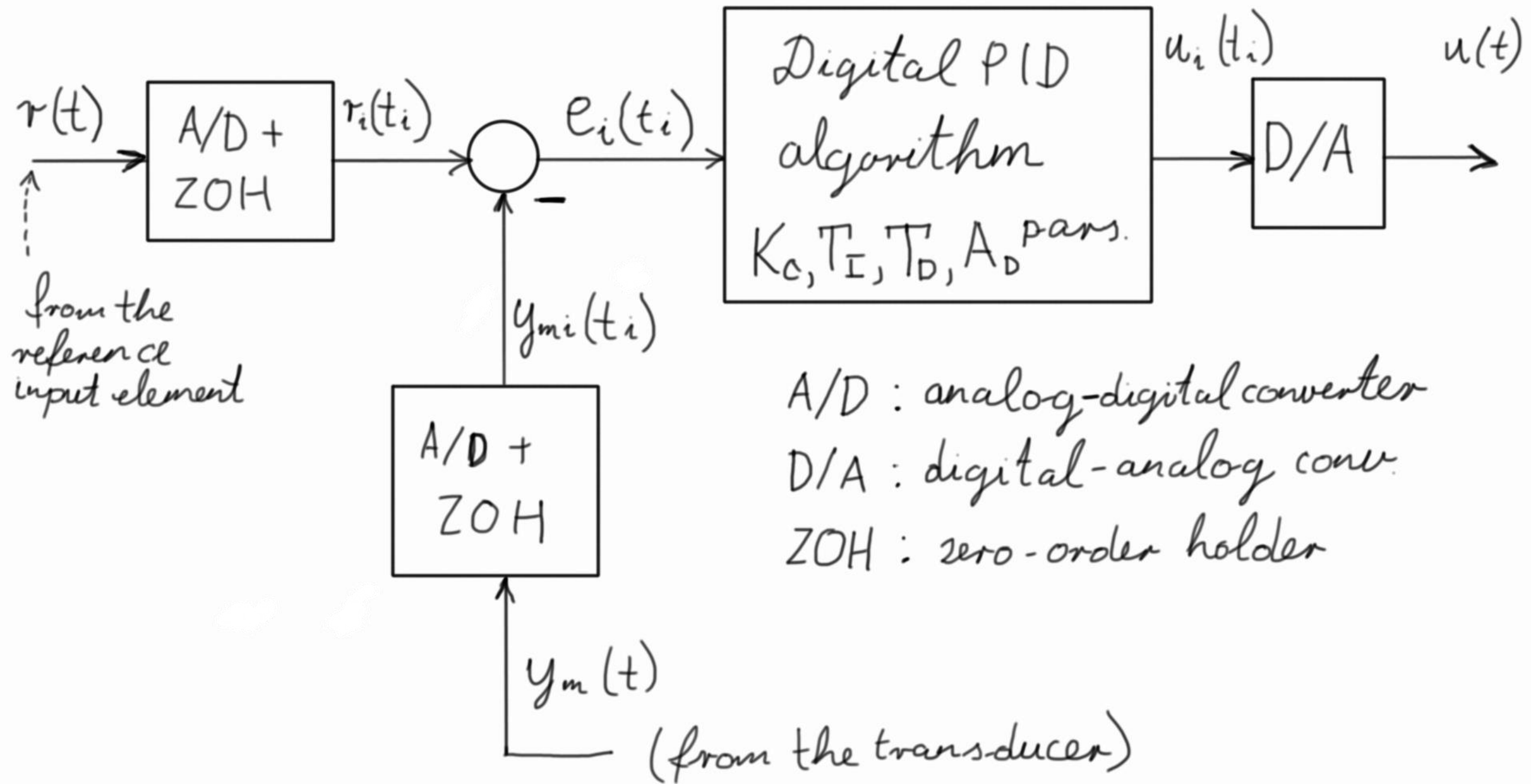


$$G_P(s) = K_c$$

$$G_I(s) = \frac{1}{sT_I}$$

$$G_{DT1}(s) = \frac{sT_D}{sT_D + 1}$$

$$G_{PIDT1}(s) = [1 + G_I(s) + G_{DT1}(s)] \cdot K_c$$

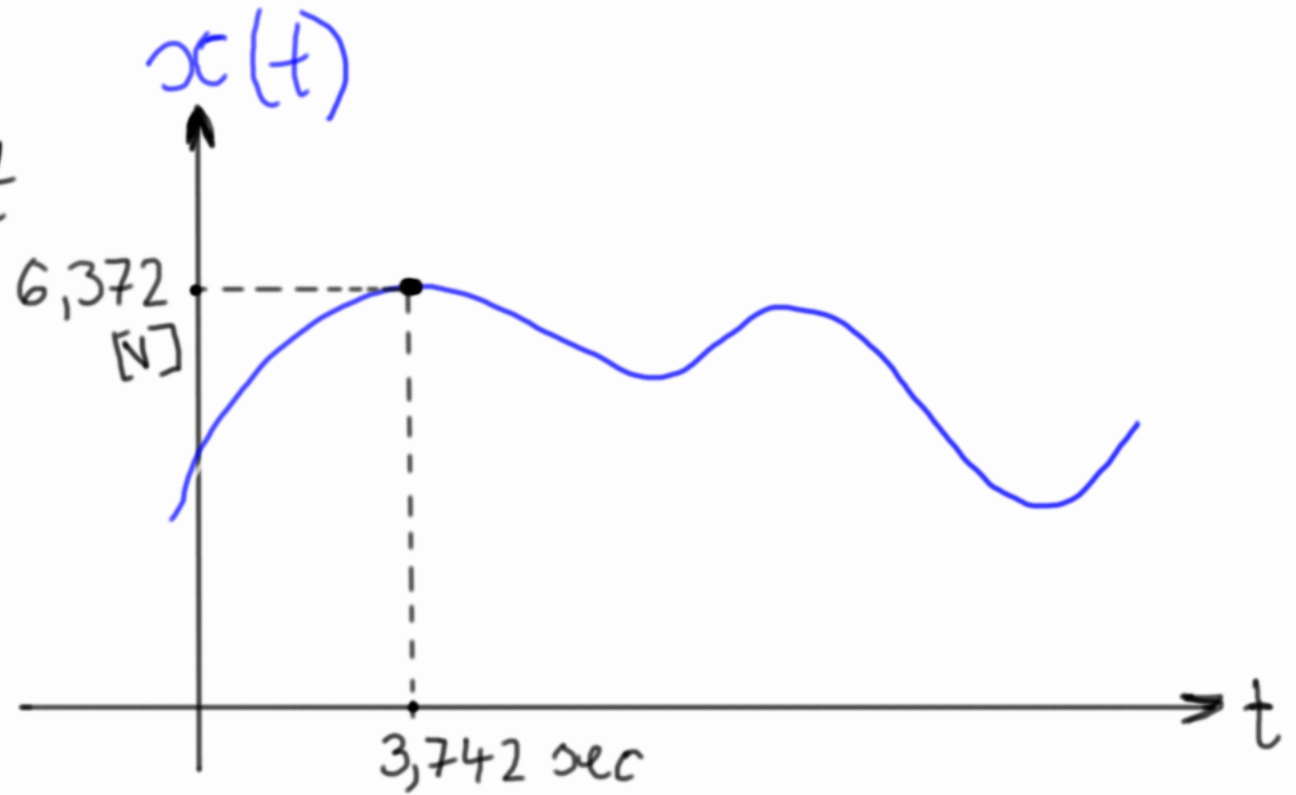


Continuous analog signal:

- interpreted at any t time instant
- x can be any real number

Digital (discrete) signal:

- interpreted only at specific t_i time instances
- x can take values from specific set of values.



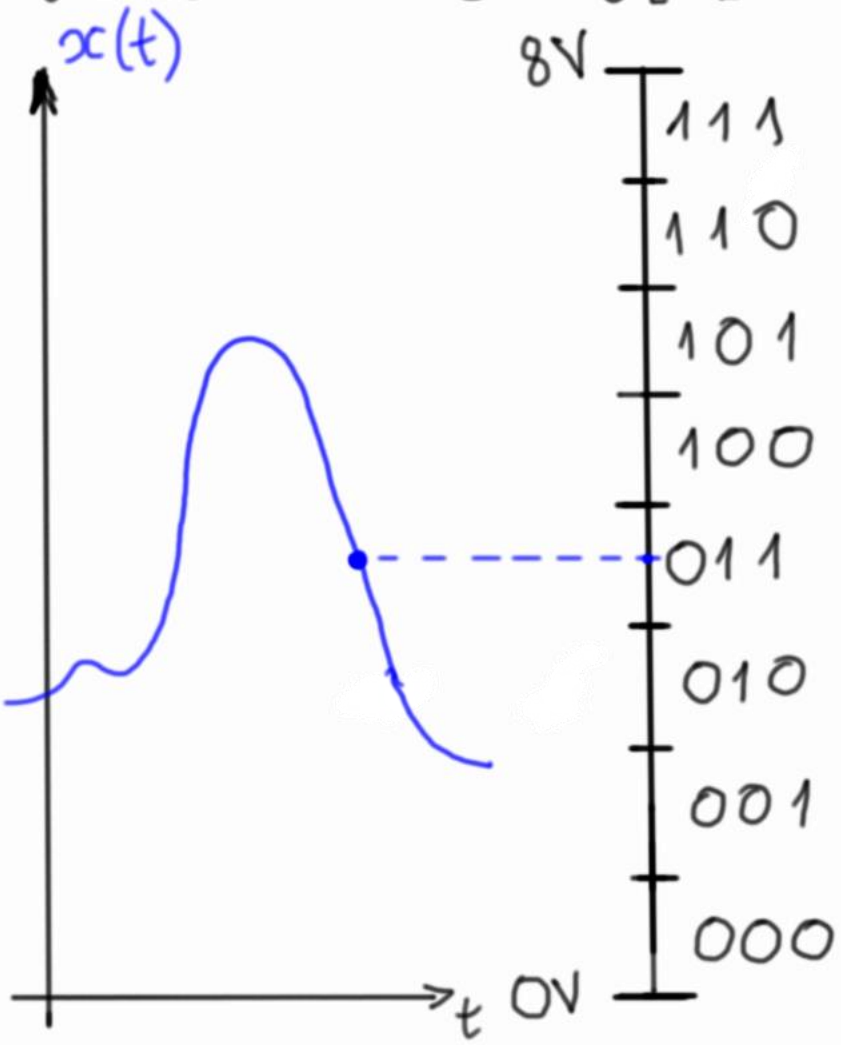
Quantification in value and time!

