Analog Filter Design

Part. 4: Discrete Time Filters Sect. 4-c: Optional Subjects

Summary of analog filters

Pure electronic / electric filters

- RCL passive filters
- Active RC filters
- Active Gm-C filters
- Current mode filters*
- Switched-Capacitor filters

Electro-mechanical filters

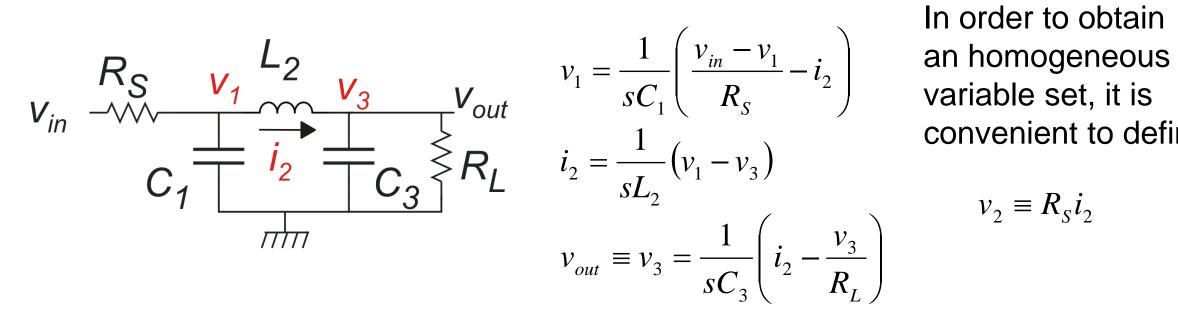
- SAW (surface acoustic wave) filters*
- BAW (Bulk acoustic wave) filters*
- MEMS resonators*

* Not covered in this course

Filter synthesis by means of LC ladder network simulation with SC integrators

Advantage: low sensitivity with respect to component value variations

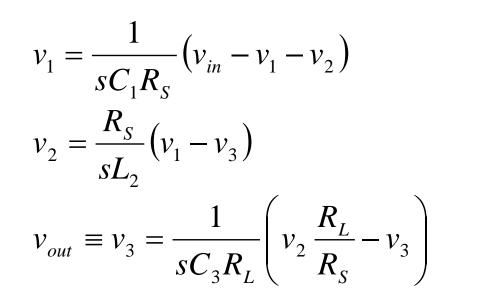
Example



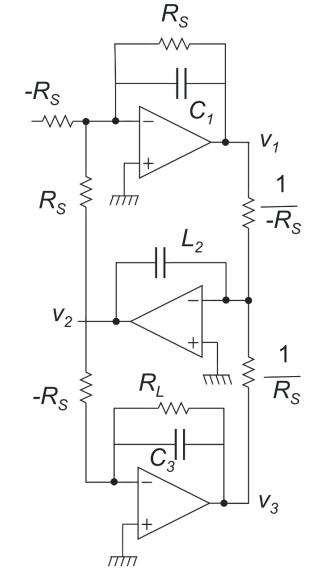
In order to obtain convenient to define:

$$v_2 \equiv R_s i_2$$

Example: ladder LC network simulation



- -) Indicated values represent numerical identities (dimensions are not relevant)
- -) Resistors are implemented with either positive or negative parasitic insensitive switched capacitors resistances.

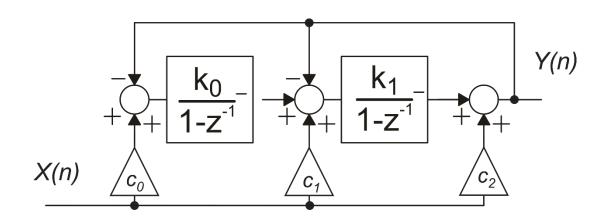


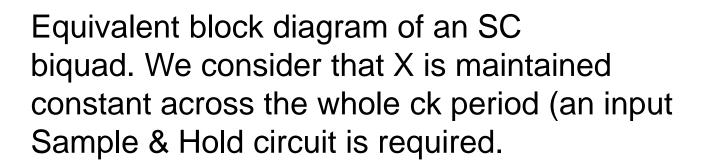
SC Filters that do not require the equivalent resistance approximation

