

TFE4188 - Lecture 5

Switched-Capacitor Circuits

Housekeeping

Added "Continuous-time analog filters in nano-scale CMOS, challenges and opportunities" video to Lecture 4

Added Lecture 6 papers. Read them until next time.

Lecture in week 8 will be digital

No-one is showing up to the physical exercise hours, so we go digital

Week	Book	Monday	Project plan	Exercise
2	CJM 1-6	Course intro, what I expect you to know, Specification project, analog design fundamentals		
3	Slides	ESD and IC Input/Output	Specification	x
4	CJM 7,8	Reference and bias	Specification	
5	CJM 12	Analog Front-end	M1. Specification review	x
6	CJM 11-14	Switched capacitor circuits	Design	
7	JSSC, CJM 18	State-of-the-art ADCs	Design	x
8	Slides	Low power radio receivers	Design	
9	Slides	Communication standards from circuit perspective	M2. Design review	x
10	CJM 7.4, CFAS,+DC/DC	Voltage regulation	Layout	
11	CJM 19, CFAS	Clock generation	M3. Layout review	x
12	Paper	Energy sources	Layout/LPE simulation	
13	Slides	Chip infrastructure	Layout/LPE simulation	x
14		Tapeout review	M4. Tapeout review	
15		Easter		
16		Easter		
17		Exam repetition		

Goal for today

Understand **why** would use switched capacitor circuits

Introduction to **discrete-time**, and **switched capacitor**

W A W h n y

Active-RC

$$\omega_{p|z} \propto \frac{1}{RC}$$

Gm-C

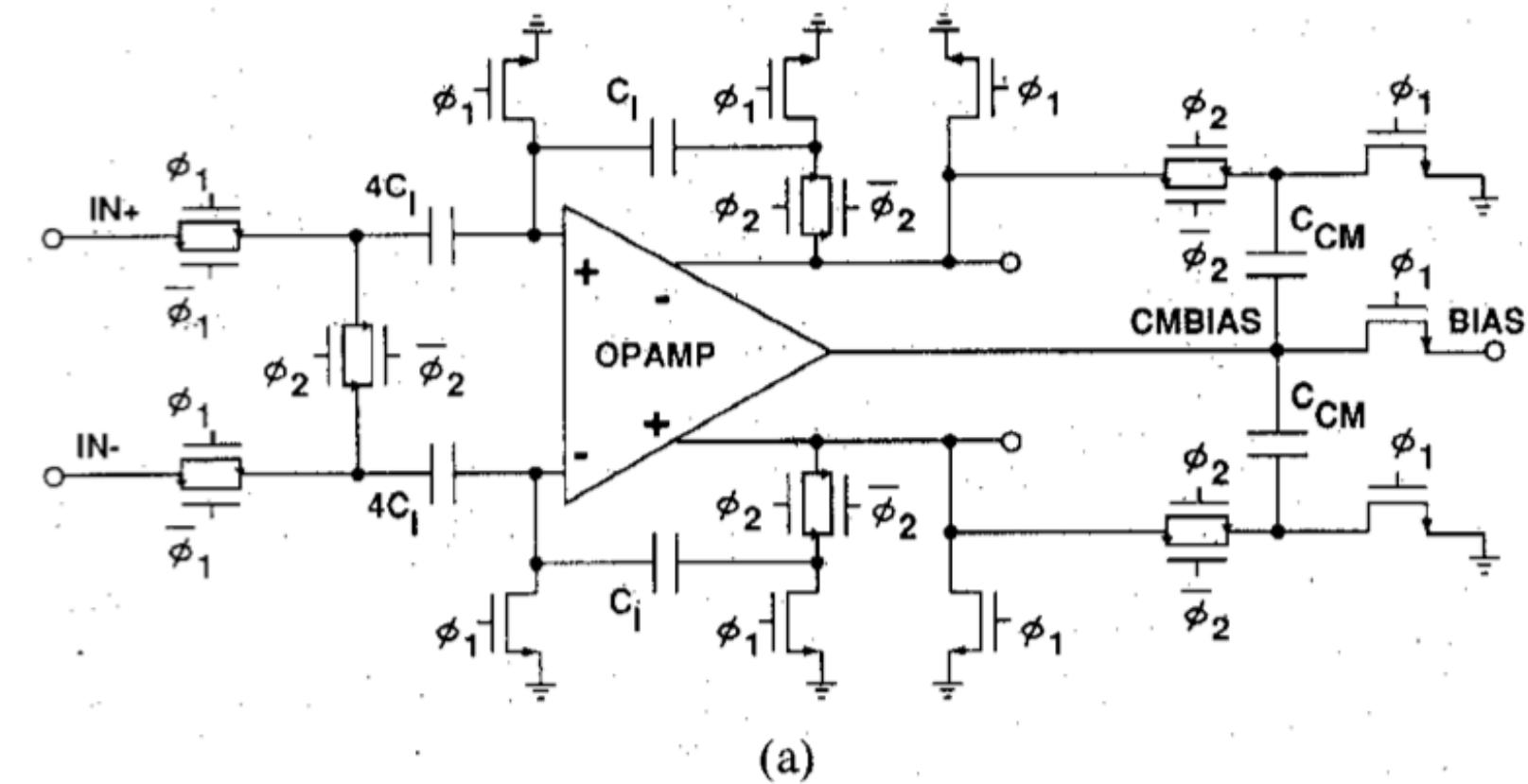
$$\omega_{p|z} \propto \frac{G_m}{C}$$

$\omega_{p|z}$ \propto $c_1 - \overline{c_2}$

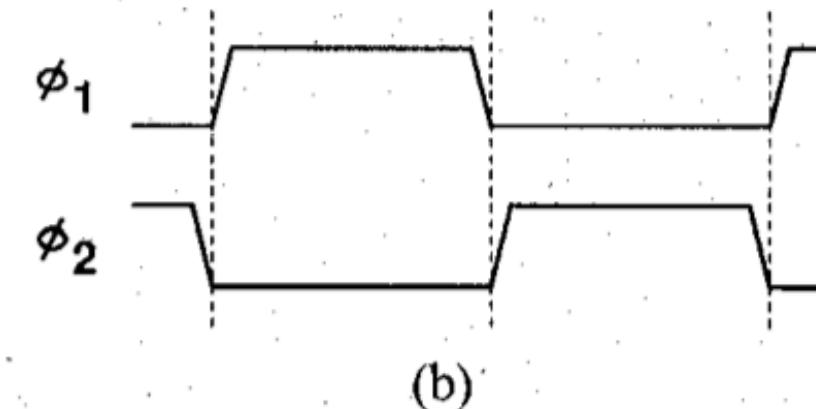
A pipelined 5-Msample/s 9-bit analog-to-digital converter

Good common-mode rejection

Precise gain



(a)



(b)

Fig. 6. (a) Schematic of S/H amplifier. (b) Timing diagram of a two-phase nonoverlapping clock.

Nano-scale CMOS

Advantage

Switches become smaller (less charge injection)

Capacitors become more dense

Challenge

High-gain OTA's (tricky to get 100 dB open loop gain)

Discrete-Time Signals

Define x_c as a continuous time signal, continuous value signal

Define $\ell(t) = \begin{cases} 1 & \text{if } t \geq 0 \\ 0 & \text{if } t < 0 \end{cases}$

Define

$$x_{sn}(t) = \frac{x_c(nT)}{\tau} [\ell(t - nT) - \ell(t - nT - \tau)]$$

Define $x_s(t) = \sum_{n=-\infty}^{\infty} x_{sn}(t)$

A sampled signal of an analog signal you can think of it as an infinite sum of pulse trains where the area under the pulse train is equal to the analog signal.

Why do this?

If $x_s(t) = \sum_{n=-\infty}^{\infty} x_{sn}(t)$

Then $X_{sn}(s) = \frac{1}{\tau} \frac{1 - e^{-s\tau}}{s} x_c(nT) e^{-snT}$

And $X_s(s) = \frac{1}{\tau} \frac{1 - e^{-s\tau}}{s} \sum_{n=-\infty}^{\infty} x_c(nT) e^{-snT}$

Thus $\lim_{\tau \rightarrow 0} X_s(s) = \sum_{n=-\infty}^{\infty} x_c(nT) e^{-snT}$

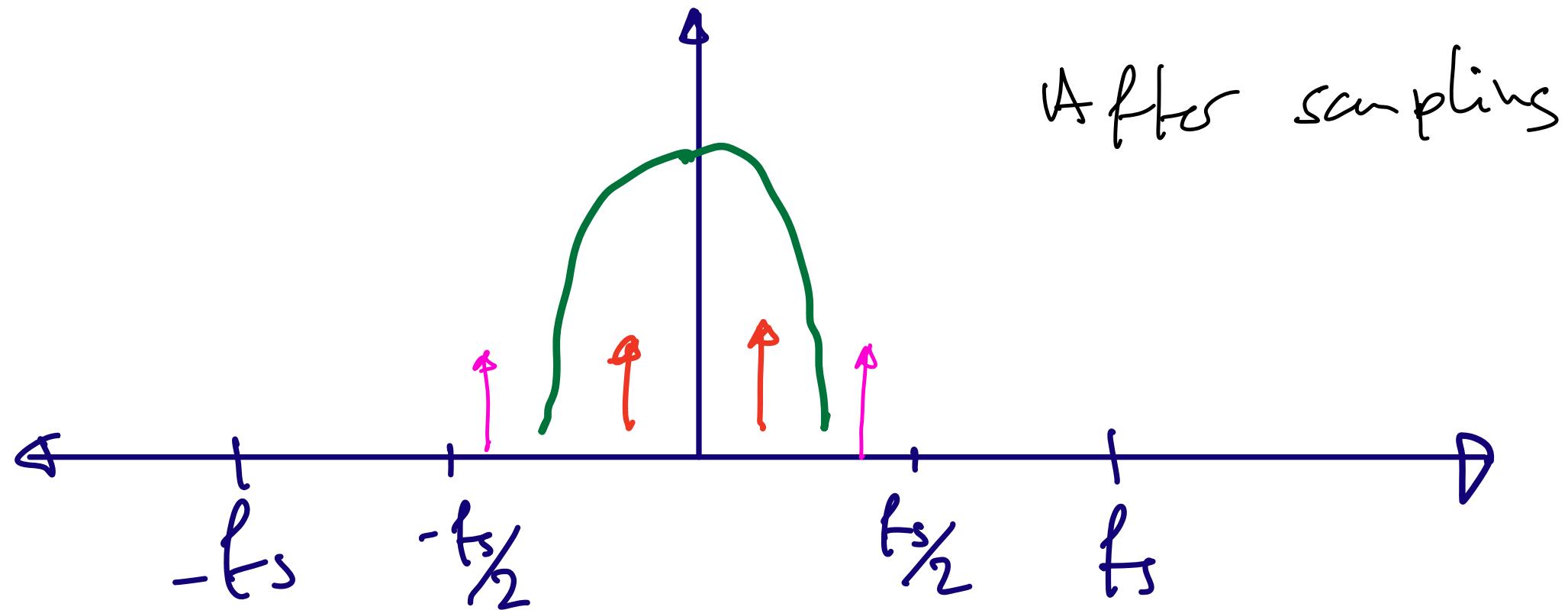
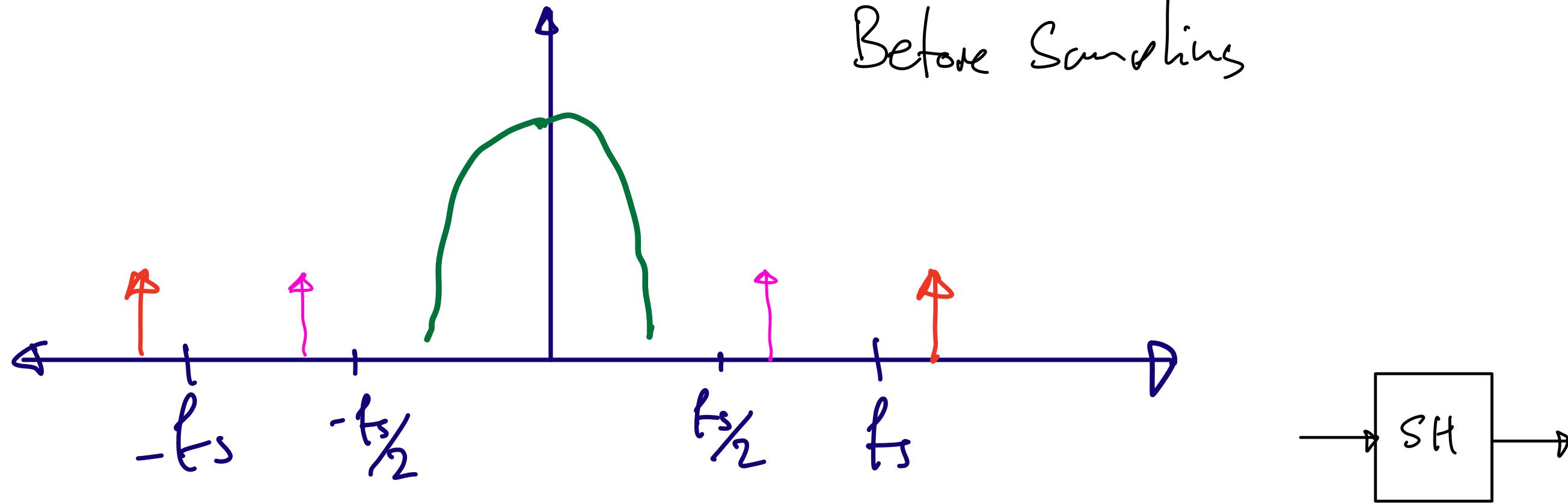
Or $X_s(j\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} X_c \left(j\omega - \frac{jk2\pi}{T} \right)$

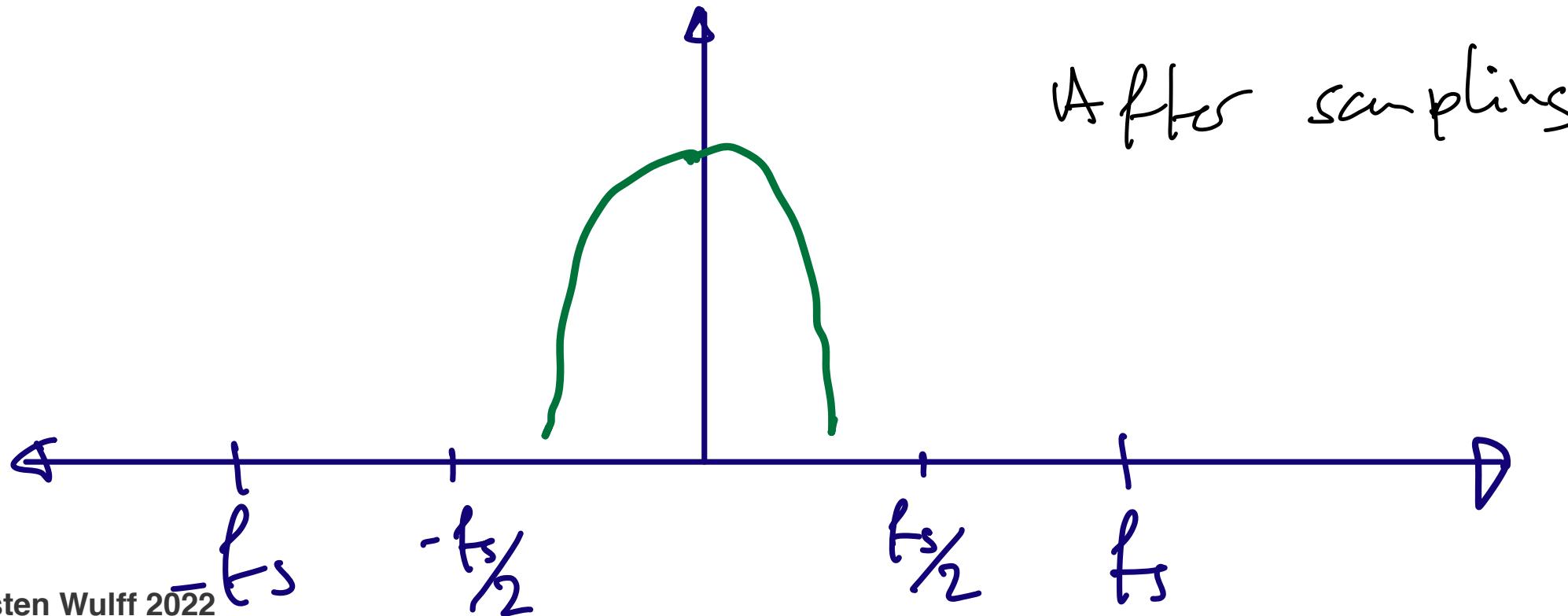
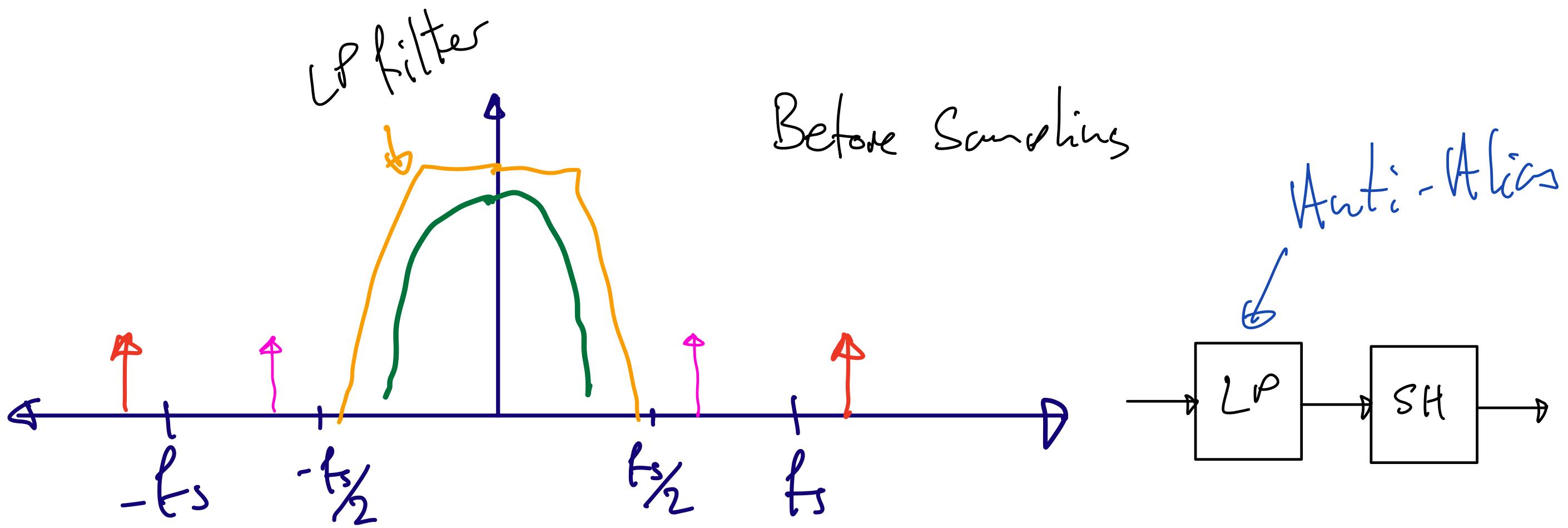
The spectrum of a sampled signal is an infinite sum of frequency shifted spectra