A/D

ZOH

Quantifying in value. Eg. using a 3 bit A/D converter:

• Some transducer provides an analog output signal varying between 0-8[V]



• dividing the 0-8V range to 8 sub-range, identified by a 3 bit binary number.

• coercing the actual analog signal value to the nearest digital number

eg. $x(t) = 2,7 \approx 3 = 011^{(b)}$

• the digitalised value can take only the values, which is defined by the sub-domain identifiers!

When using n bits in A/D we can divide the whole range of the analog signals into 2^n sub-ranges, where n is the bit number of the A/D converter:

n bit $\rightarrow 2^n$ sub-domain

eg. 3 bits $\rightarrow 2^3 = 8$ sub-domain
4 bits $\rightarrow 2^4 = 16$ ——— " ———

commonly used A/D converters:

10 bit : $2^{10} = 1024$ ——— " ———

12 bit : $2^{12} = 4096$

14 bit 16384

16 bit 65536

The smallest recognizable signal change is the resolution of the A/D converter.

eg. 0-8V, 3 bit A/D : 1 volt

eg. 0-8V, 4 bit A/D : $8V/2^4 = 8V/16 = 0,5V$

0-10V, 10 bit A/D : $10V/2^{10} = 10V/1024 =$
 $= 0,0098V \approx 0,01$

Quantification in time (sampling)

- the signal values are not interpreted at any arbitrary time instances!

- they are evaluated only at t_i time instances, where:

$$t_i = i \cdot \Delta t$$

$$i = 0, 1, 2, 3 \dots n$$

Δt : sampling time

$x_i(t_i)$:

$$t_0 \longmapsto x_0$$

$$t_1 \longmapsto x_1$$

$$t_3 \longmapsto x_3$$

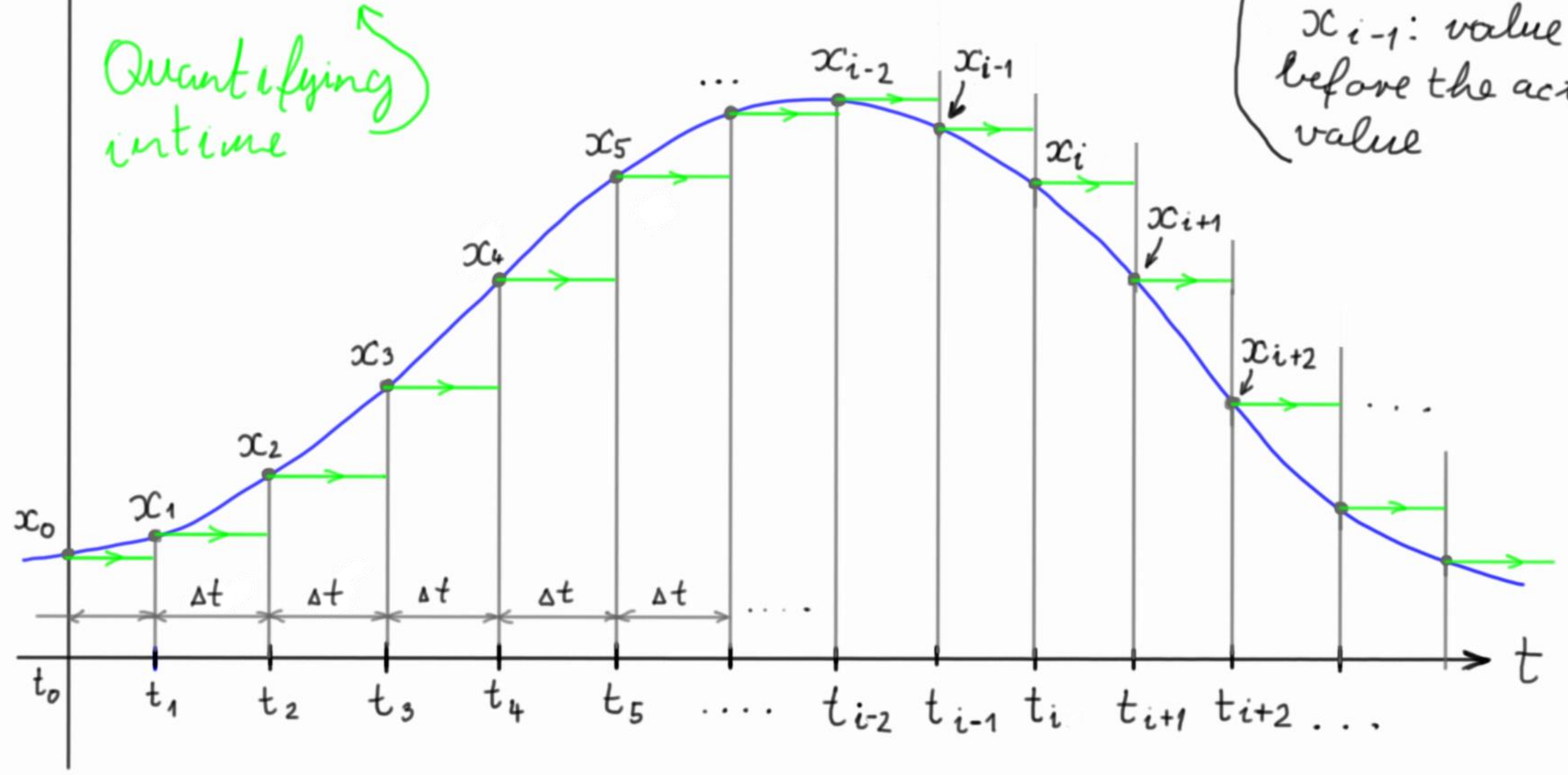
.....

$$t_n \longmapsto x_n$$

$x(t)$ ZOH: the actual x_i kept constant until the next available x_{i+1} value.

x_i : actual val.
 x_{i-1} : value before the actual value

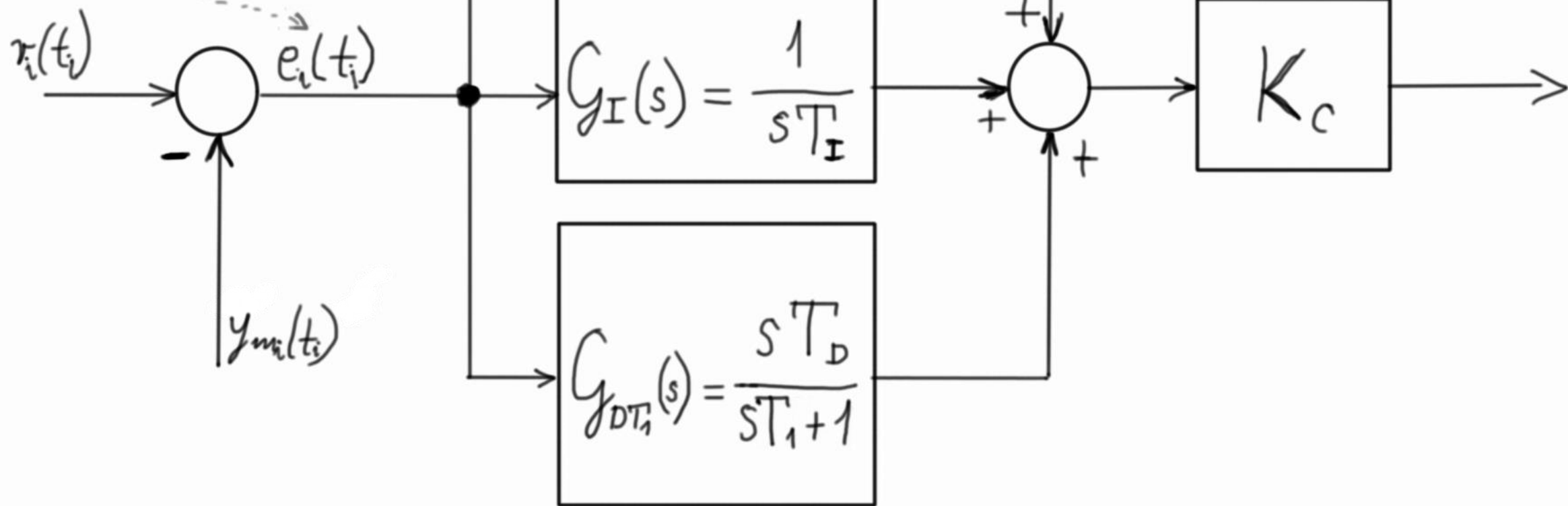
Quantizing
intime



Realizing the PID channels (EU structure):