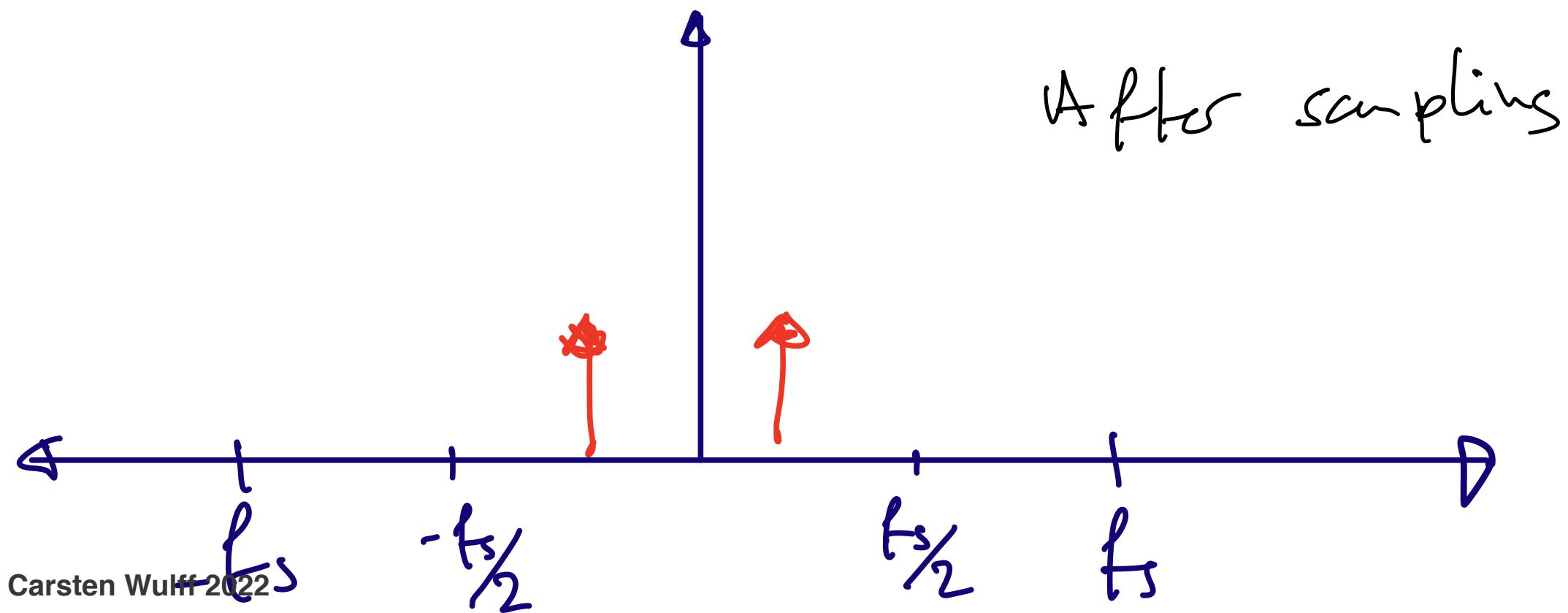
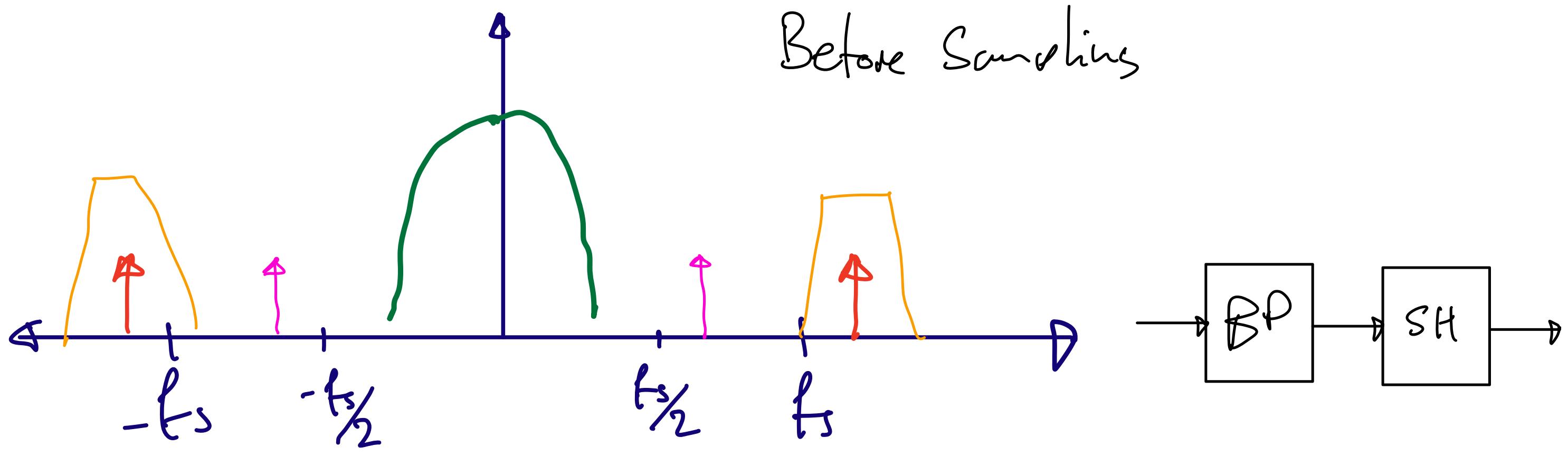
or equivalently

When you sample a signal, then there will be copies of the input spectrum at every $n f_s$

However, if you do an FFT of a sampled signal, then all those infinite spectra will fold down between $0 \rightarrow f_s/2$ or $-f_s/2 \rightarrow f_s/2$ for a complex FFT







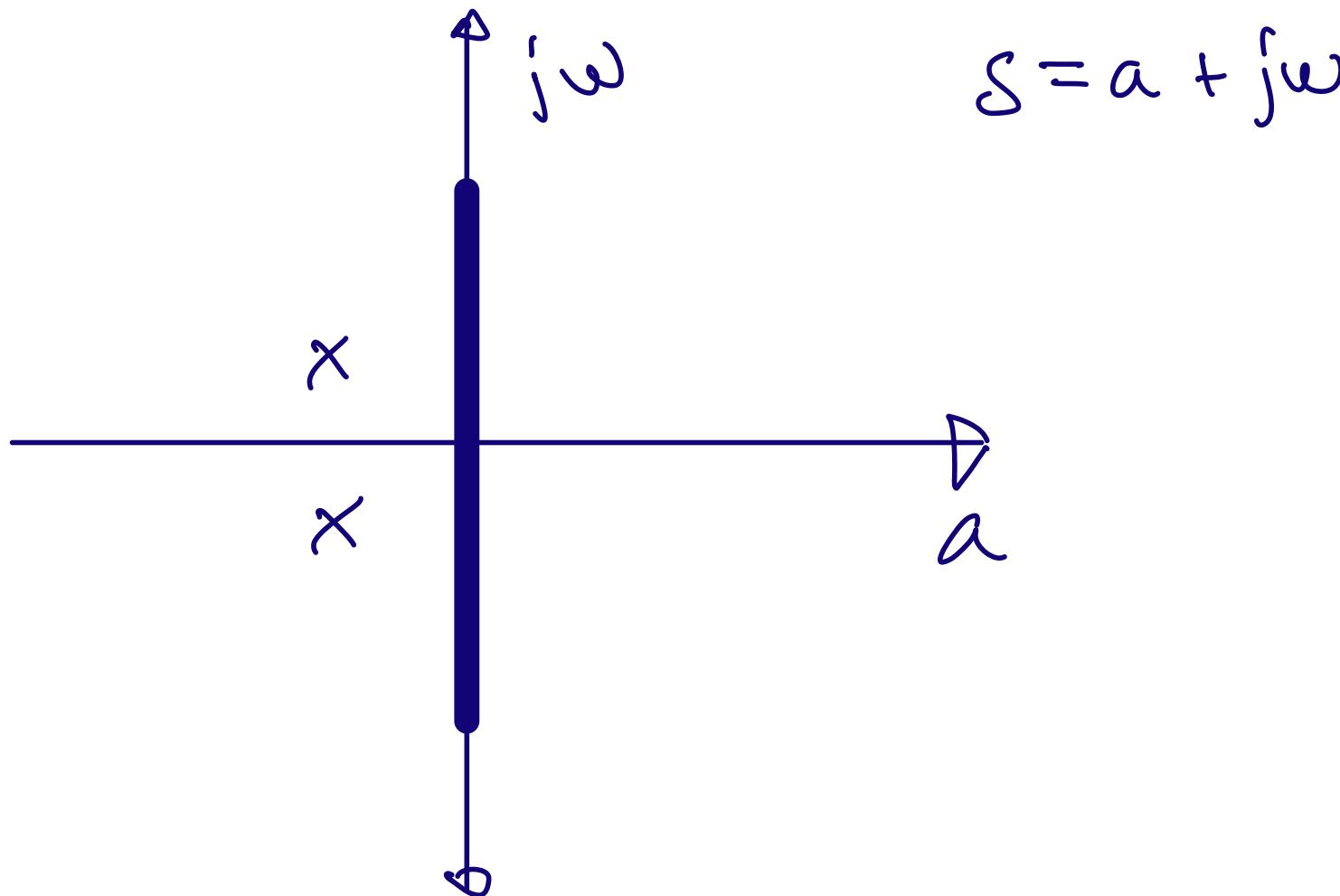
$$X_s(s) = \sum_{n=-\infty}^{\infty} x_c(nT) e^{-snT}$$

$$X_s(z) = \sum_{n=-\infty}^{\infty} x_c[n] z^{-n}$$

For discrete time signal processing we use Z-transform

If you're unfamiliar with the Z-transform, read the book or search <https://en.wikipedia.org/wiki/Z-transform>

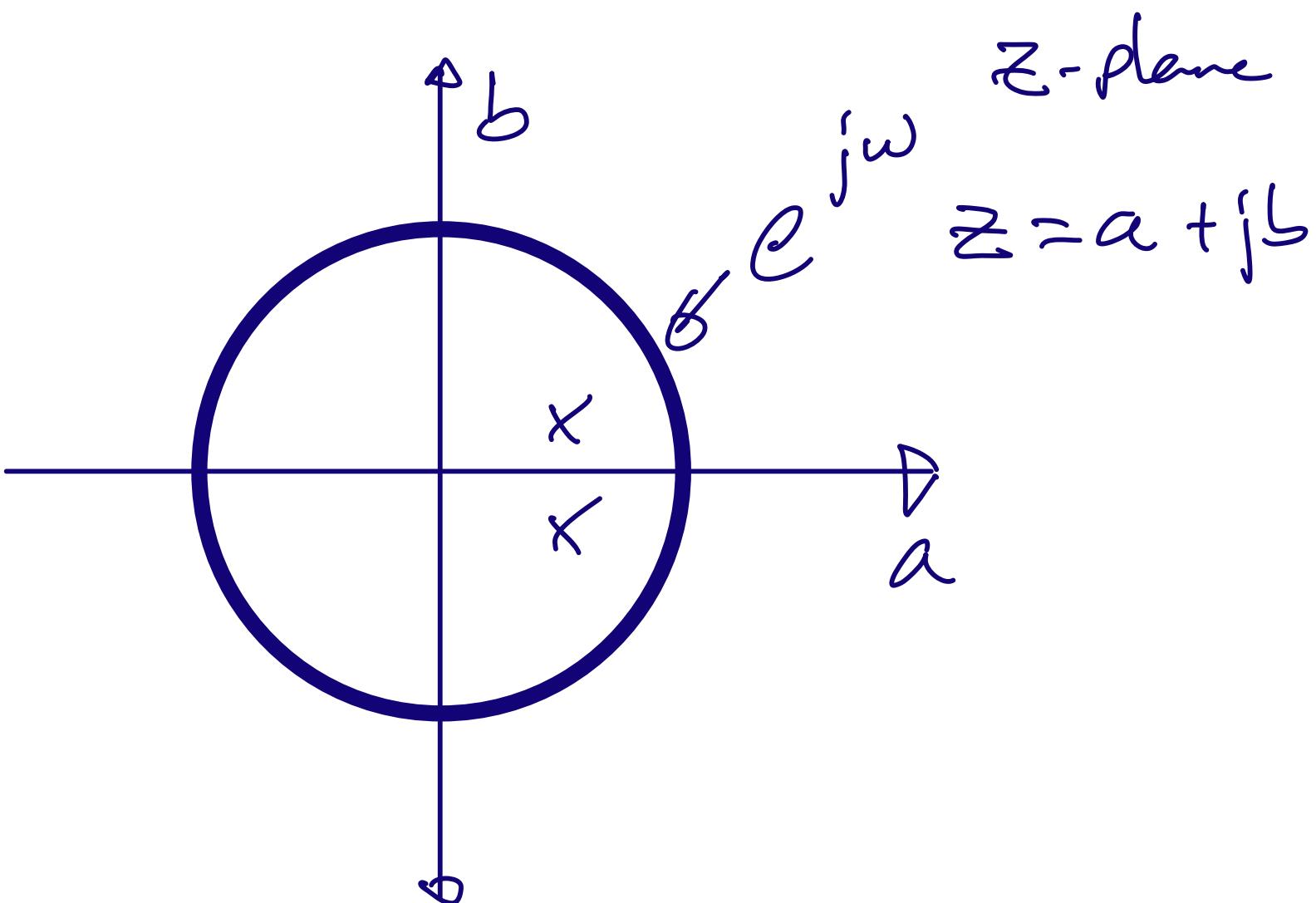
Pole-Zero plots



If you're not comfortable with pole/zero plots, have a look at

[What does the Laplace Transform really tell us](#)

Z-domain



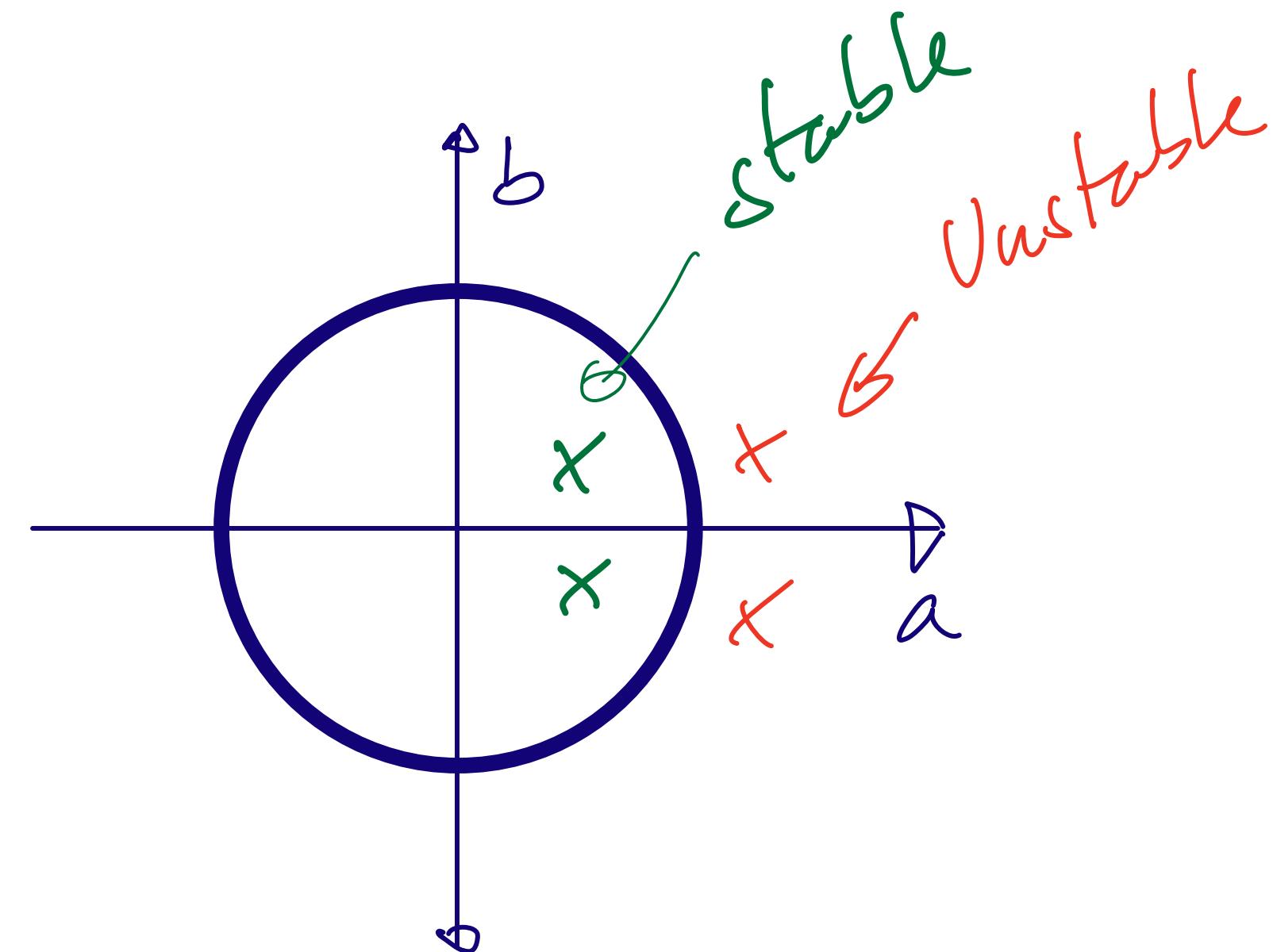
Spectra repeat every 2π (every f_s)

Bi-linear transform

$$s = \frac{z - 1}{z + 1}$$

Warning: First-order approximation https://en.wikipedia.org/wiki/Bilinear_transform

First order filter



$$y[n + 1] = bx[n] + ay[n] \Rightarrow yz = bx + ay$$

$$y[n] = bx[n - 1] + ay[n - 1] \Rightarrow y = bxz^{-1} + ayz^{-1}$$

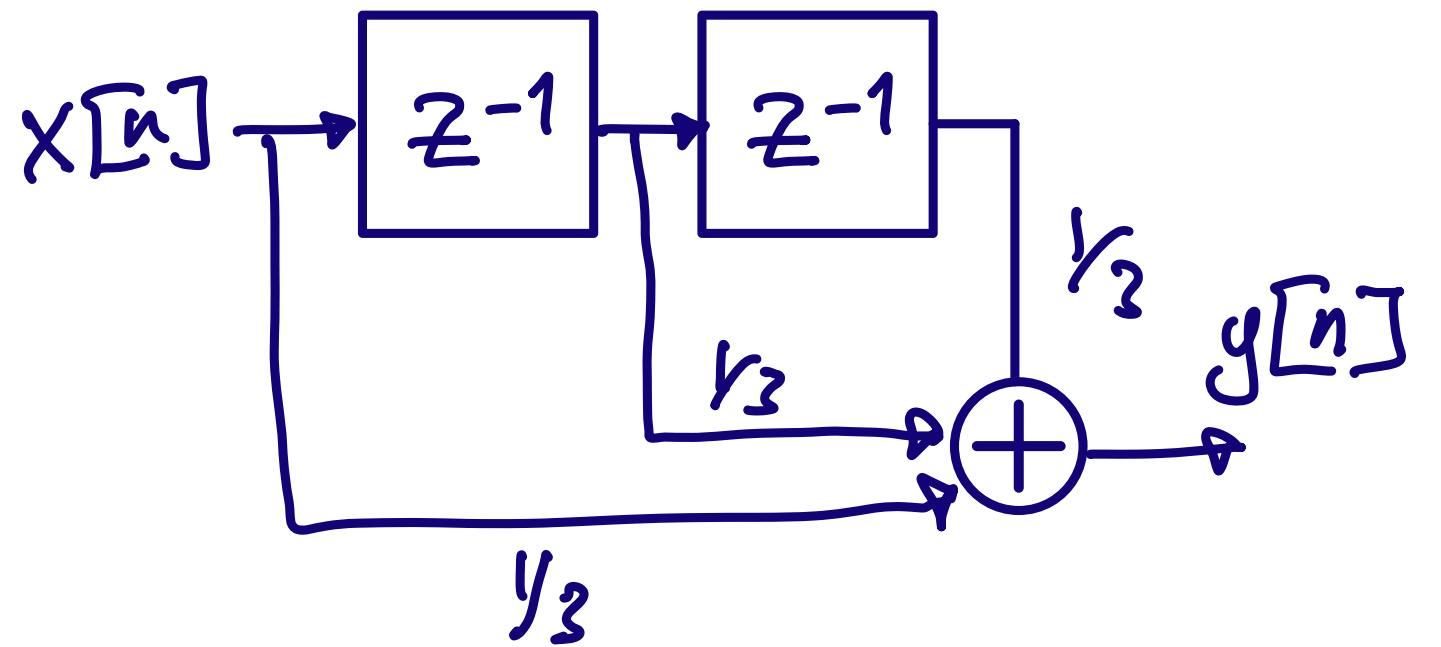
$$H(z) = \frac{b}{z - a}$$

Infinite-impulse response (IIR)

$$h[n] = \begin{cases} k & \text{if } n < 1 \\ a^{n-1}b + a^n k & \text{if } n \geq 1 \end{cases}$$

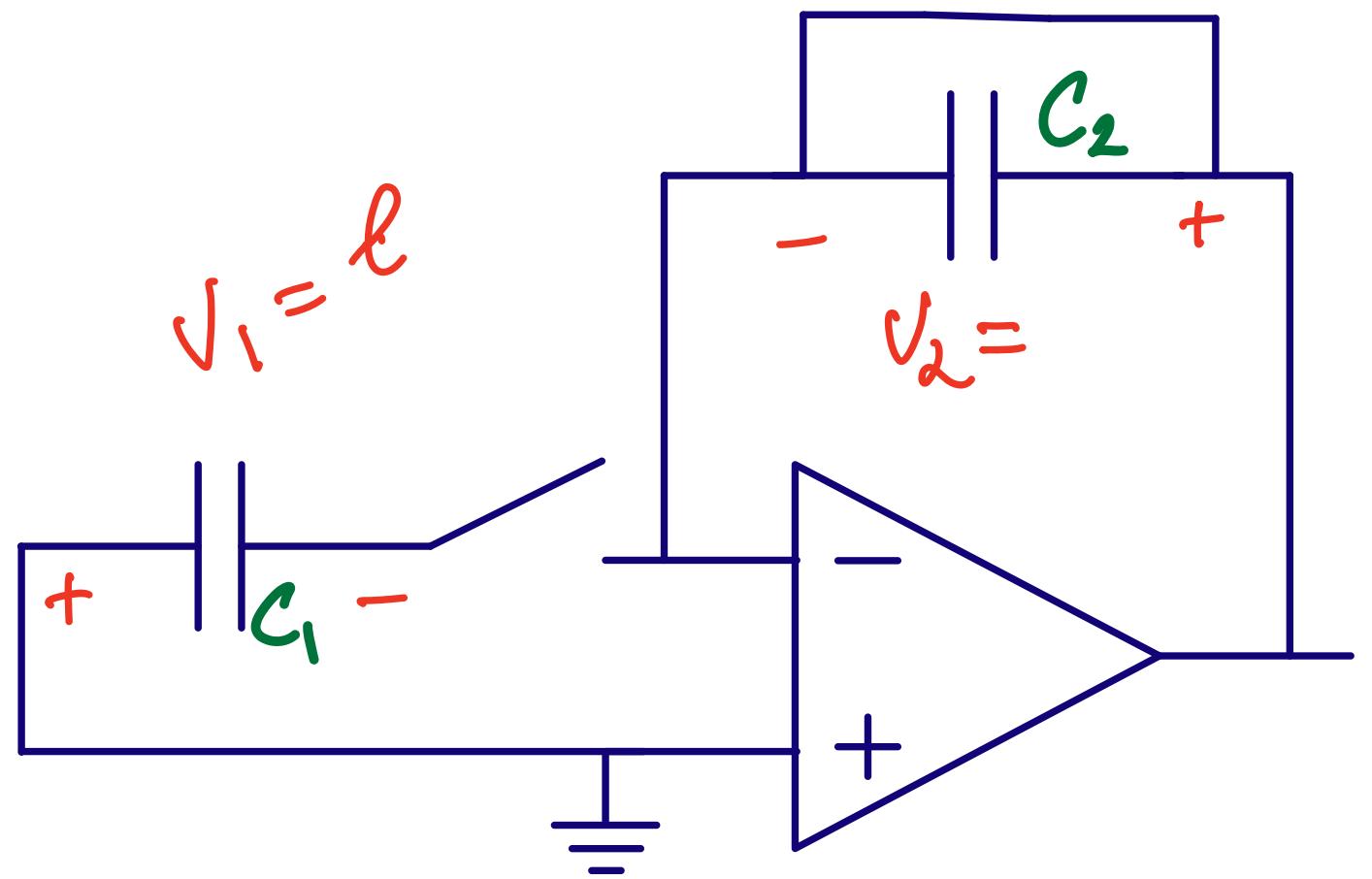
Head's up: Fig 13.12 in AIC is wrong

Finite-impulse response(FIR)



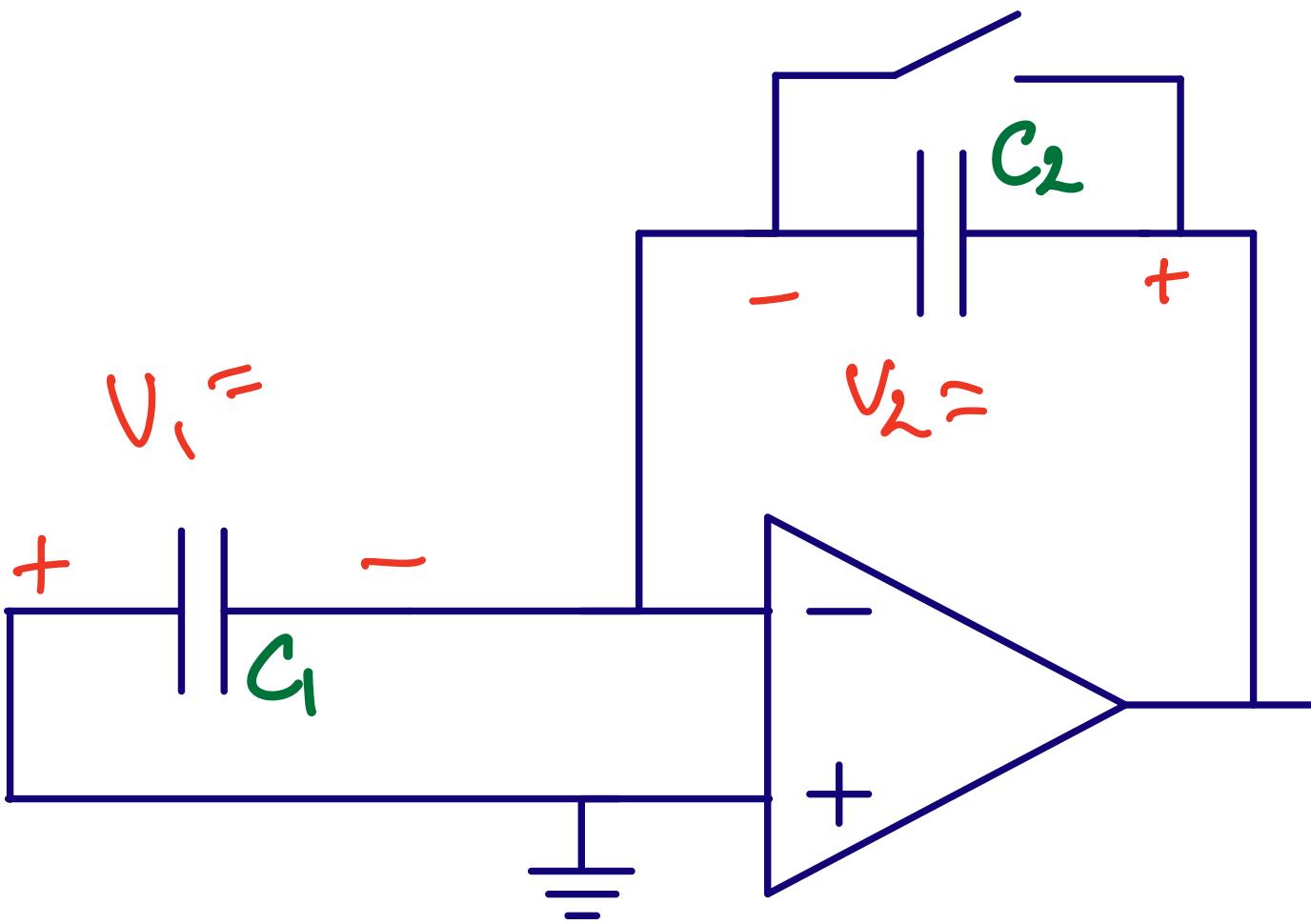
$$H(z) = \frac{1}{3} \sum_{i=0}^2 z^{-i}$$

Switched-Capacitor



$$Q_1 =$$

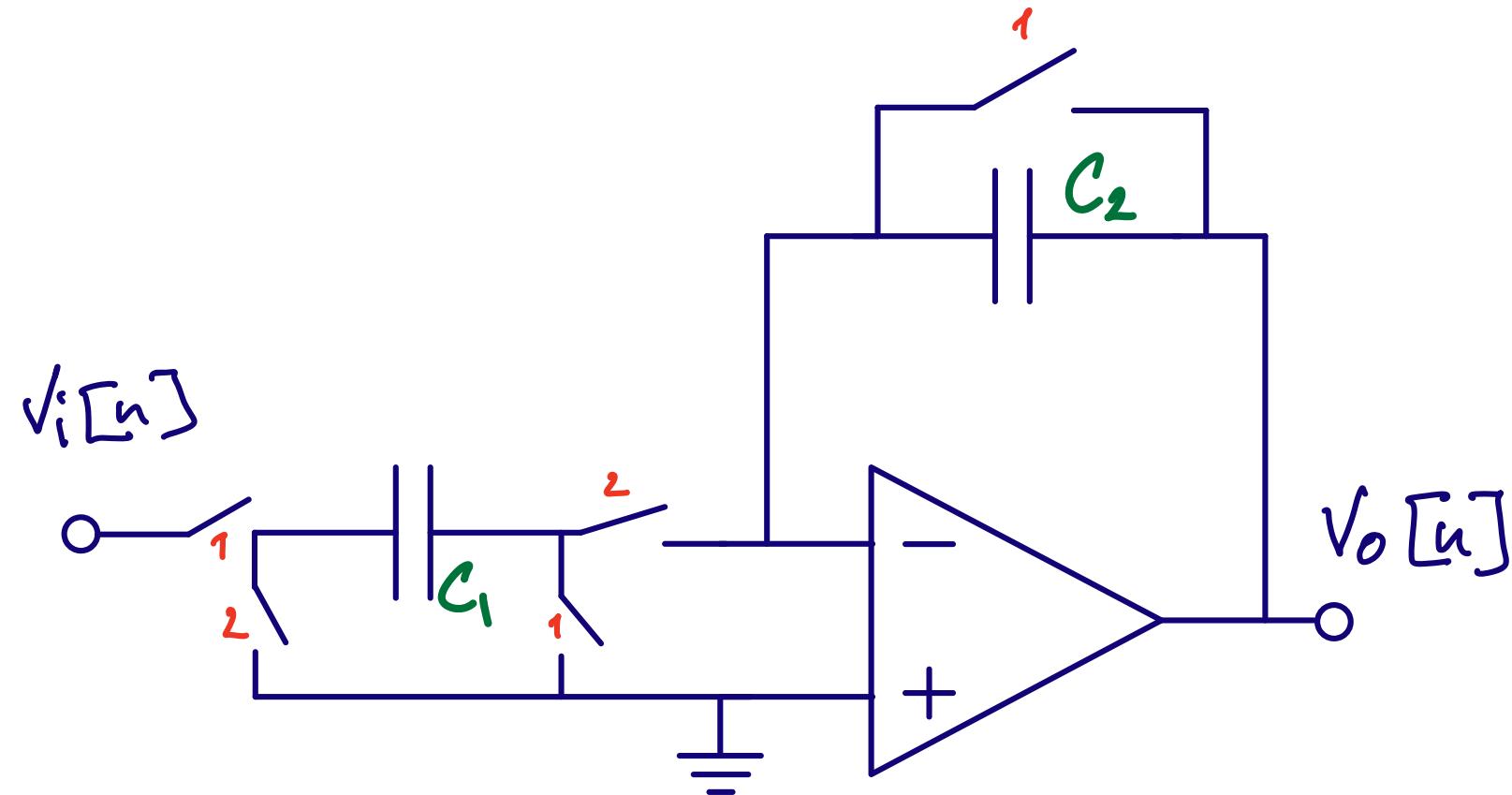
$$Q_2 =$$



$$Q_1 =$$

$$Q_2 =$$

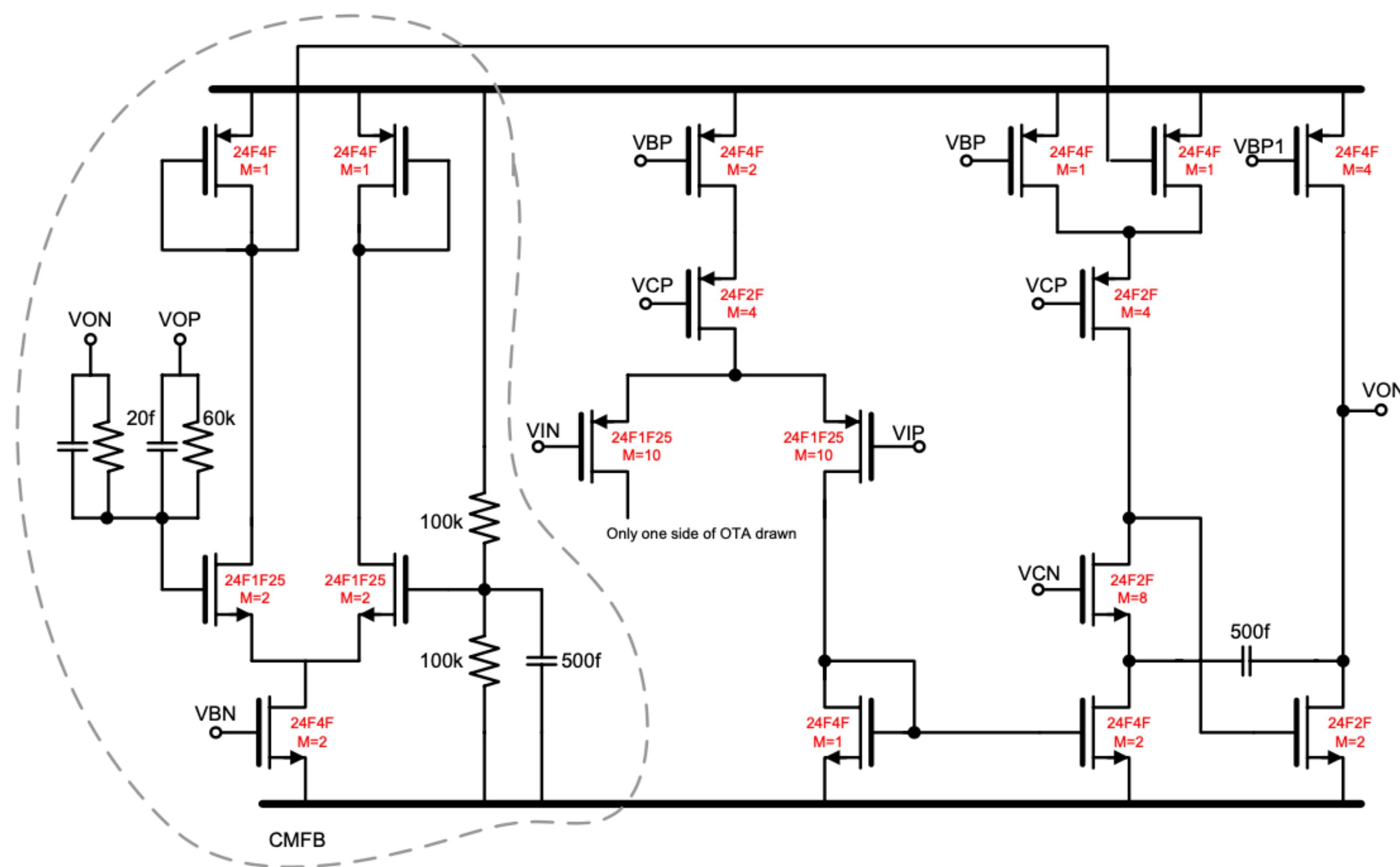
What do we need?