$$e(t) \rightarrow e_i(t_i)$$

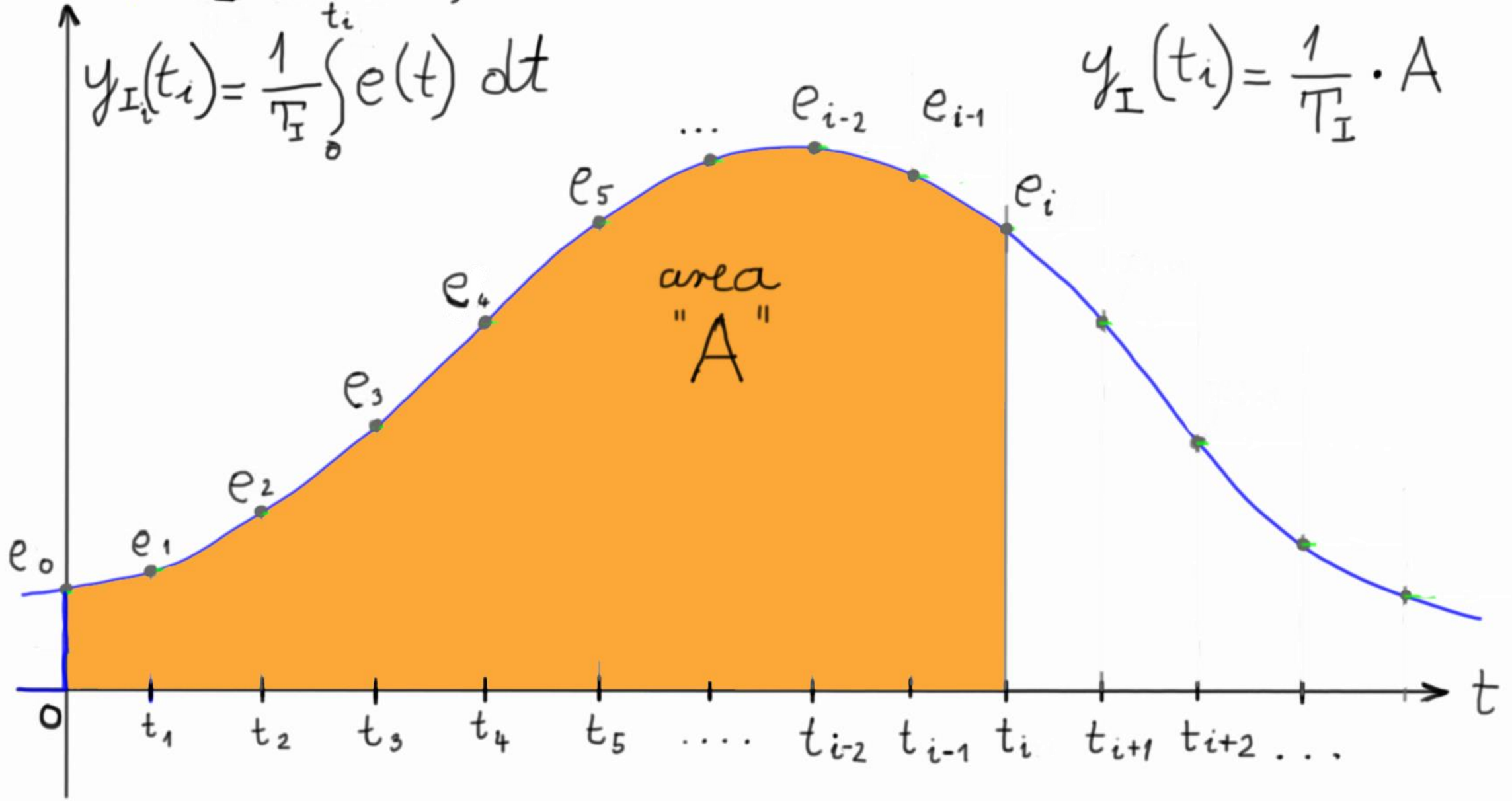
$$e_i(t_i) = r_i(t_i) - y_{mi}(t_i)$$



$e(t)$ I channel, continuous case:

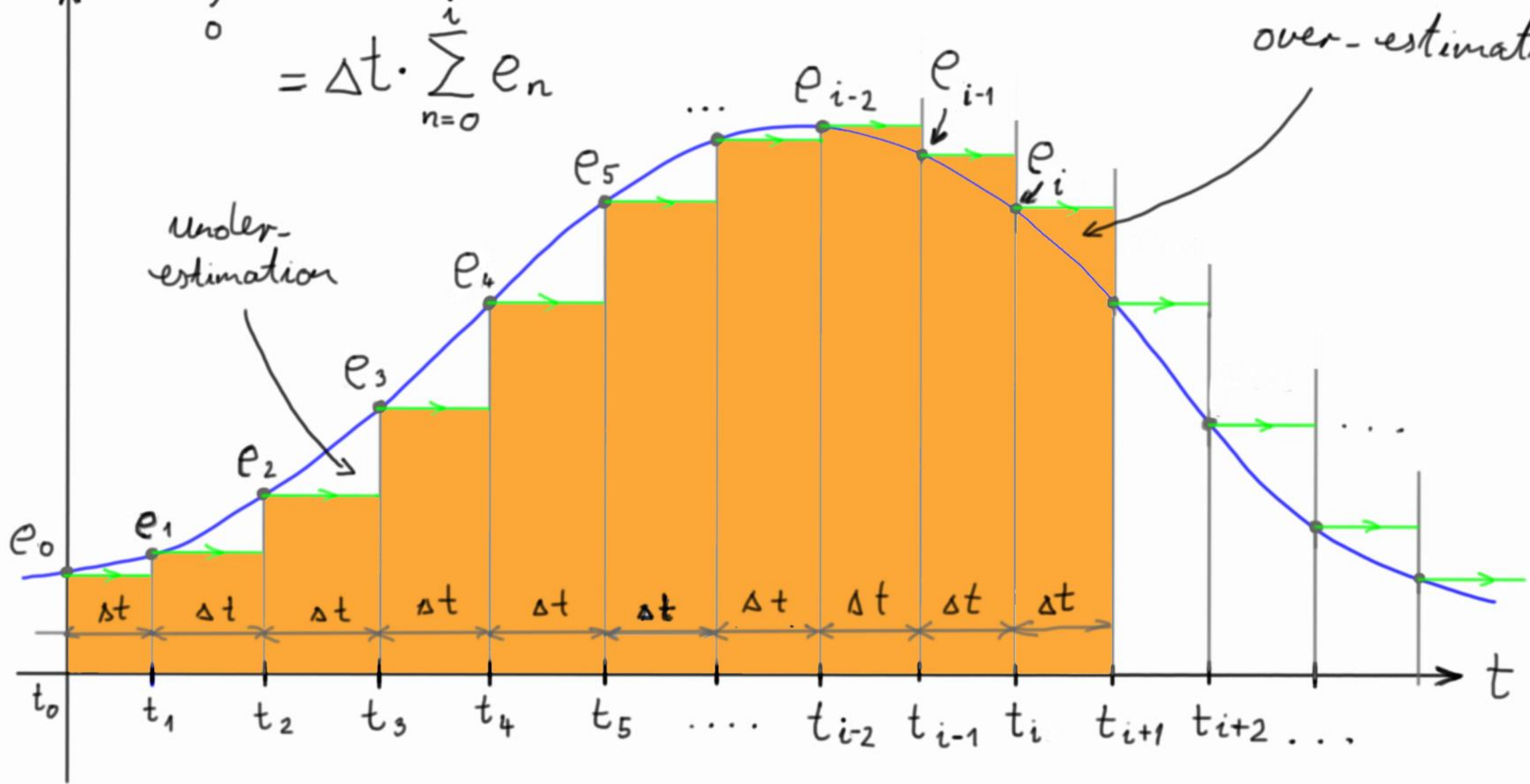
$$y_{I_i}(t_i) = \frac{1}{T_I} \int_0^{t_i} e(t) dt$$

$$y_I(t_i) = \frac{1}{T_I} \cdot A$$



$$e(t) \int_0^t e(t) dt \approx e_0 \Delta t + e_1 \Delta t + e_2 \Delta t + \dots + e_{i-1} \Delta t + e_i \Delta t =$$

$$= \Delta t \cdot \sum_{n=0}^i e_n$$



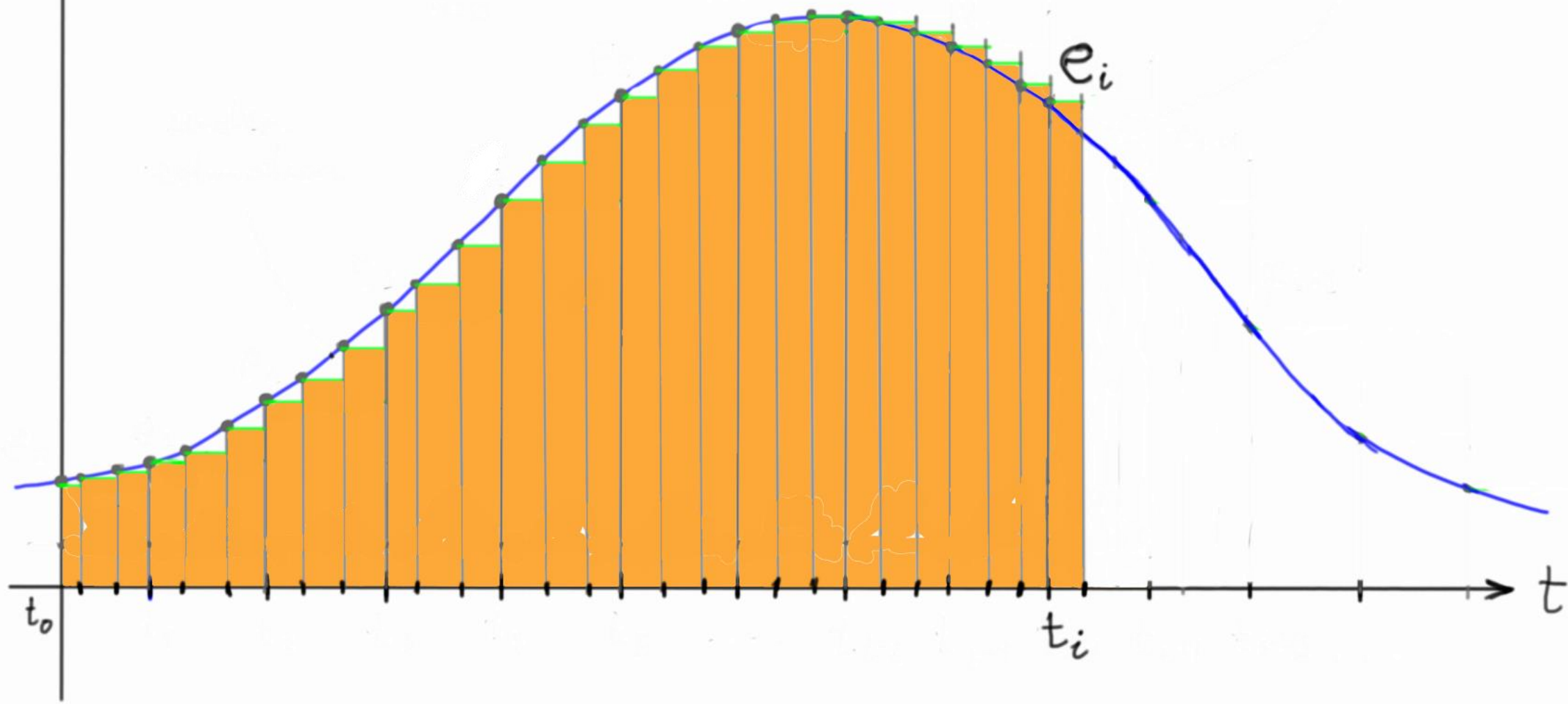
Digital I channel:

$$y_{Ii}(t_i) = \Delta t \cdot \frac{1}{T_I} \cdot \sum_{n=0}^i e_n$$

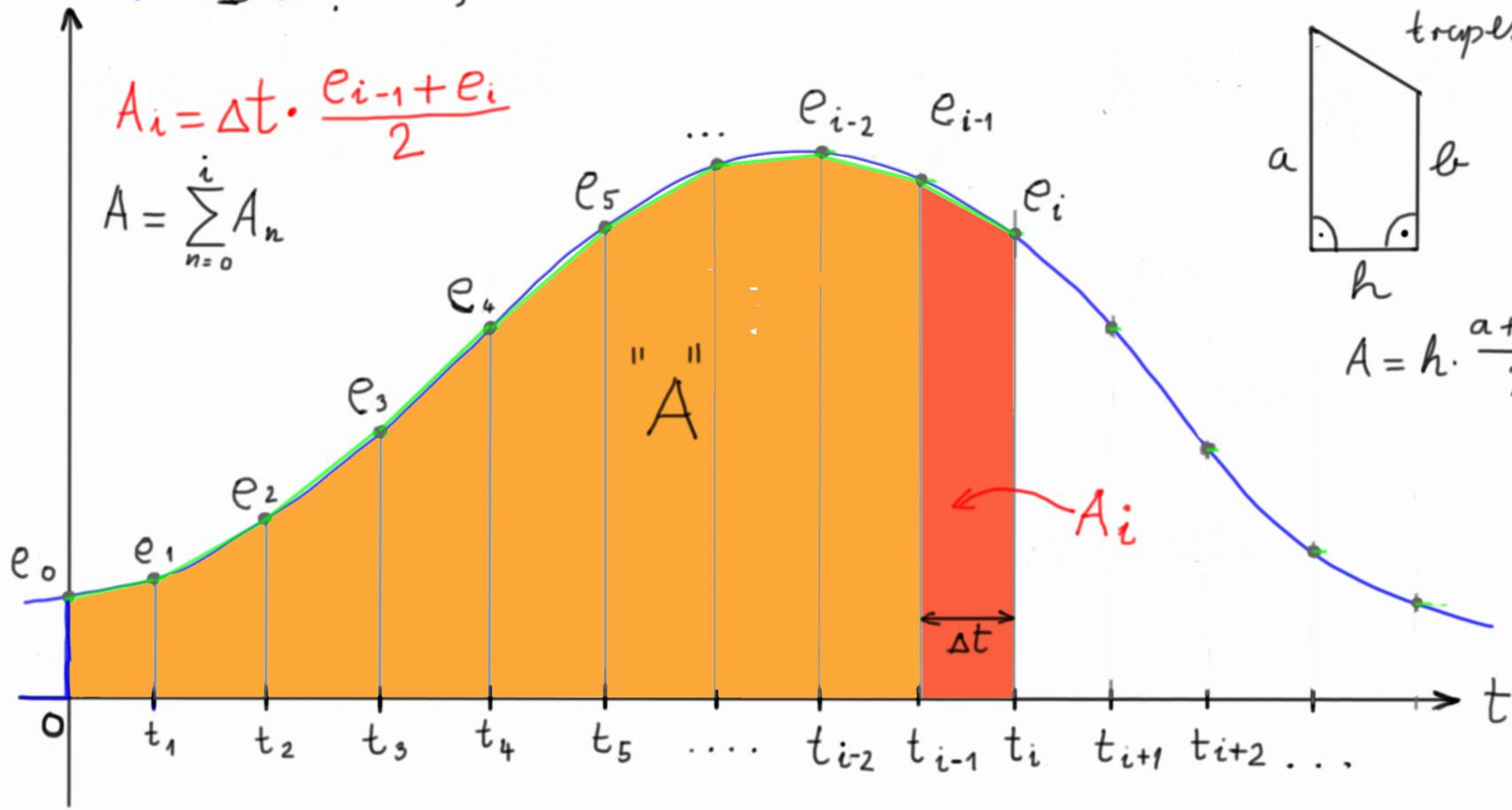
- precision of the digital I channel can be improved by:
- decreasing Δt sampling time
 - using the trapezoid rule

$e(t)$

if $\Delta t \mapsto 0$ $\Delta t \cdot \sum_{n=0}^i e_n \mapsto \int_0^{t_i} e(t) dt$



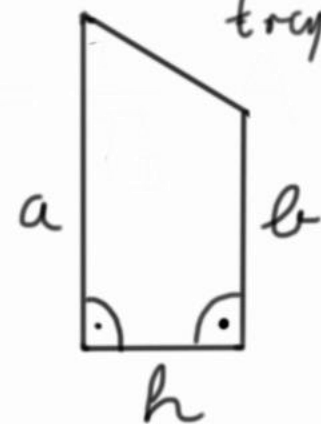
$e(t)$ I channel, continuous case:



$$A_i = \Delta t \cdot \frac{e_{i-1} + e_i}{2}$$

$$A = \sum_{n=0}^i A_n$$

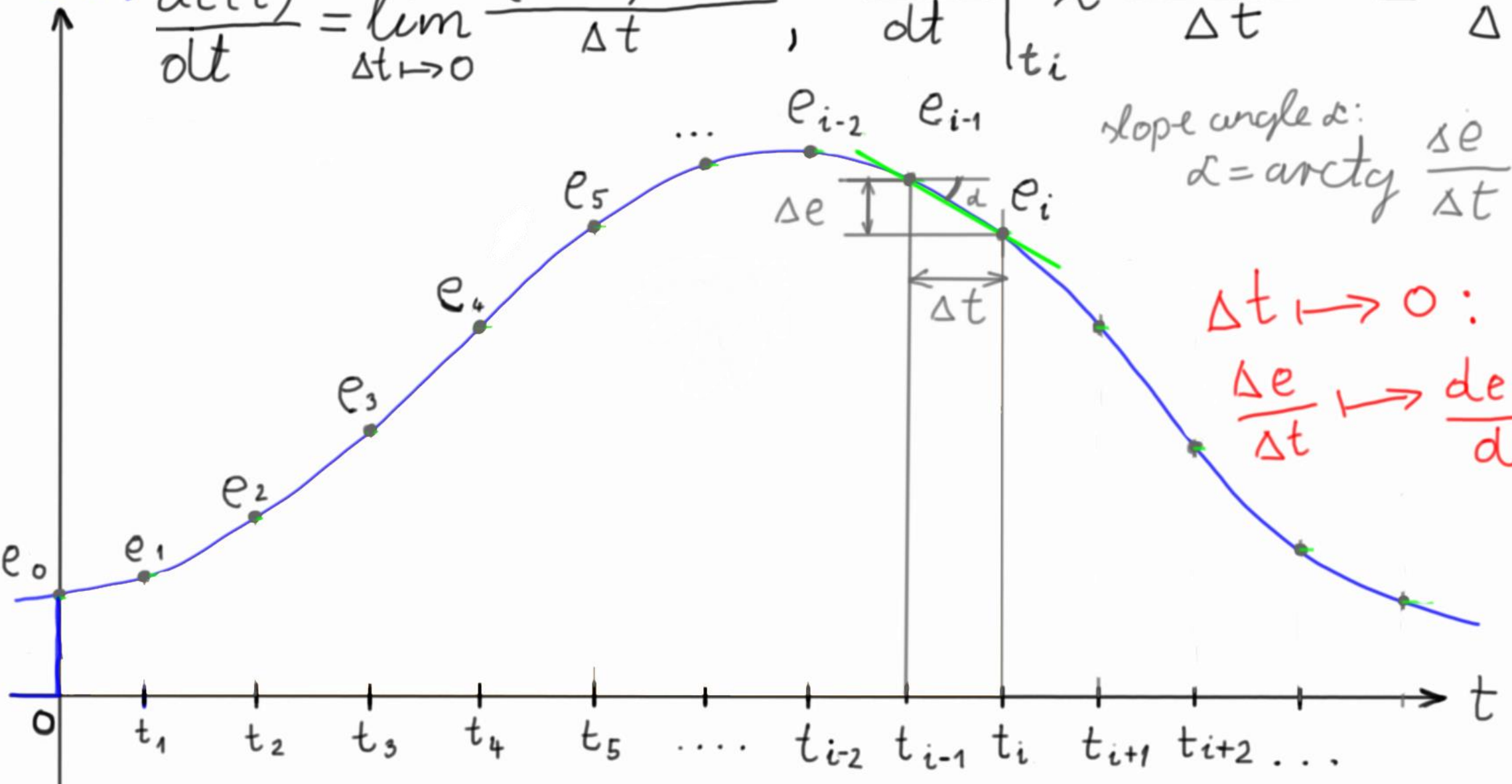
Area of a trapezoid:



$$A = h \cdot \frac{a+b}{2}$$

Pure D channel, continuous and digital case:

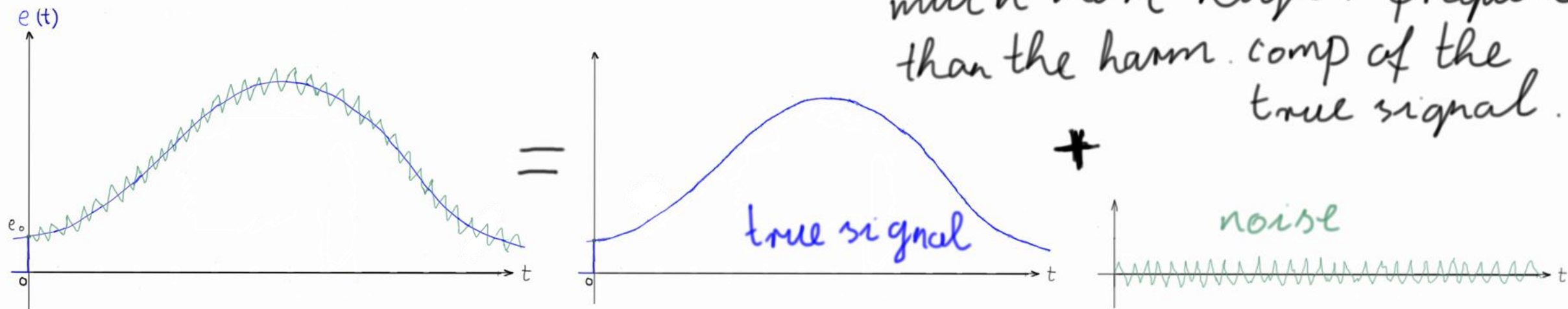
$$e(t) \frac{de(t)}{dt} = \lim_{\Delta t \rightarrow 0} \frac{e(t+\Delta t) - e(t)}{\Delta t}; \quad \left. \frac{de(t)}{dt} \right|_{t_i} \approx \frac{e_i - e_{i-1}}{\Delta t} = \frac{\Delta e}{\Delta t}$$