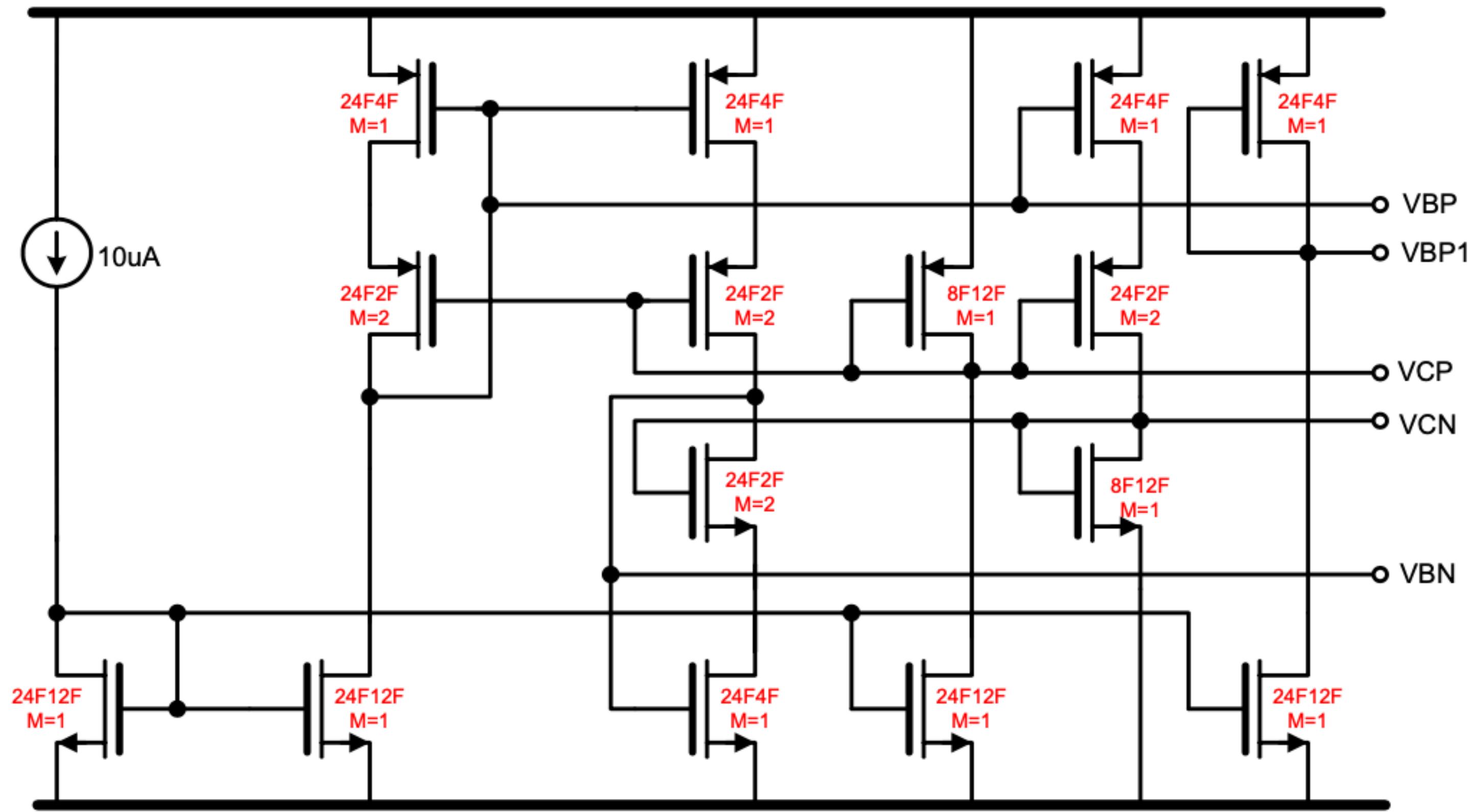


O

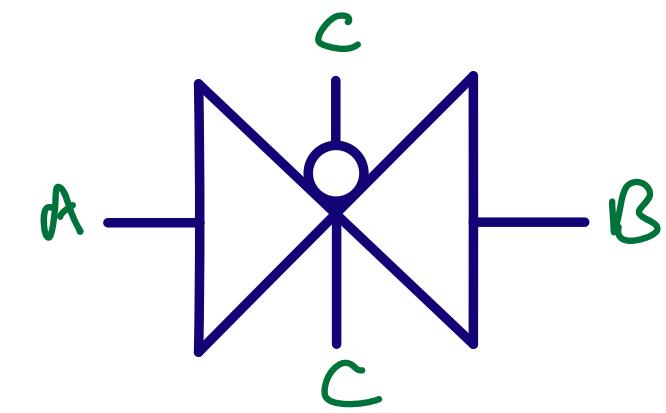
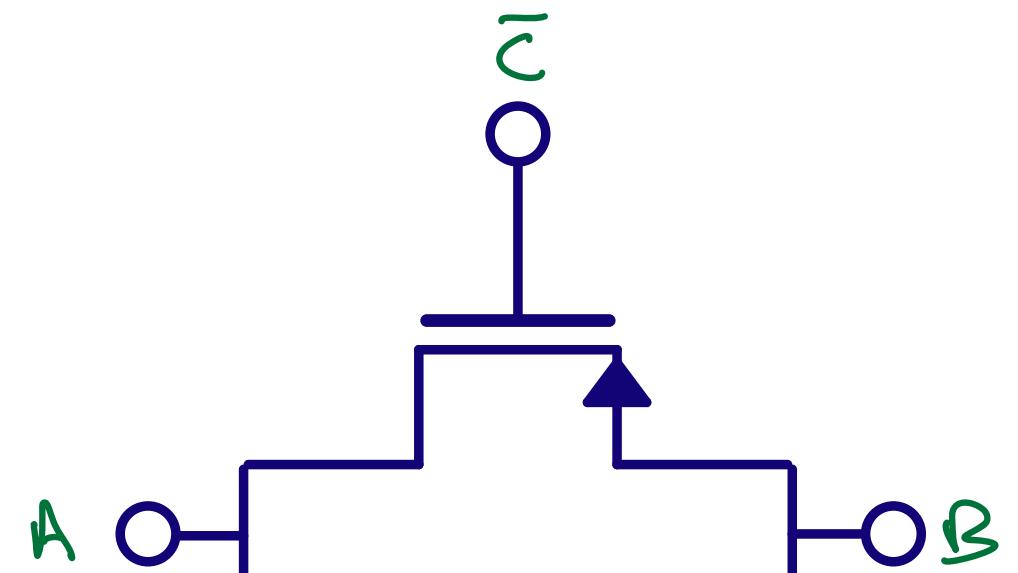
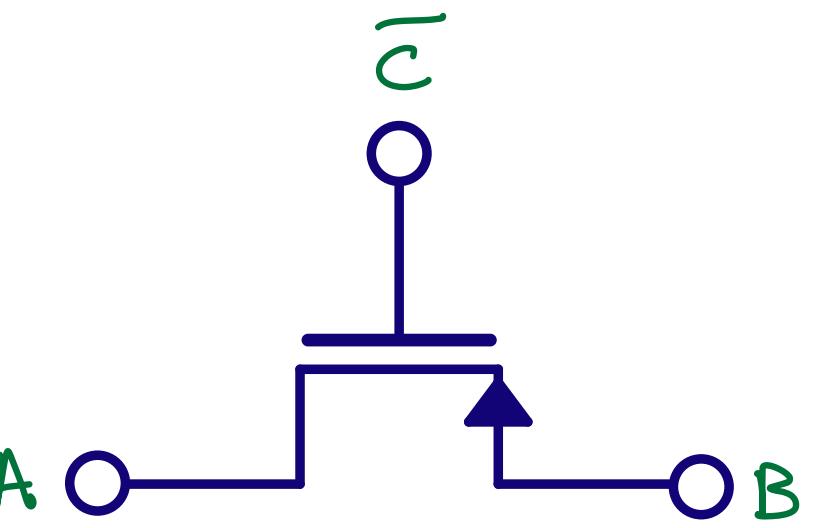
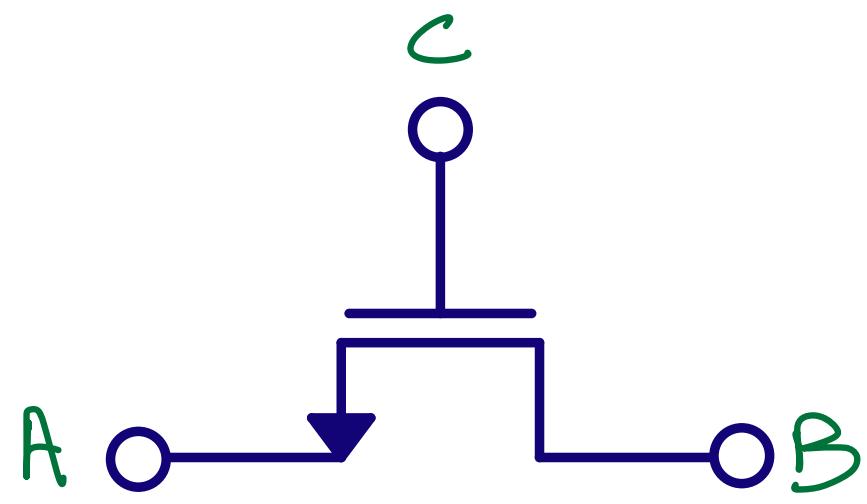
TU

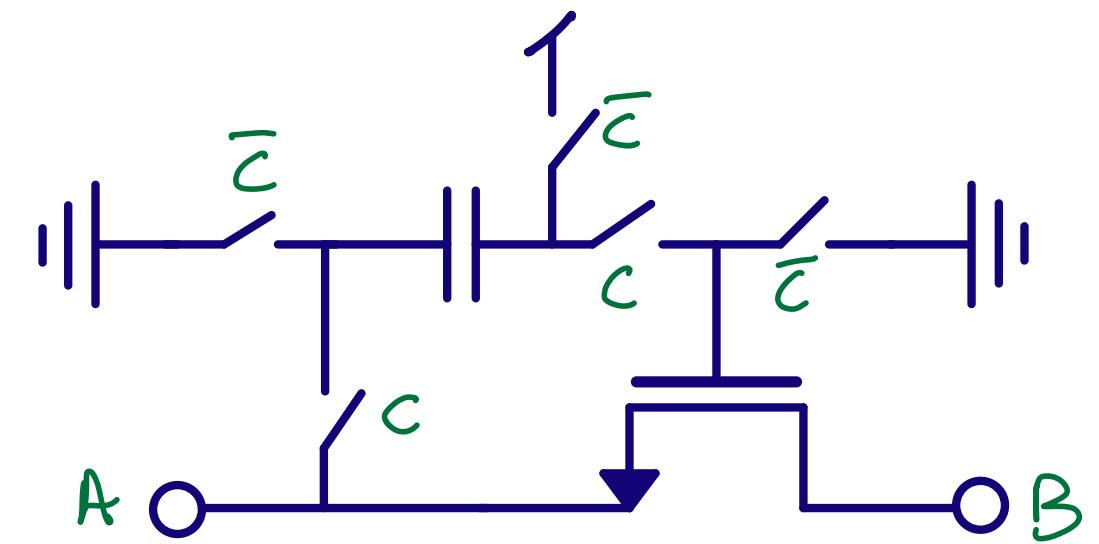
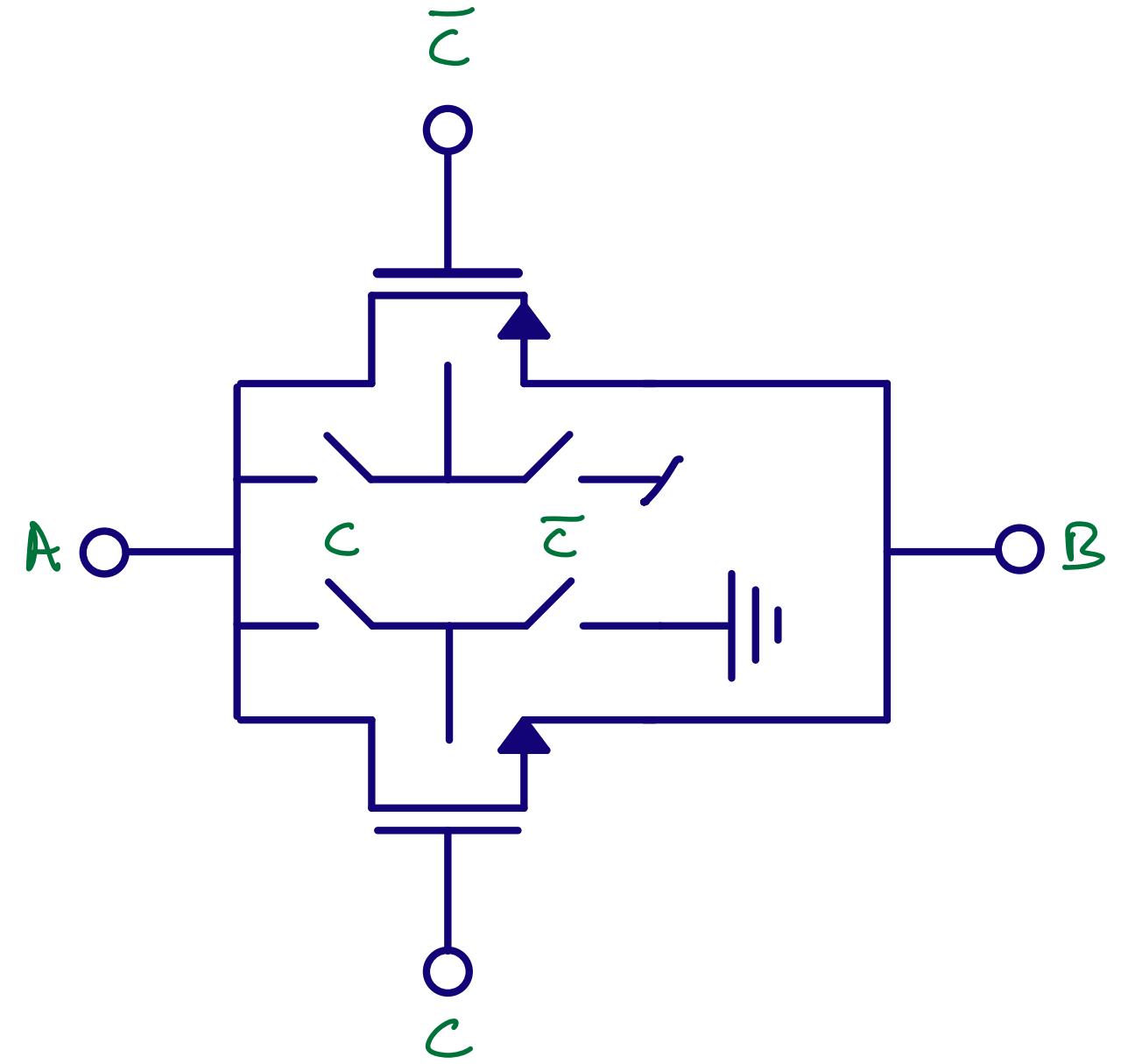
A

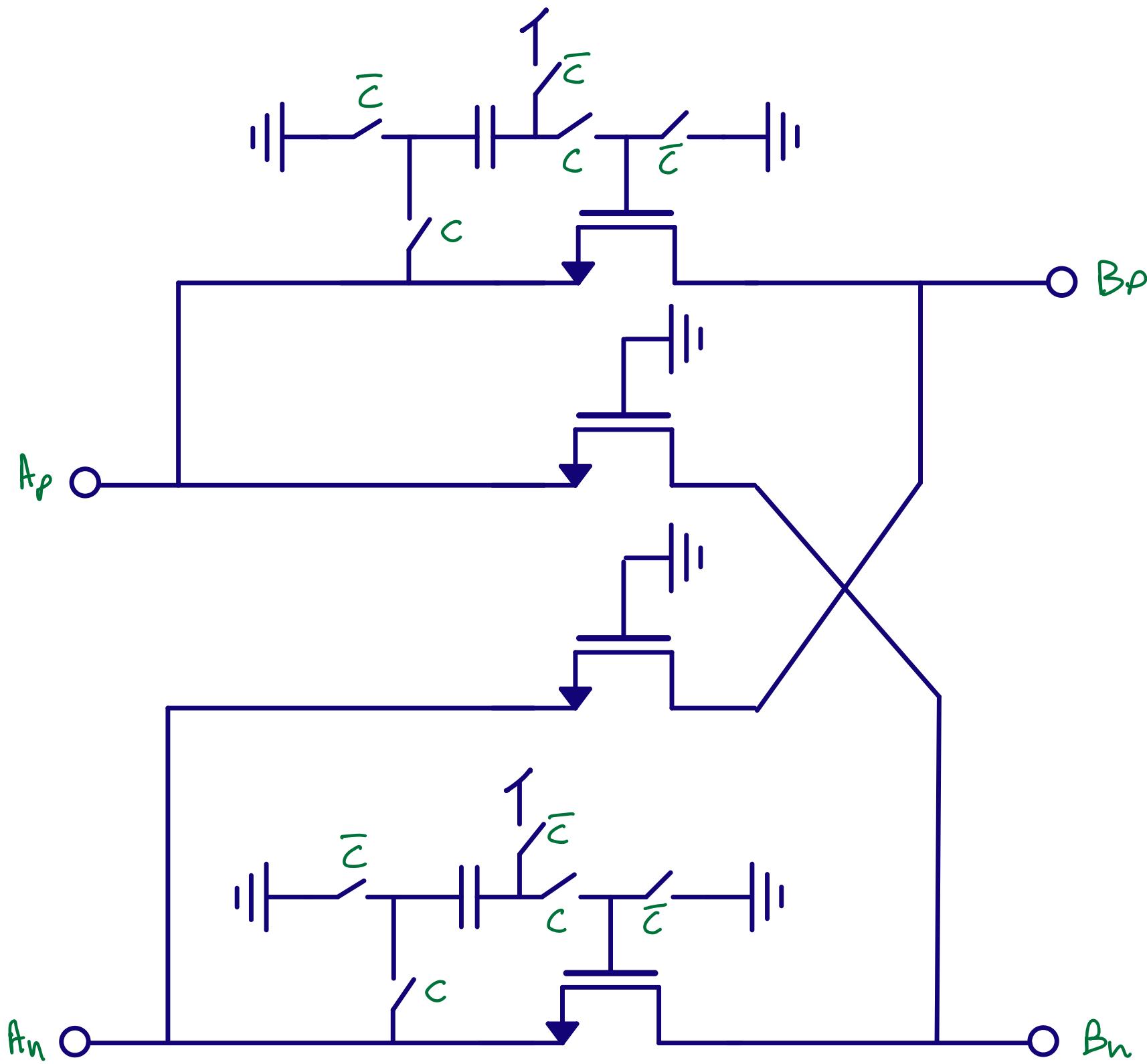




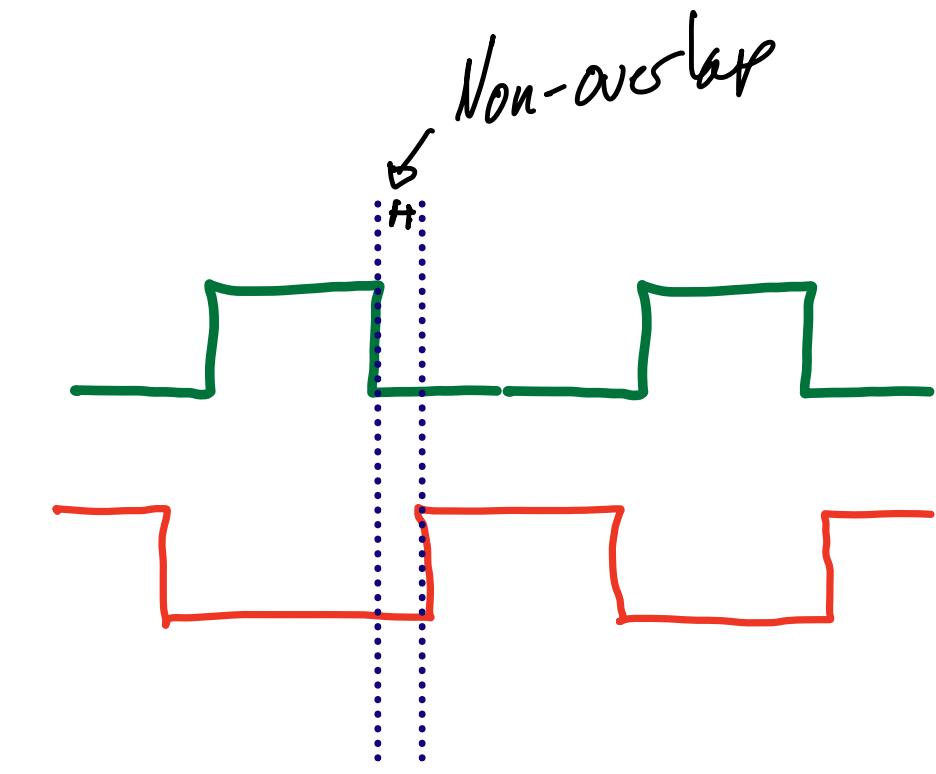
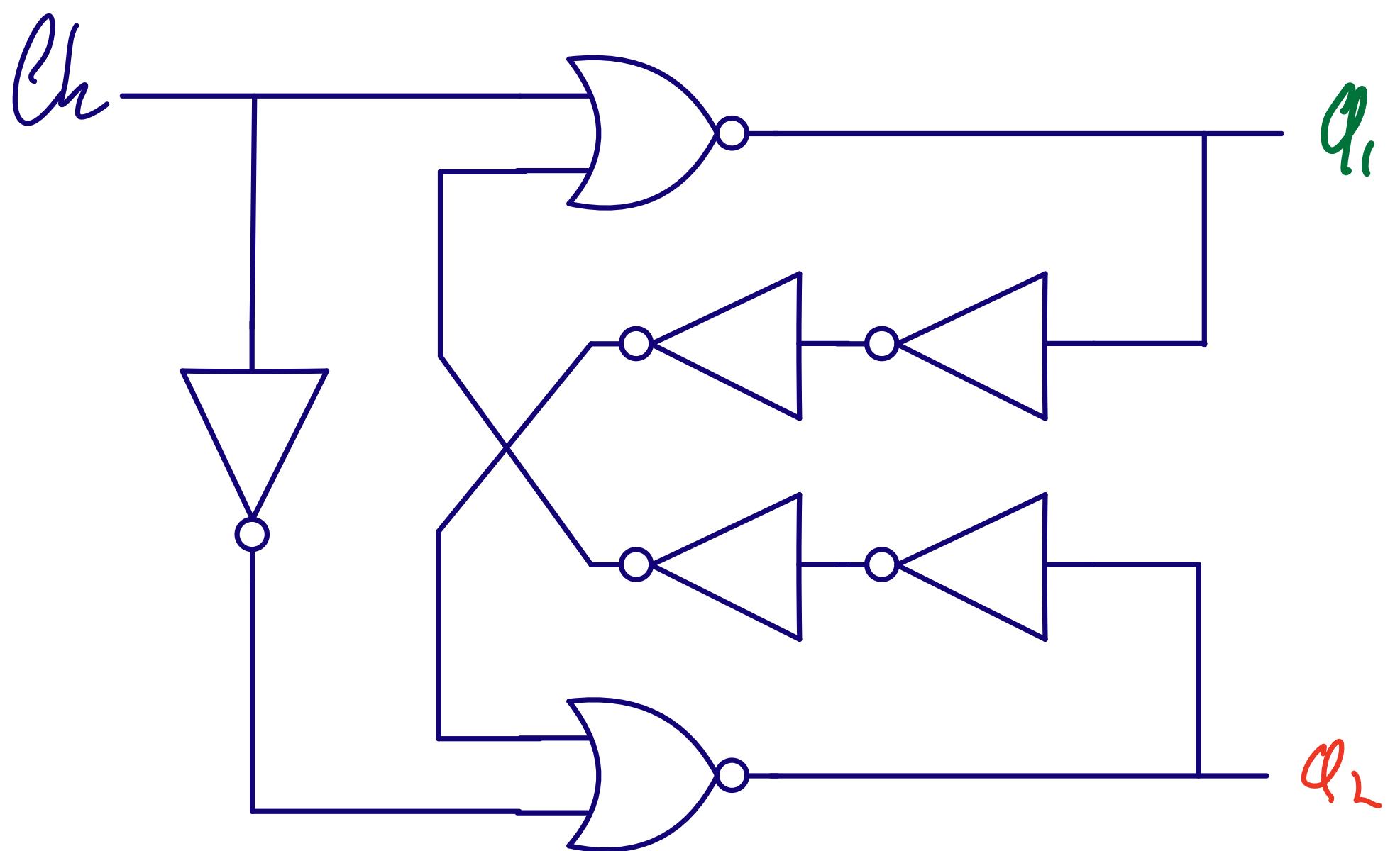
Switches