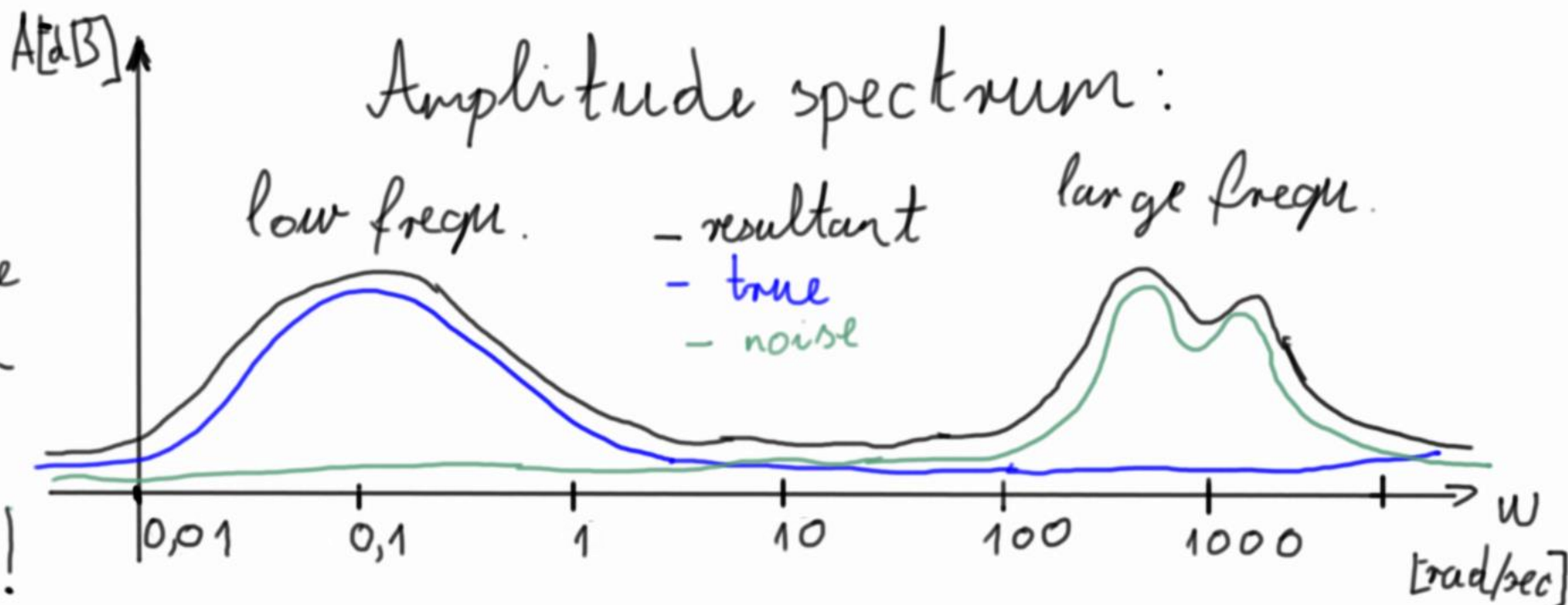
slope angle α :
 $\alpha = \arctan \frac{\Delta e}{\Delta t}$

$\Delta t \rightarrow 0$:
 $\frac{\Delta e}{\Delta t} \rightarrow \frac{de(t)}{dt}$

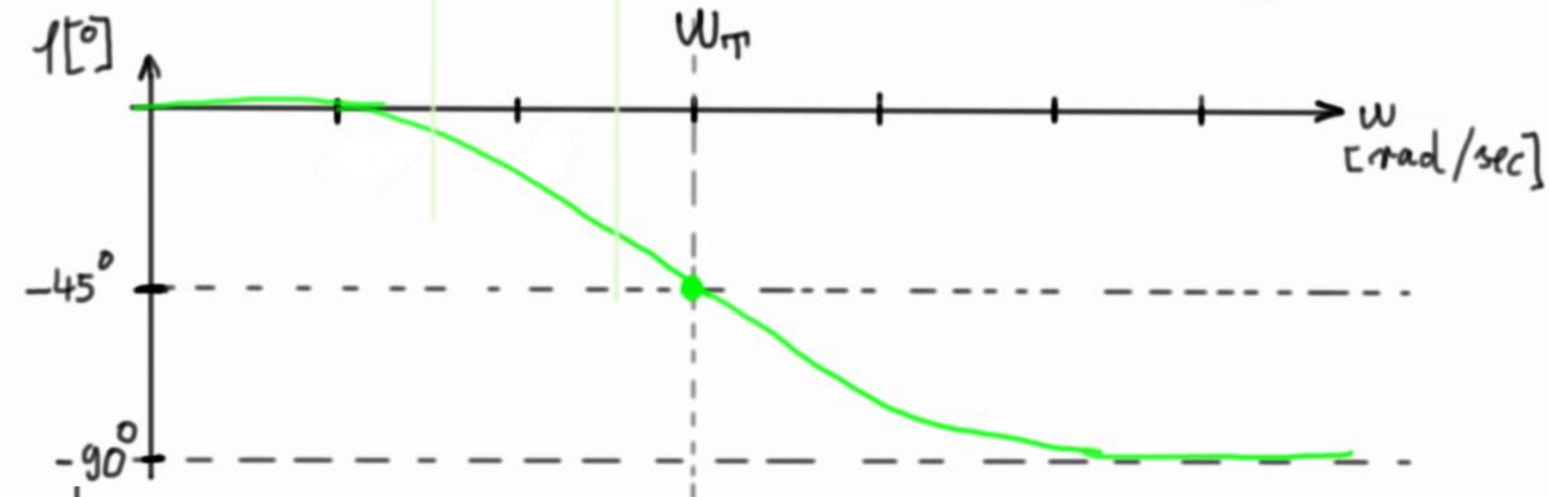
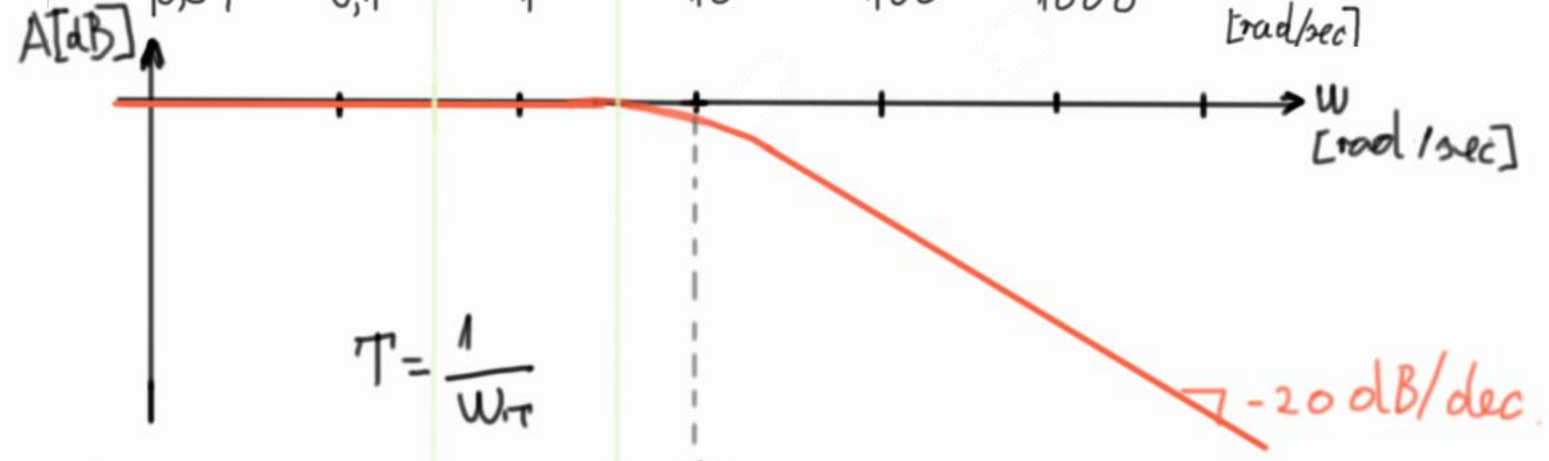
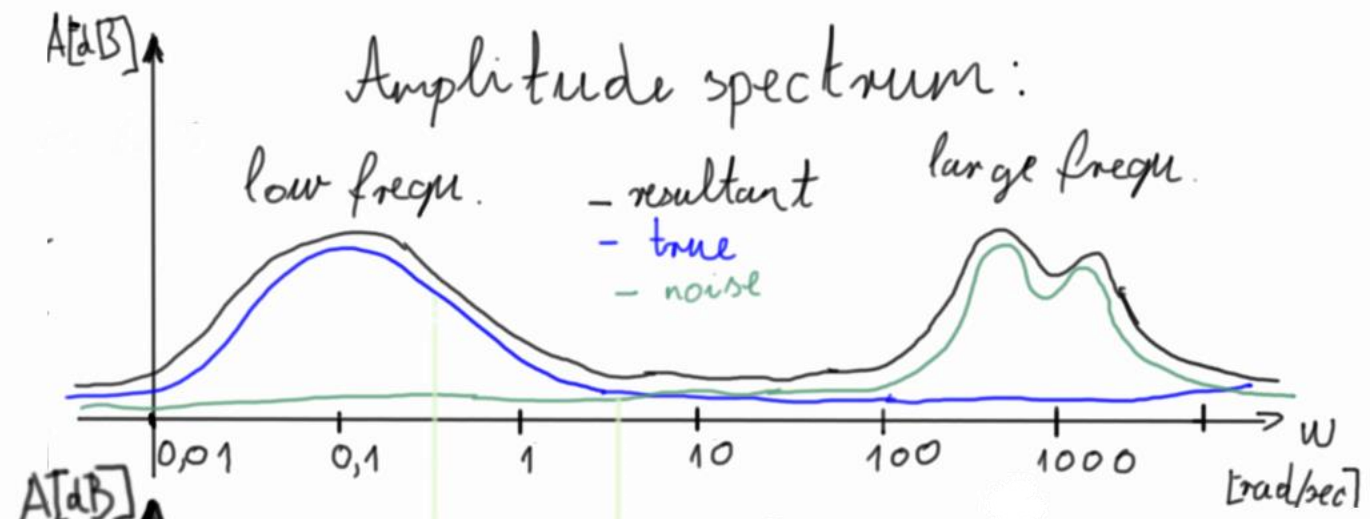
Usually, the noise is consisting of harmonic components with much more higher frequencies than the harm. comp of the true signal.



- the noise content can be eliminated by filtering out the high frequency components!
 → low pass filter!