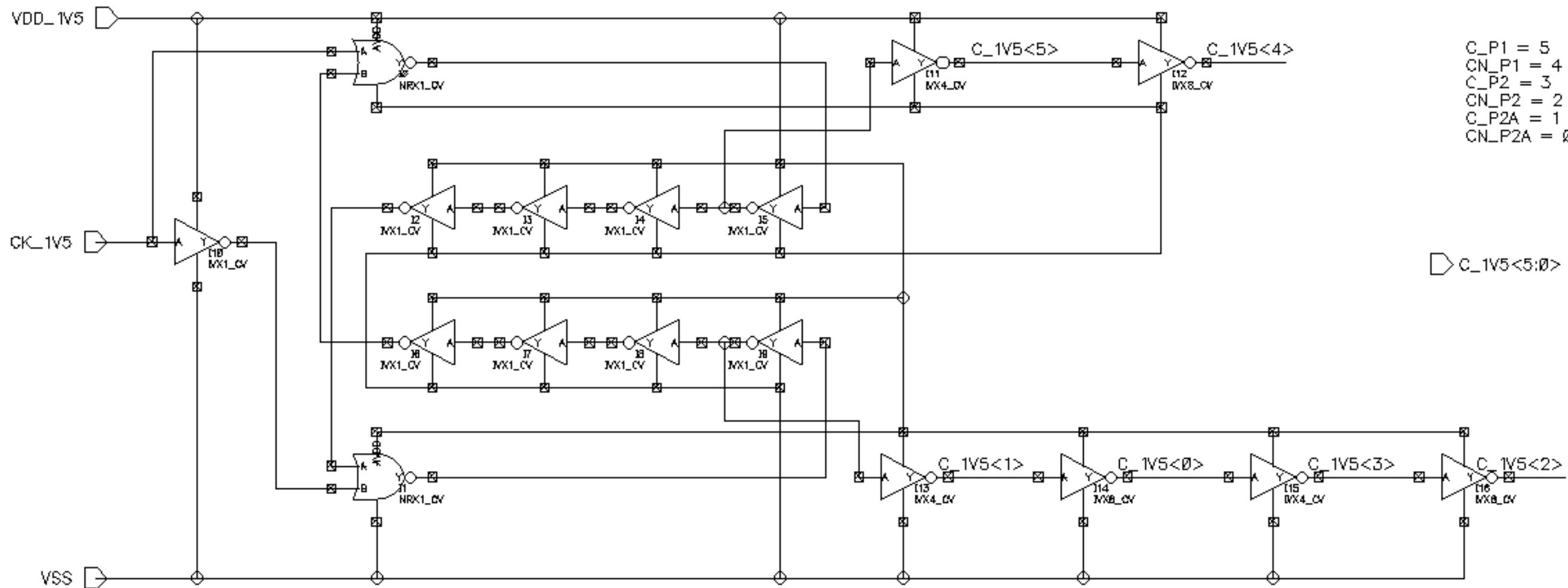






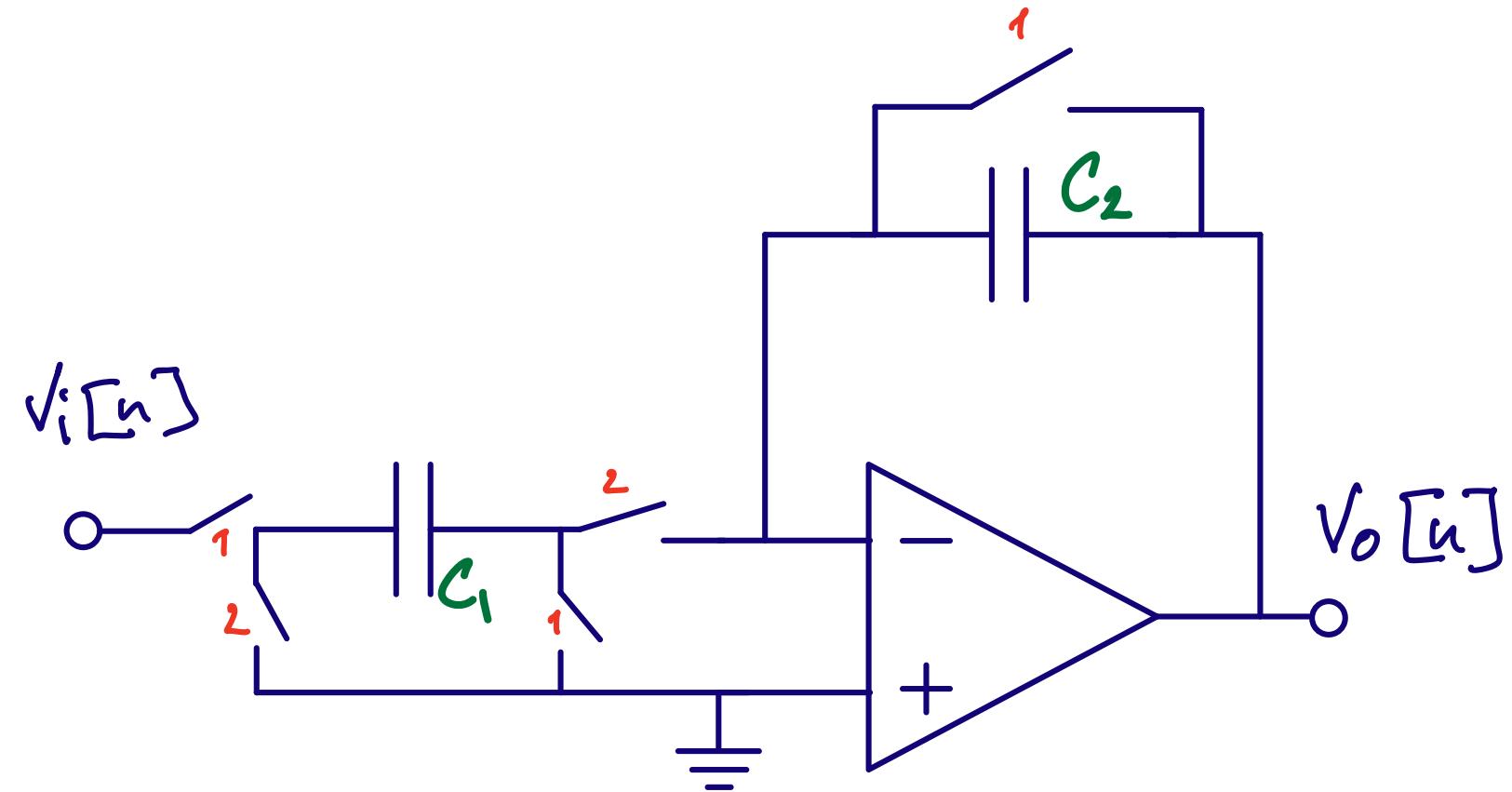
Non-overlapping clocks



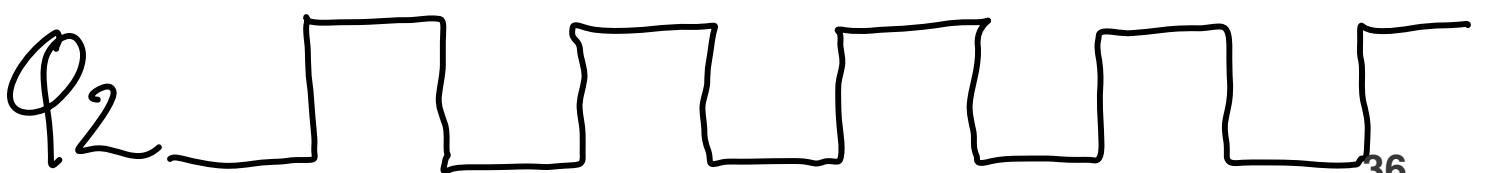
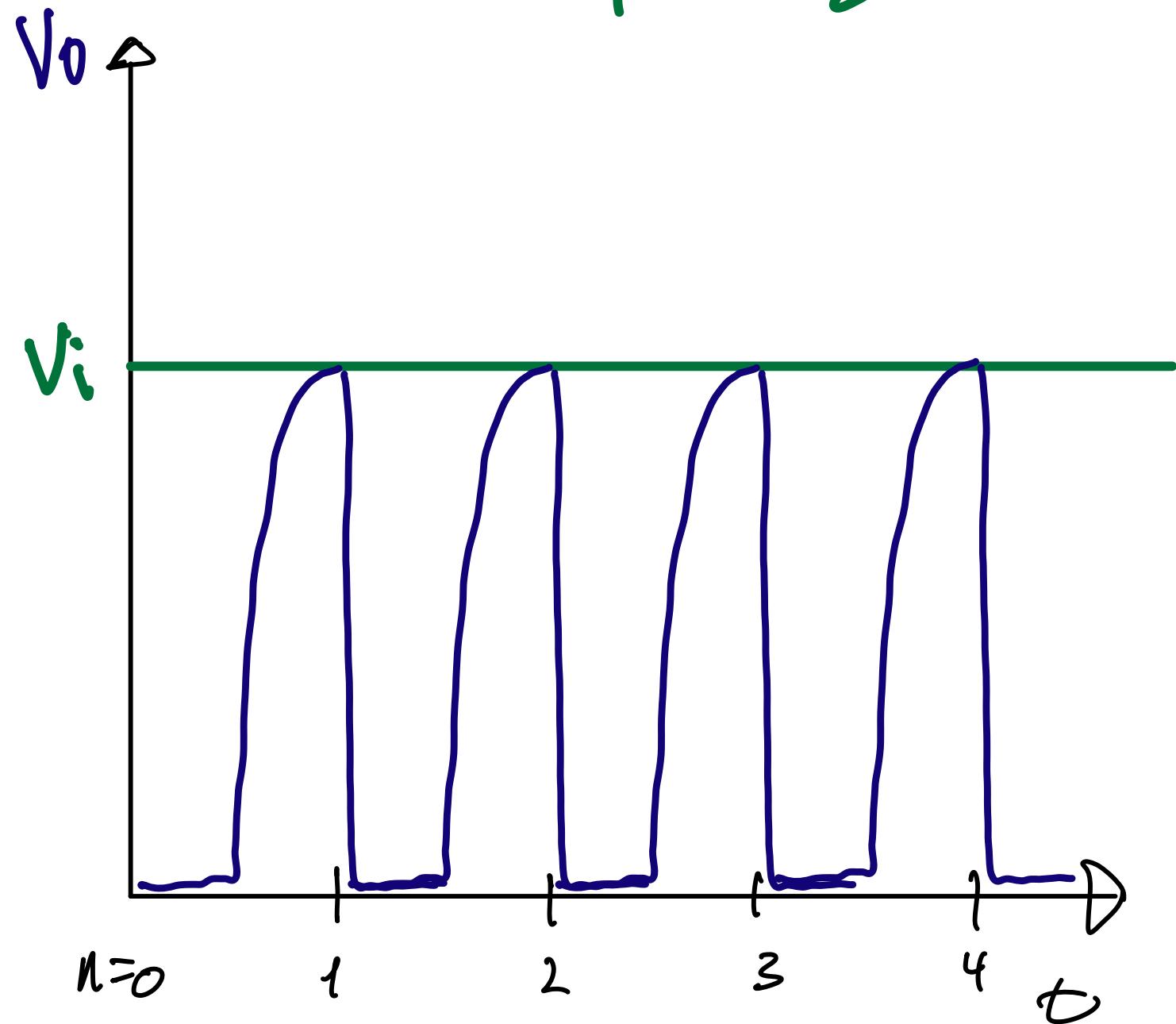
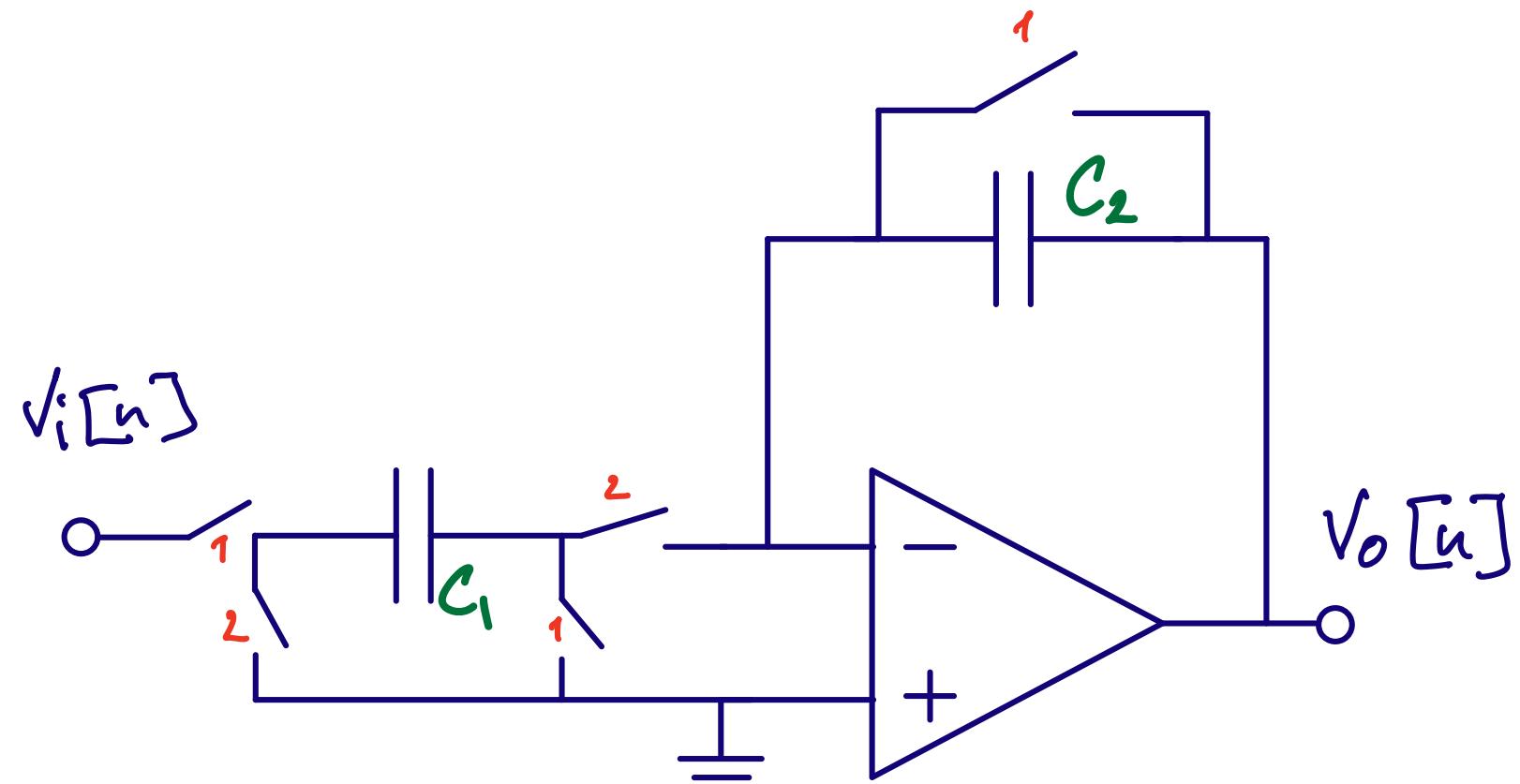


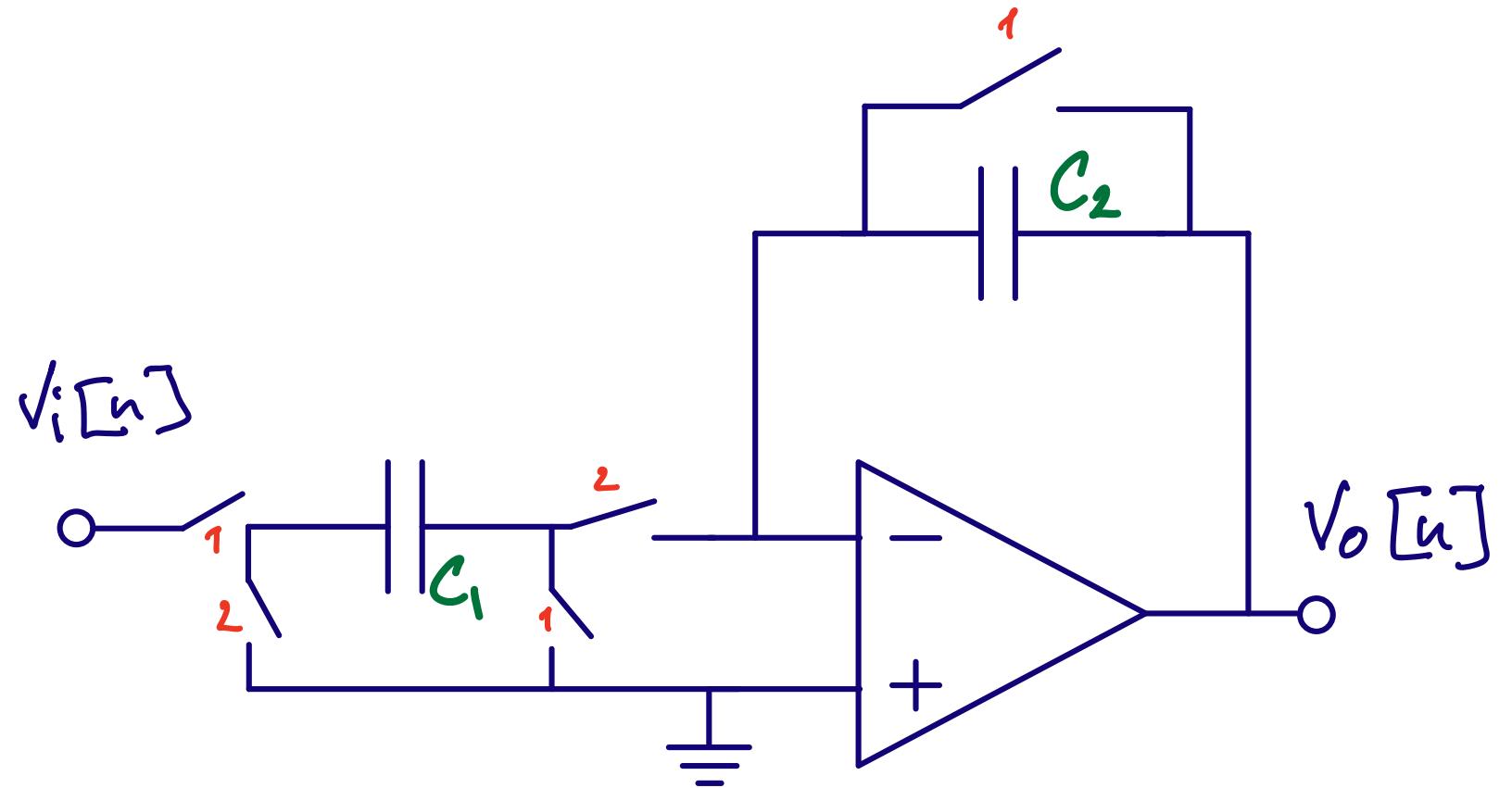
SUN_WF_NOVL	
Designer	wulff
Created	Feb 5 21:11:46 2022
Modified	Feb 5 21:30:16 2022
copyright NTNU	
SUN_WULFF_GF130N	Path
/home/wulff/pro/sun/sun_wulff_gf130n/design/SUN_WULFF_GF130N/SUN_WF_NOVL	

Principles



$$C_1 = C_2$$

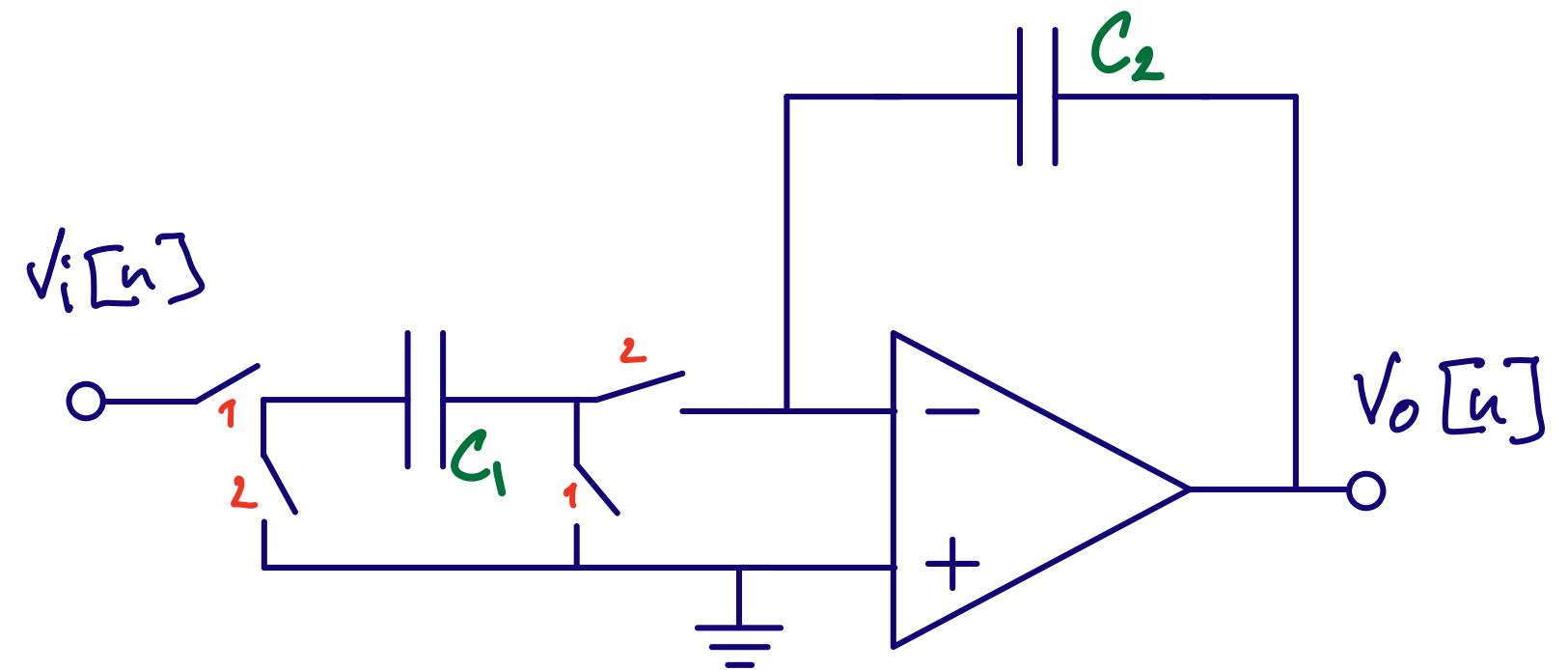


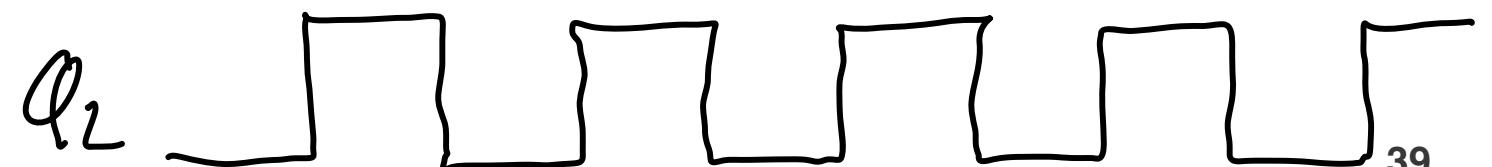
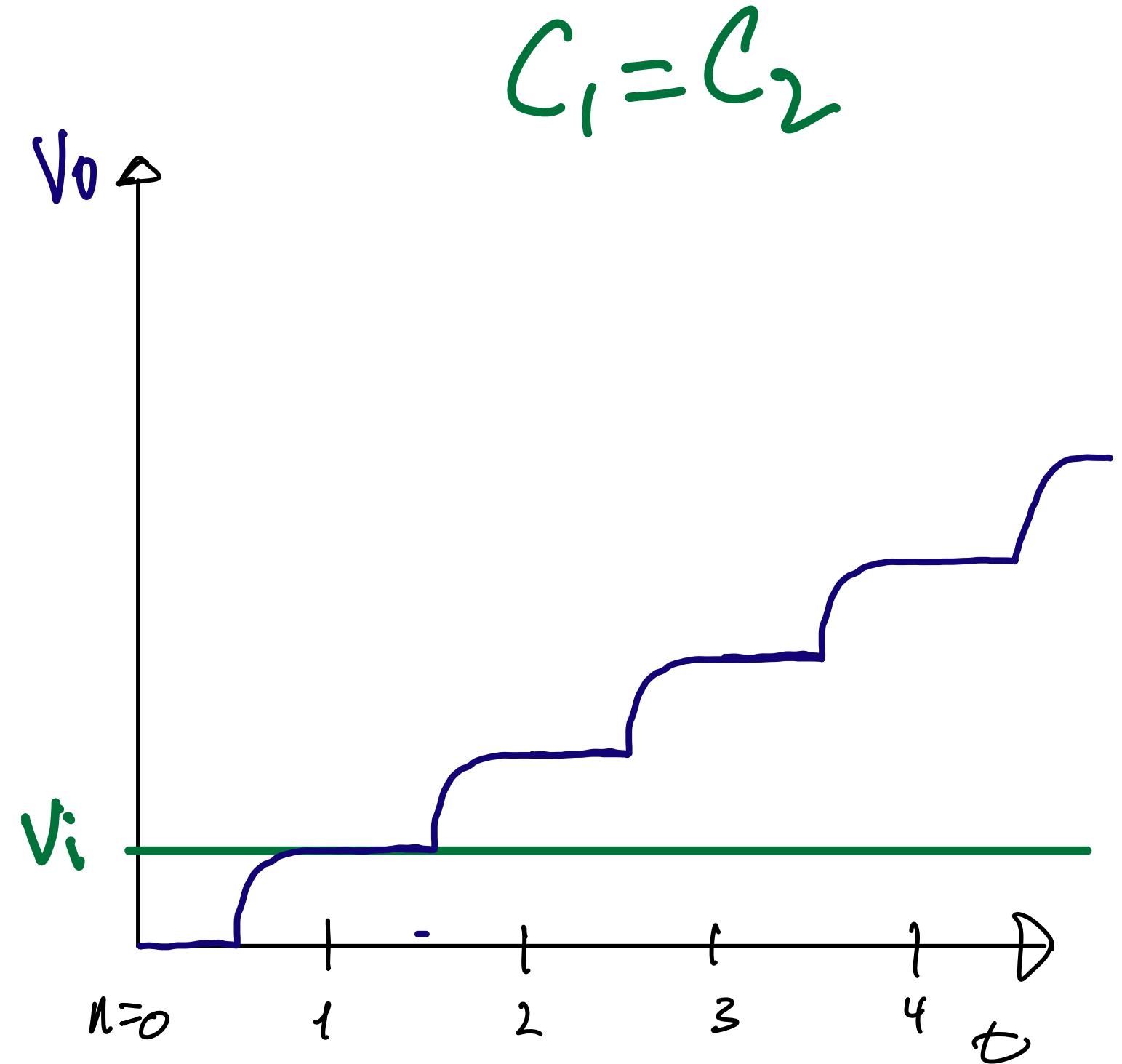
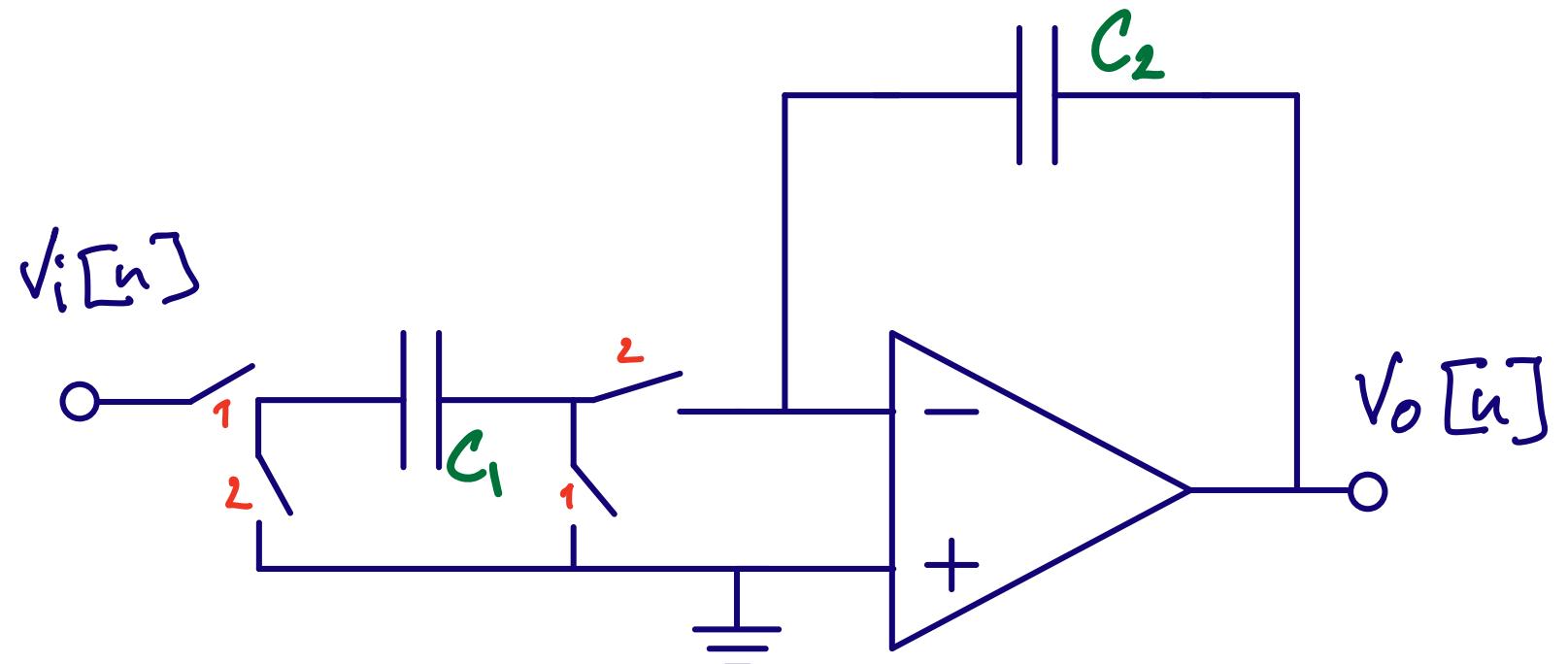