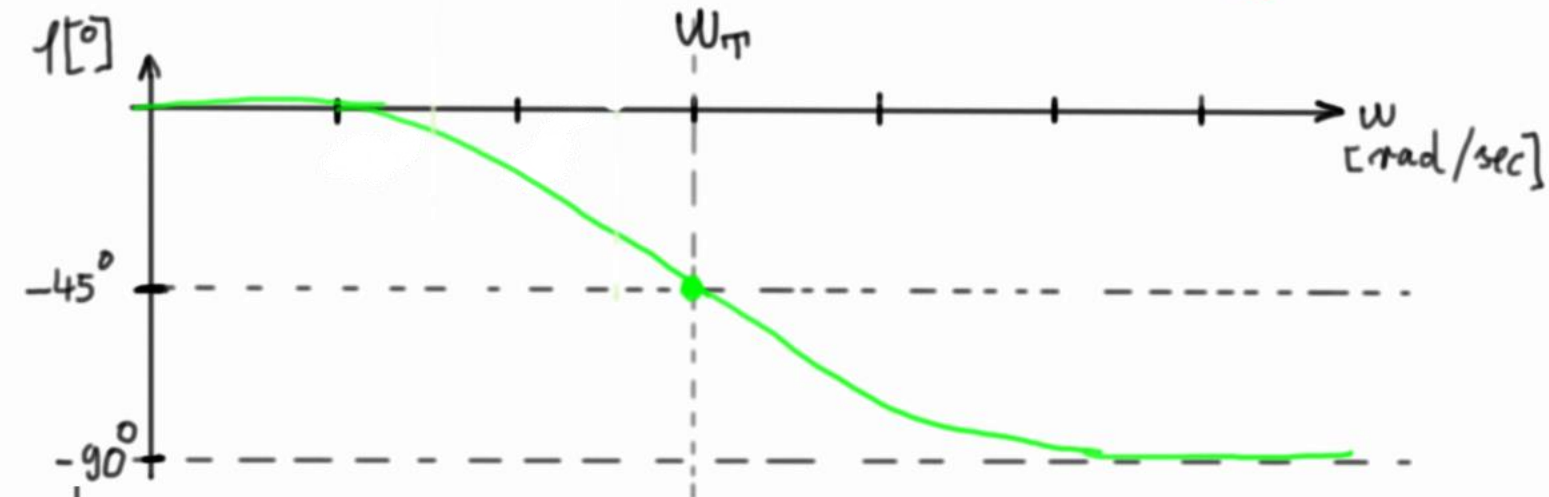
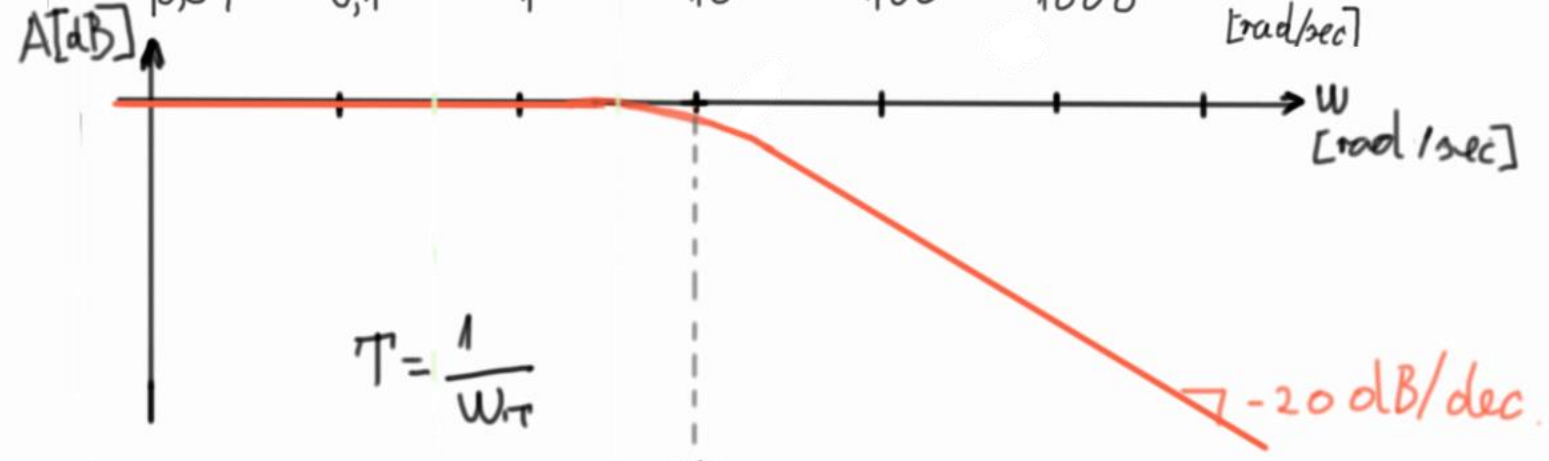
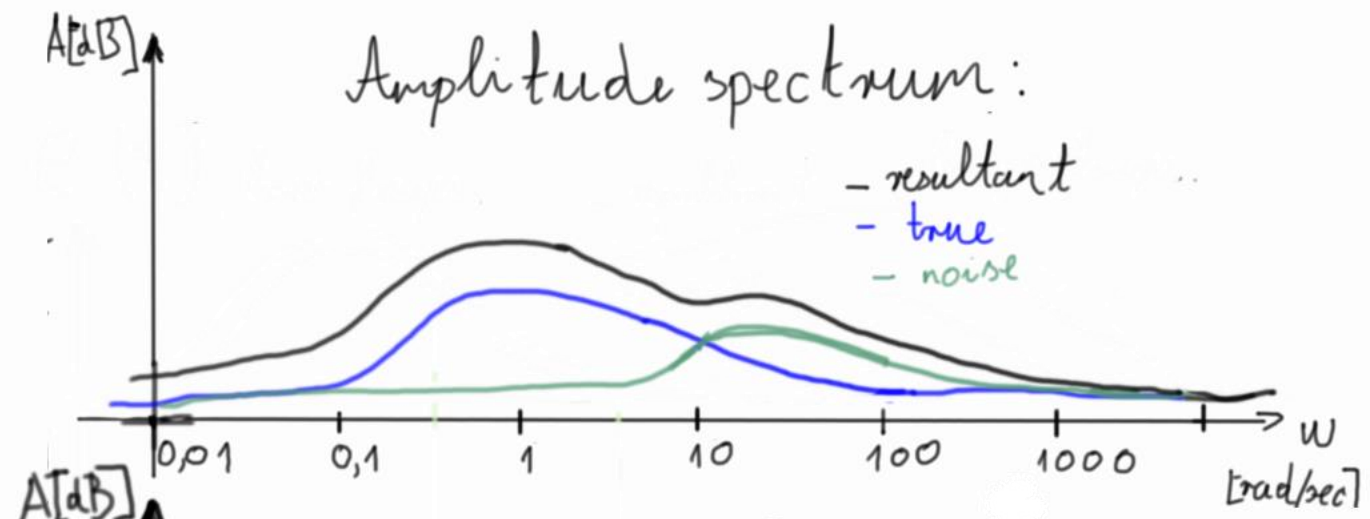


Amplitude spectrum:



- if T (time constant of the T_1 block inside the DT_1 channel) is selected carefully, the noise content can be damped.
- however, some components of the true signal can also get some extra phasift
- if the noise and the true signal does not separates clearly, the true signal components can be attenuated, as well.

Amplitude spectrum:



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How can we realise a digital T_1 low-pass filter?

Diffe: $T \frac{dy(t)}{dt} + y(t) = x(t)$

$y_0(t_0=0) = 0 \rightarrow$ known

$\frac{dy(t)}{dt} \approx \frac{y_1 - y_0}{\Delta t}$ unknown

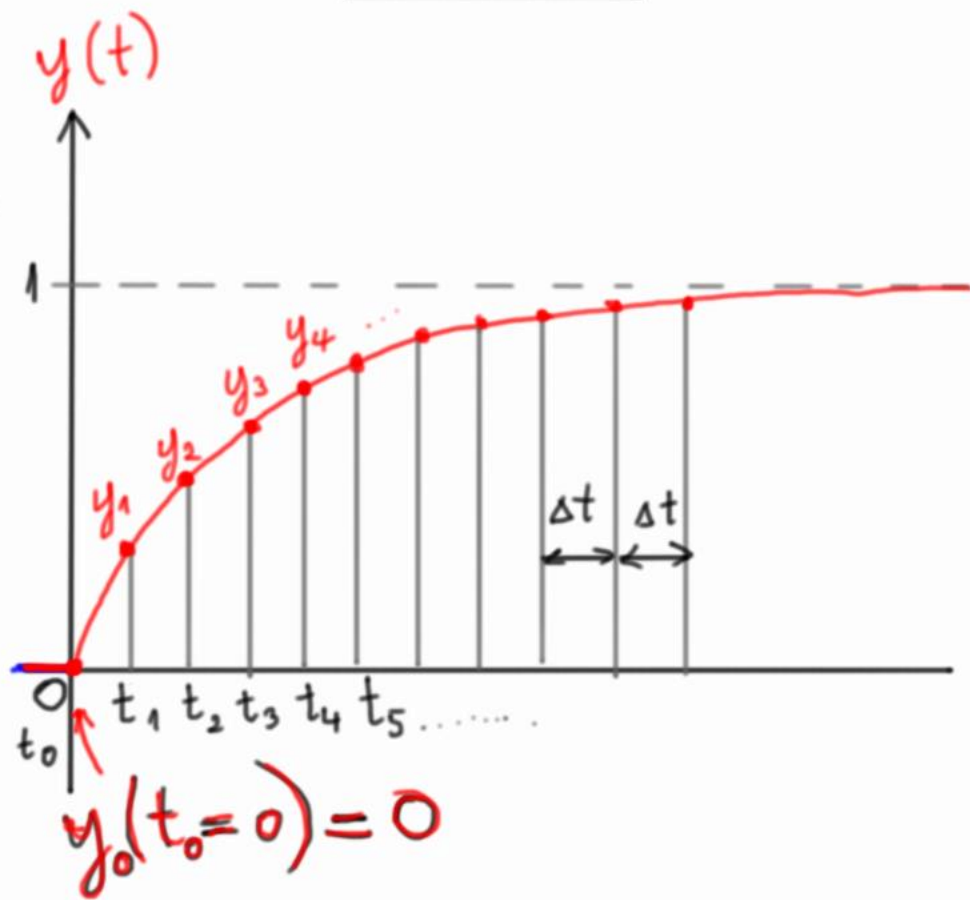
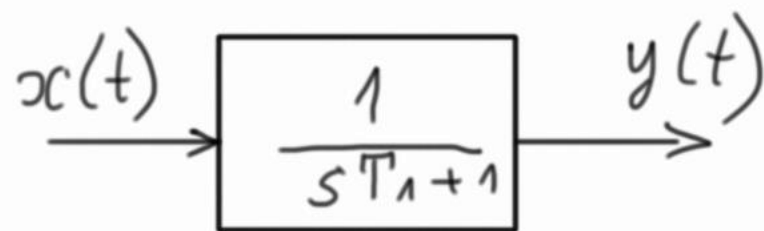
let's use a unit step input first
 $1(t) \approx 1$

$T \cdot \frac{y_1 - y_0}{\Delta t} + y_0 = 1(t)$

$T \cdot \frac{y_1 - y_0}{\Delta t} + y_0 = 1 \rightarrow (y_1 - y_0) = \frac{\Delta t}{T} (1 - y_0)$

$y_1 = y_0 + \frac{\Delta t}{T} (1 - y_0)$

$T \cdot \frac{y_1 - y_0}{\Delta t} = 1 - y_0$



$$y_1 = y_0 + \frac{\Delta t}{\tau} \cdot (1 - y_0)$$

if y_1 is known, we can calc y_2 :

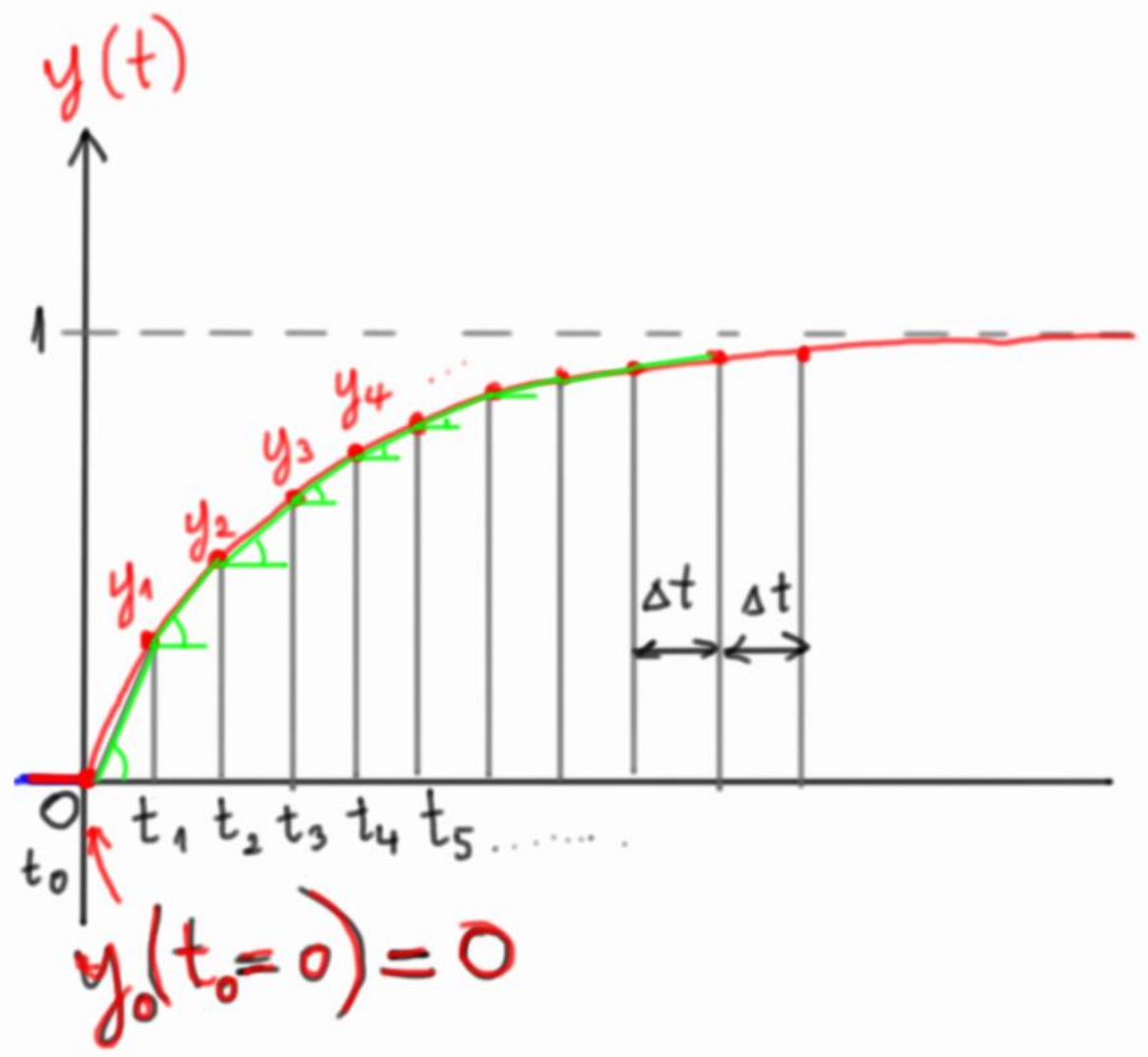
$$y_2 = y_1 + \frac{\Delta t}{\tau} (1 - y_1)$$

$$y_3 = y_2 + \frac{\Delta t}{\tau} (1 - y_2)$$

$$\underbrace{y_i}_{\text{actual}} = \underbrace{y_{i-1}}_{\text{prev.}} + \underbrace{\frac{\Delta t}{\tau} (1 - y_{i-1})}_{\text{increase}}$$

$y \uparrow \longrightarrow \downarrow \text{increase}$

(iterative calc. method)



If the input is an arbitrary $x(t)$ function

$$y_i = y_{i-1} + \frac{\Delta t}{T} (x_i - y_{i-1})$$

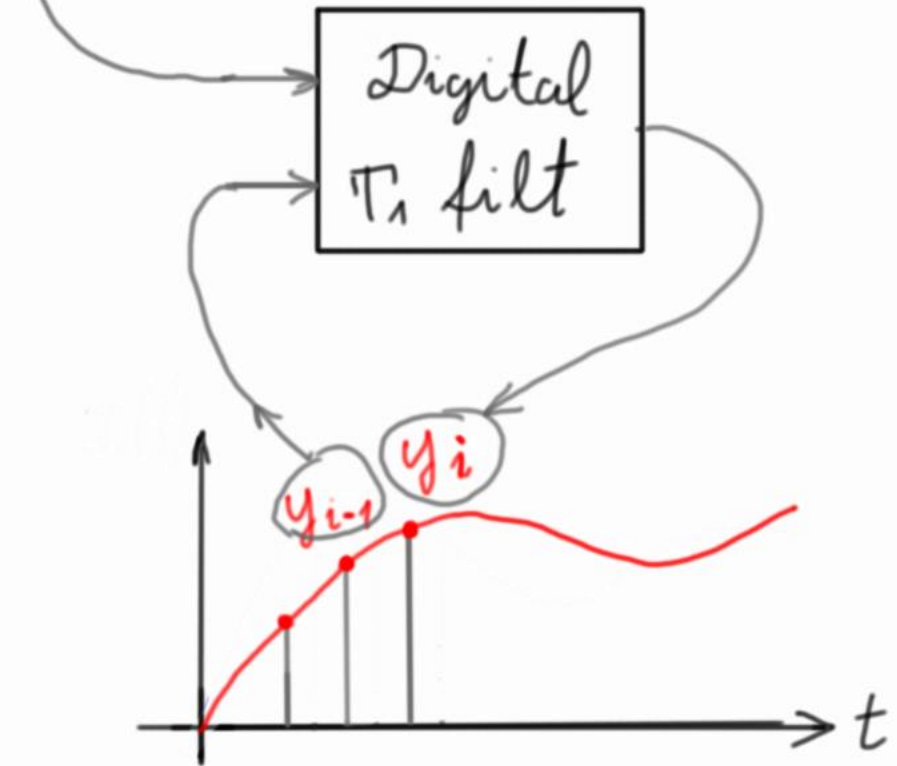
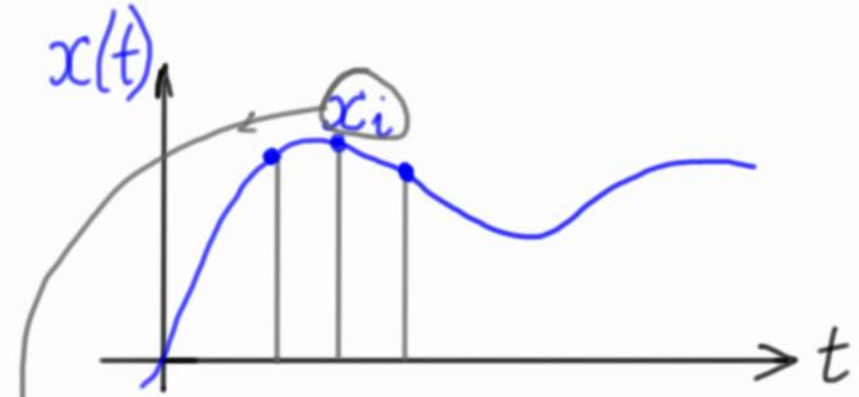
y_i : current filter output