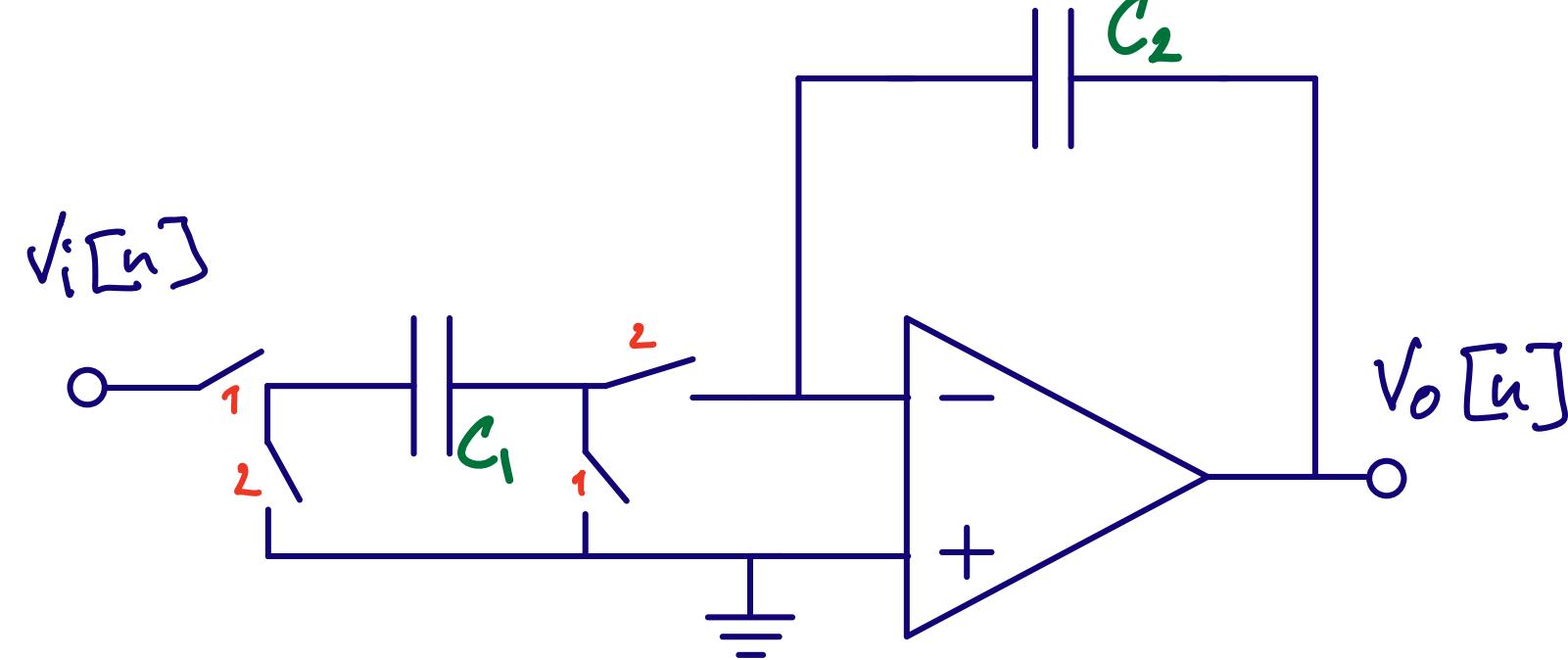
$$V_o[n+1] = \frac{C_1}{C_2} V_i[n]$$

$$V_o z = \frac{C_1}{C_2} V_i$$

$$\frac{V_o}{V_i} = H(z) = \frac{C_1}{C_2} z^{-1}$$



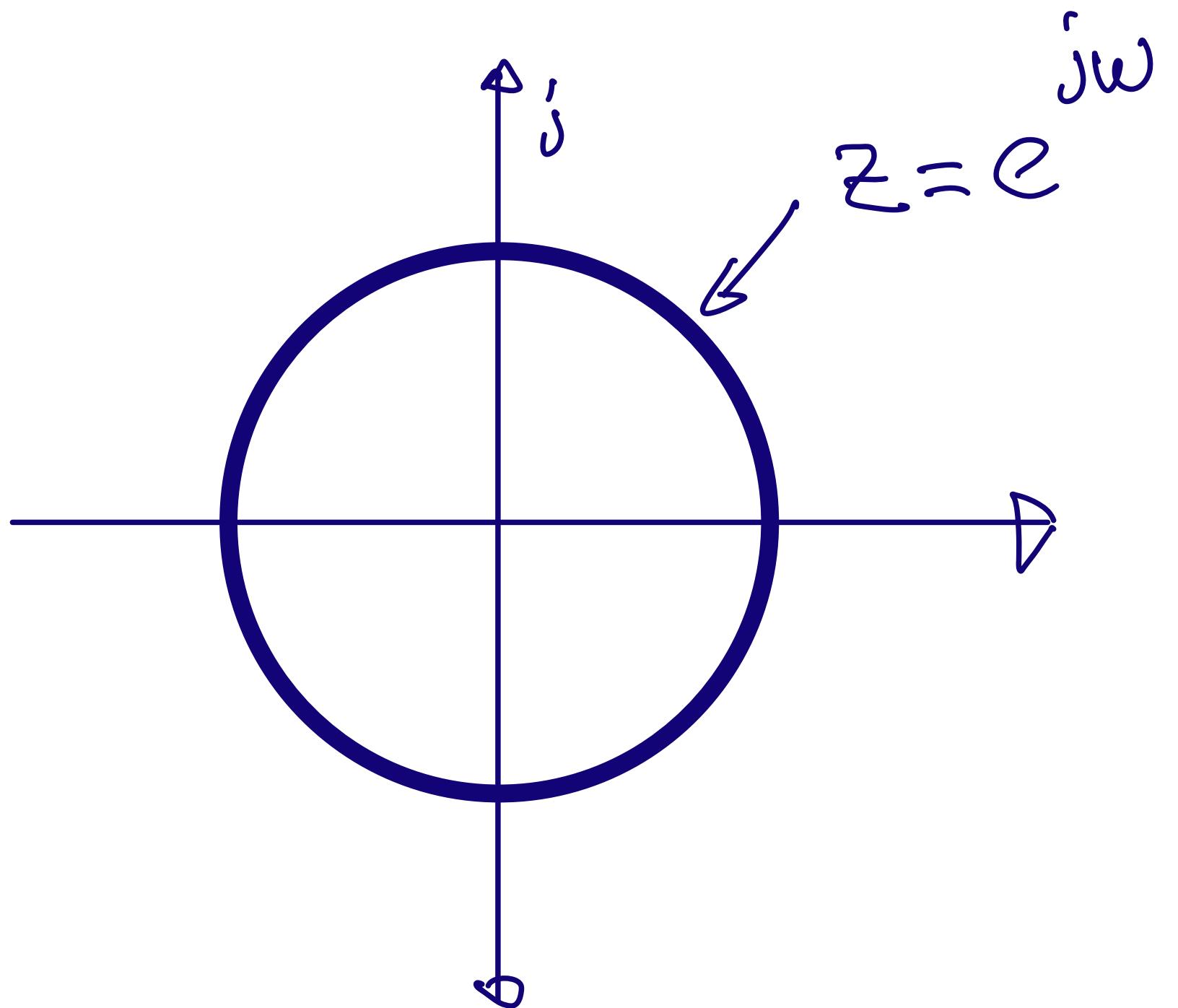




$$V_o[n] = V_o[n - 1] + \frac{C_1}{C_2} V_i[n - 1]$$

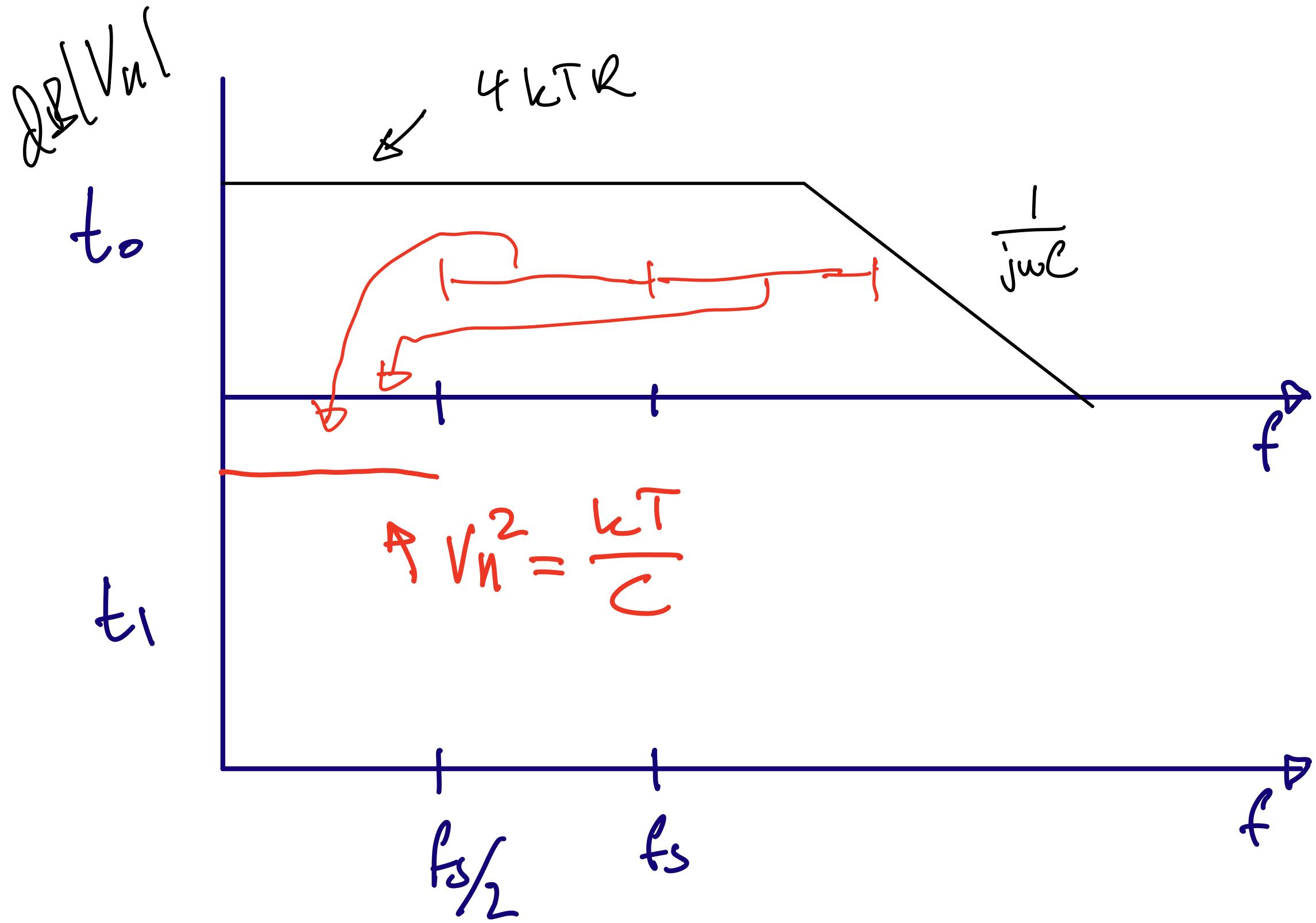
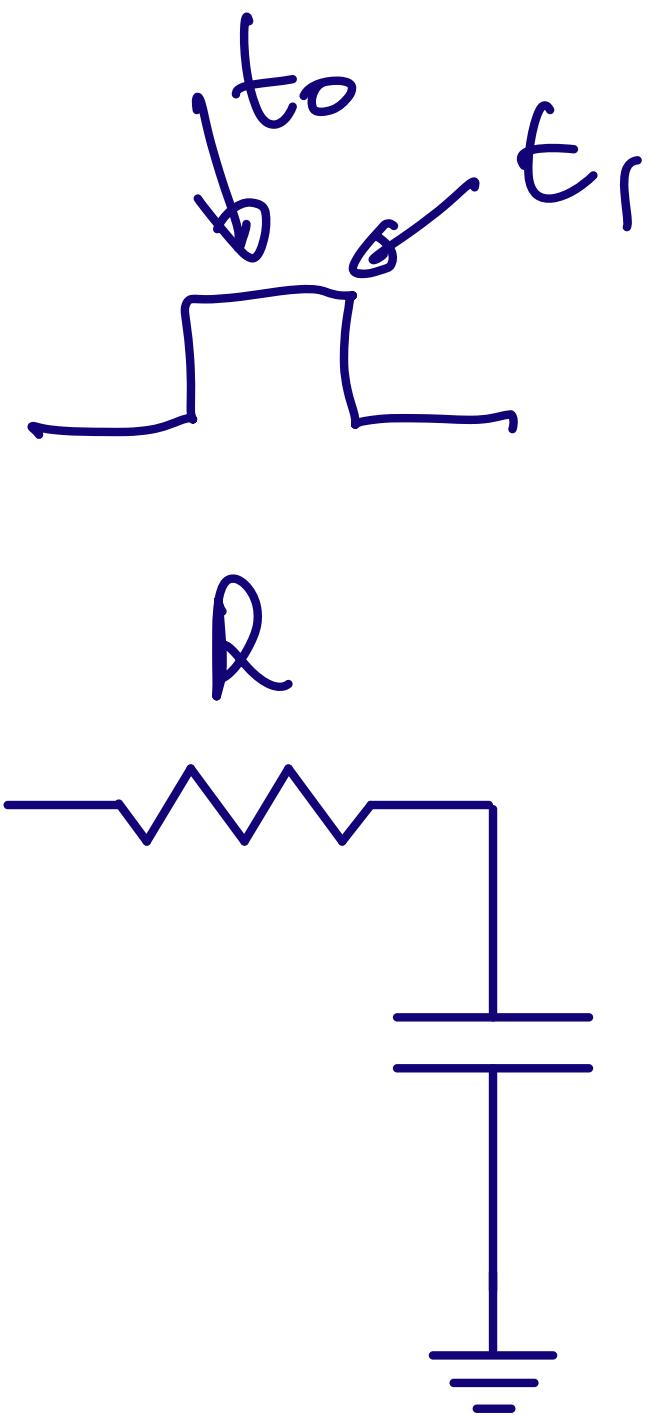
$$V_o - z^{-1}V_o = \frac{C_1}{C_2}z^{-1}V_i$$

$$H(z) = \frac{C_1}{C_2} \frac{z^{-1}}{z^{-1} + 1} = \frac{C_1}{C_2} \frac{1}{z - 1}$$



$$H(z) = \frac{C_1}{C_2} \frac{1}{z - 1}$$

Noise



Both phases add noise, $V_n^2 > \frac{2kT}{C}$

Mean

$$\overline{x(t)} = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} x(t) dt$$

Mean Square

$$\overline{x^2(t)} = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} x^2(t) dt$$

Variance

$$\sigma^2 = \overline{x^2(t)} - \overline{x(t)}^2$$

where σ is the standard deviation.