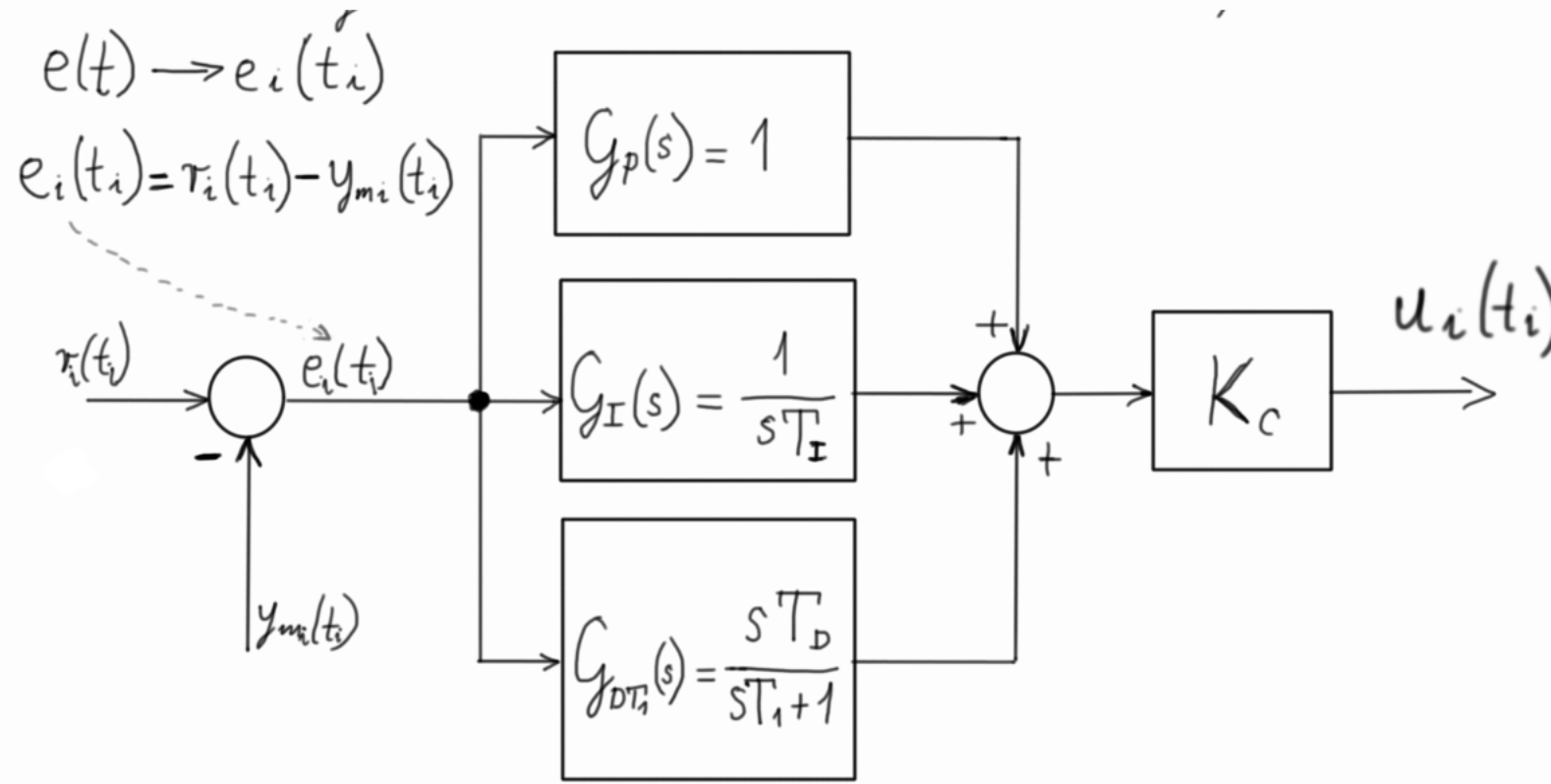
y_{i-1} : previous filter output

x_i : current filter input

D T_1 output:

$$y_{DT_1,i} = \frac{y_i - y_{i-1}}{\Delta t}$$





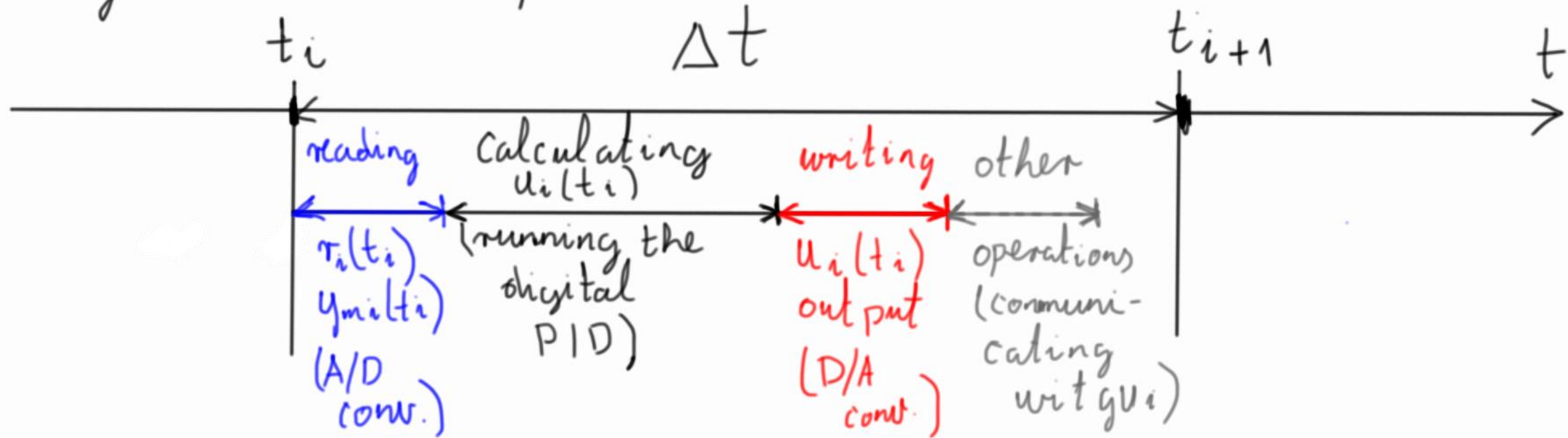
$$u_i(t_i) = \left[e_i(t_i) + \frac{\Delta t}{T_I} \sum_{n=0}^i e_i + \frac{e_{\text{filt},i} - e_{\text{filt},i-1} \cdot T_D}{\Delta t} \right] \cdot K_c$$

$$e_{\text{filt},i} = e_{\text{filt},i-1} + \frac{\Delta t}{T_I} (e_i(t_i) - e_{\text{filt},i-1})$$

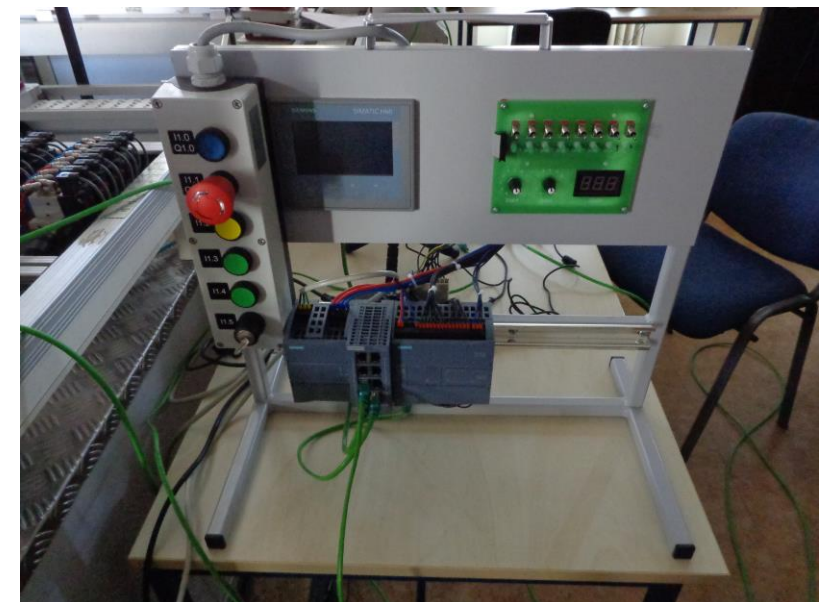
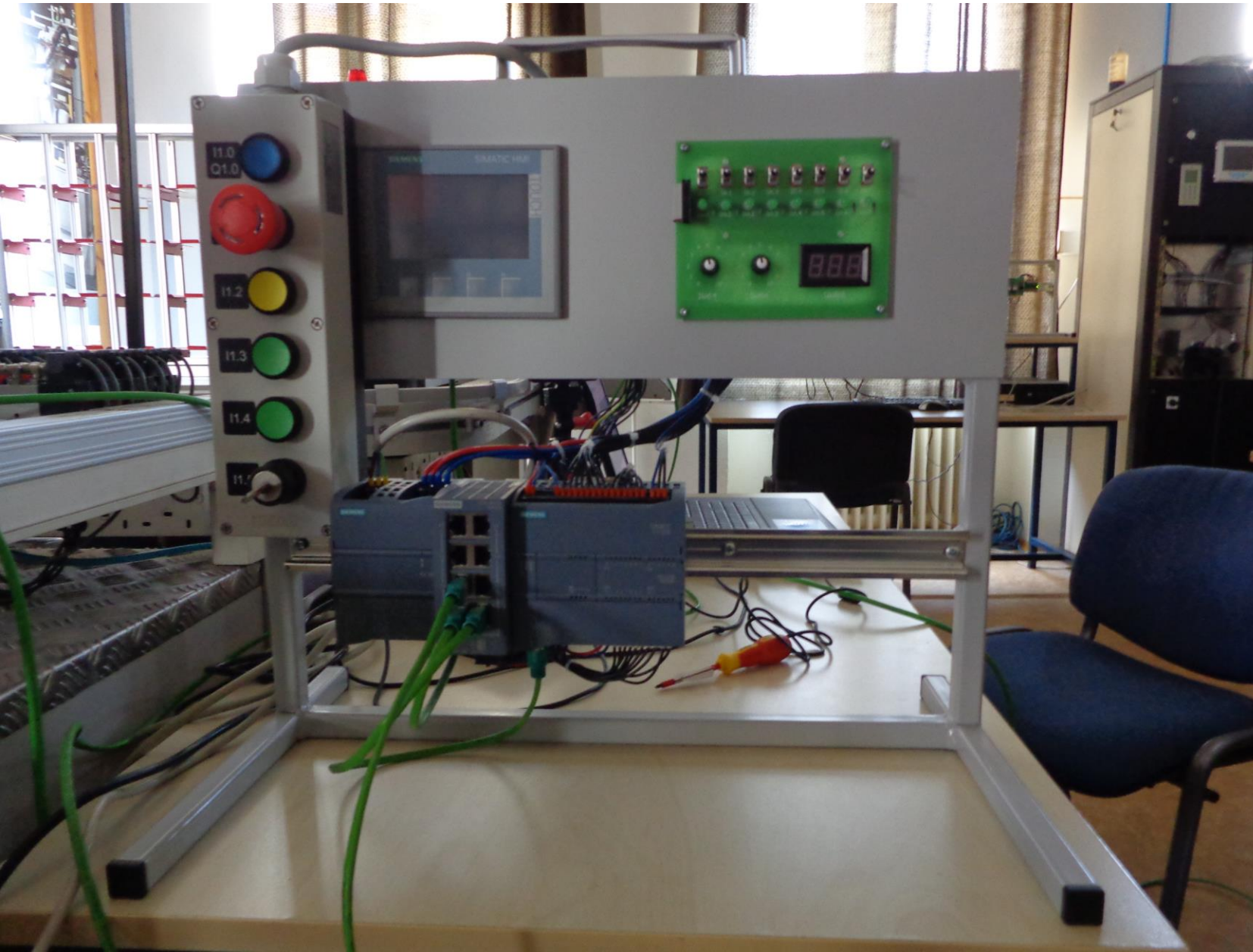
Why not use even smaller Δt values? ^{digital}

– with smaller Δt values, the D, I or T_i channel outputs approximates the continuous case even better...

– but: the microcontroller has to finish all operations during the Δt time period!



Siemens S7 1200 PLC



All lab materials available at:

<https://rosz.uni-obuda.hu/esemenyek/plc-s-szabalyzasok-tanfolyam/>

Or:

<https://classroom.google.com/c/NjlyMzYxODg4MTY5?cjc=dhc4ud7>