If mean is removed, or is zero, then

$$\sigma^2 = \overline{x^2(t)}$$

Assume two random processes, $x_1(t)$ and $x_2(t)$ with mean of zero (or removed).

$$x_{tot}(t) = x_1(t) + x_2(t)$$

$$x_{tot}^2(t) = x_1^2(t) + x_2^2(t) + 2x_1(t)x_2(t)$$

Variance (assuming mean of zero)

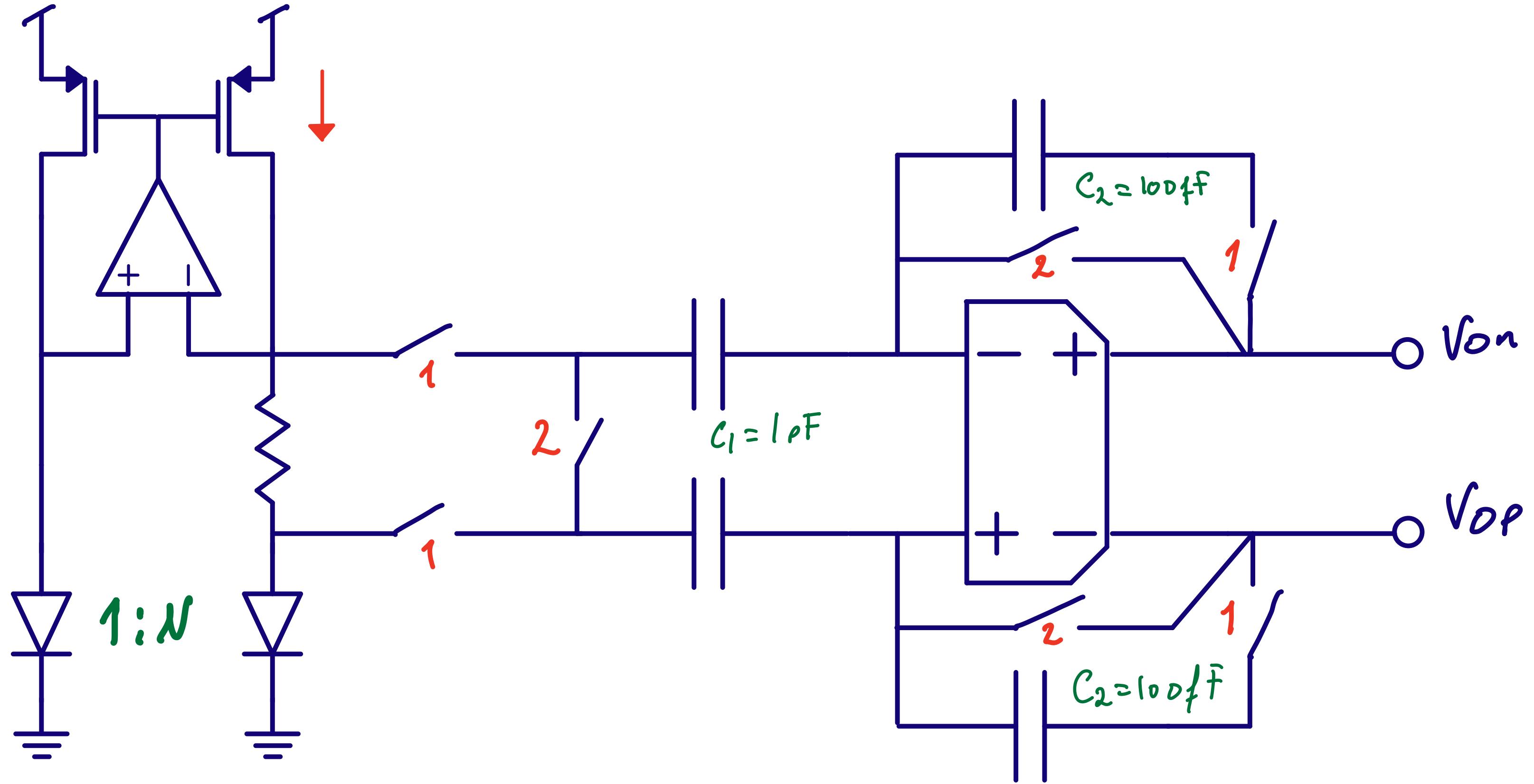
$$\sigma_{tot}^2 = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} x_{tot}^2(t) dt$$

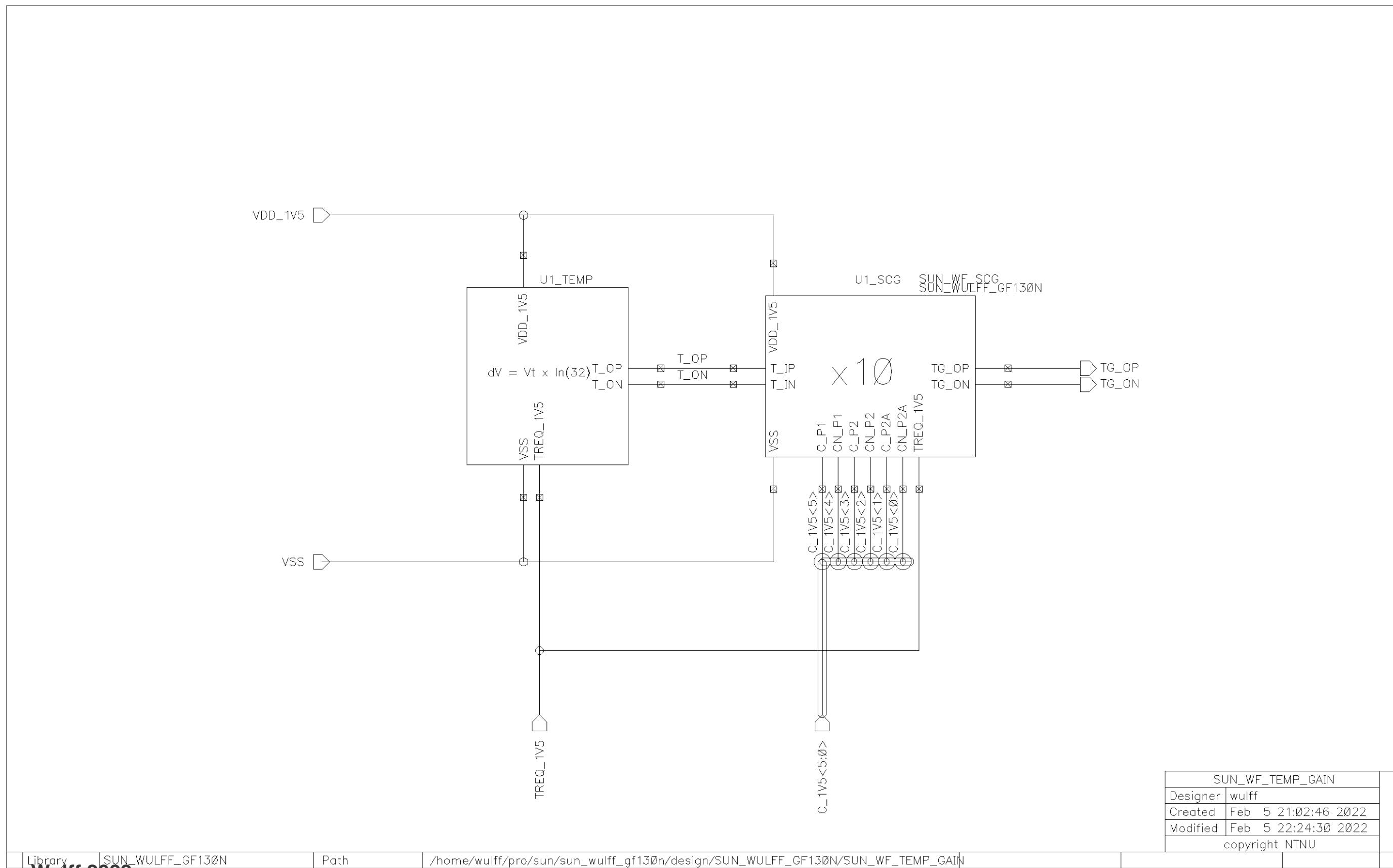
$$\sigma_{tot}^2 = \sigma_1^2 + \sigma_2^2 + \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-T/2}^{+T/2} 2x_1(t)x_2(t) dt$$

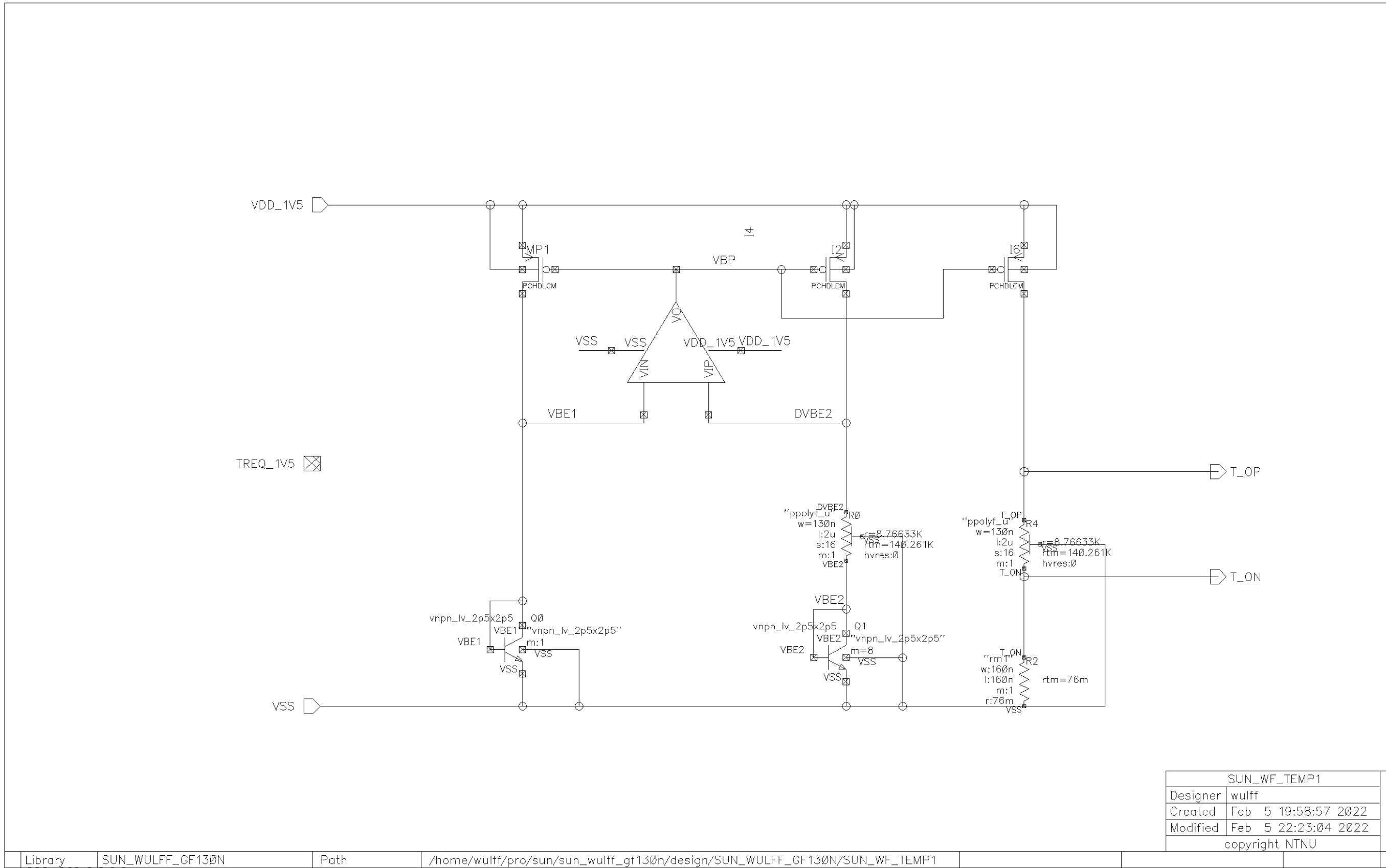
**Assuming uncorrelated processes
(covariance is zero), then**

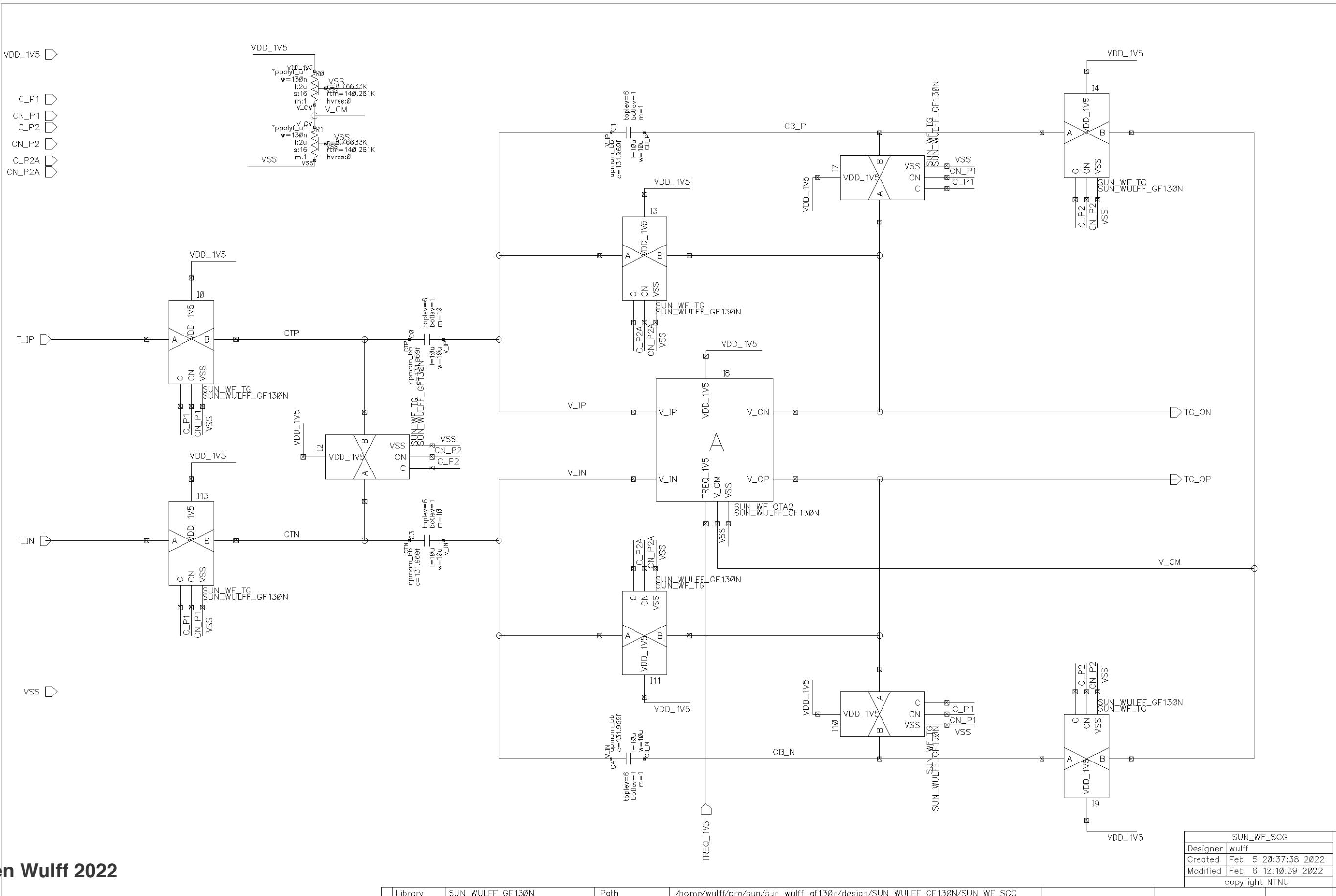
$$\sigma_{tot}^2 = \sigma_1^2 + \sigma_2^2$$

Example









 Virtuoso® ADE Assembler Editing: TB_SUN_WULFF_GF130N TB_SUN_SC_TEMP_GAIN maestro

sun EAD Parasitics/LDE Window Calibre Hel



No Sweeps Single Run, Sweeps and Corners Preferences

Outputs Setup

Results



10 rows

Test	Output	Spec	sig	ss/F	Min	Max	t_sweep_0	t_sweep_1	t_sweep_2	t_sweep_3	t_sweep_4	t_sweep_5	t_sweep_6	t_sweep_7	t_sweep_8	t_sweep_9
Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter	Filter
MAIN	t_op						↙	↙	↙	↙	↙	↙	↙	↙	↙	↙
MAIN	t_on						↙	↙	↙	↙	↙	↙	↙	↙	↙	↙
MAIN	tg_op						↙	↙	↙	↙	↙	↙	↙	↙	↙	↙
MAIN	tg_on						↙	↙	↙	↙	↙	↙	↙	↙	↙	↙
MAIN	dvbe						↙	↙	↙	↙	↙	↙	↙	↙	↙	↙
MAIN	dvbe_x10						↙	↙	↙	↙	↙	↙	↙	↙	↙	↙
MAIN	t_sample				19u	...	19u									
MAIN	v_dvbe				50....	...	50.8m	54.15m	57.39m	60.54m	63.61m	66.61m	69.56m	72.46m	75.33m	78.2m
MAIN	v_dvbe_x10				50...	...	502.4m	535.6m	567.6m	598.7m	629.1m	658.8m	688m	716.7m	745.1m	773.5m
MAIN	GAIN				9.89	...	9.89	9.89	9.89	9.891	9.89	9.89	9.891	9.891	9.891	9.891

In the book that you really should read

Parasitic sensitive integrator

Gain circuit that zero's OTA offset

Correlated double sampling

vco

Peak detectors

Thanks!