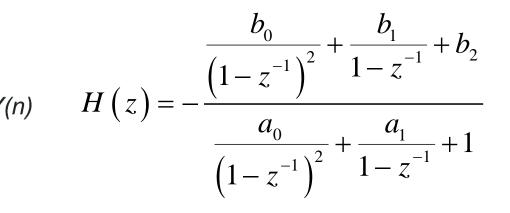
- These filters are obtained by direct implementation of the H(z) transfer function.
- > The H(z) can be obtained by means of:
 - -) conversion of a CT transfer function into the DT domain, by substituting
 - "s" with a proper rational function of "z" (e.g. bilinear transformation);
 - -) synthesis with the typical approaches of digital filters (e.g. FIR filters)
- > Analog implementation of the H(z) can be obtained using:
 - -) Conventional SC integrators cascades
 - -) Bilinear integrators
 - -) Direct implementation of the one-cycle delay function

Exact H(z) synthesis by means of SC Euler integrators

Example: biquad function







$$a_0 = k_1 k_0$$
 $a_1 = k_1$
 $b_0 = c_0 a_0$ $b_1 = c_1 a_1$ $b_2 = c_2$

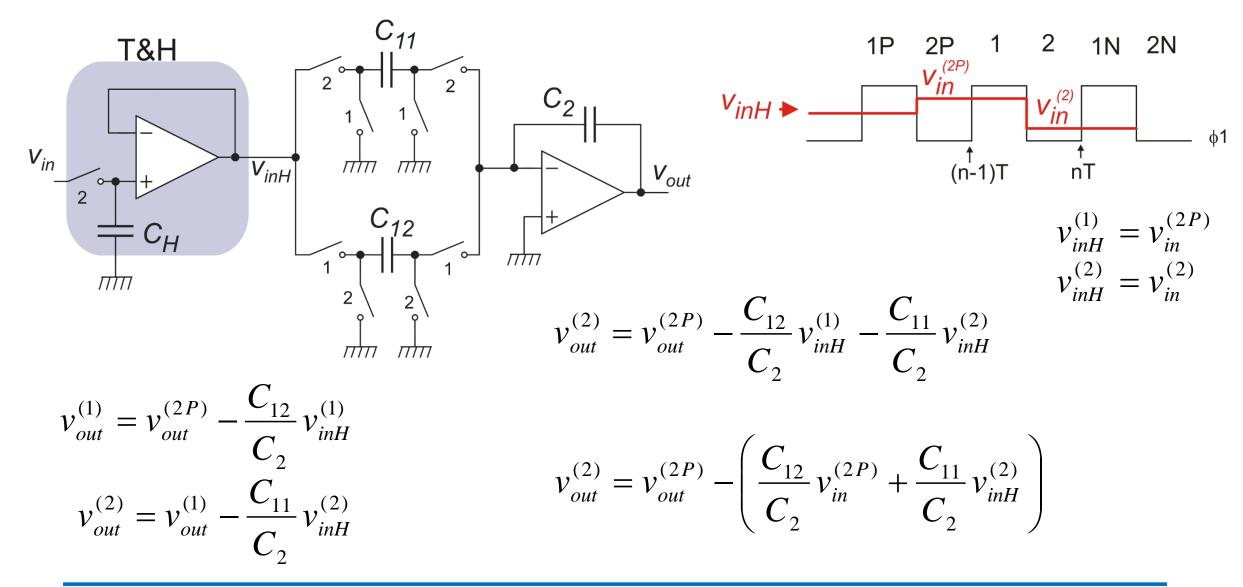
Exact H(z) synthesis by means of SC Euler integrators

$$H(z) = -\frac{\frac{b_0}{\left(1-z^{-1}\right)^2} + \frac{b_1}{1-z^{-1}} + b_2}{\frac{a_0}{\left(1-z^{-1}\right)^2} + \frac{a_1}{1-z^{-1}} + 1} = \frac{b_0 + b_1\left(1-z^{-1}\right) + b_2\left(1-z^{-1}\right)^2}{a_0 + a_1\left(1-z^{-1}\right) + \left(1-z^{-1}\right)^2}$$

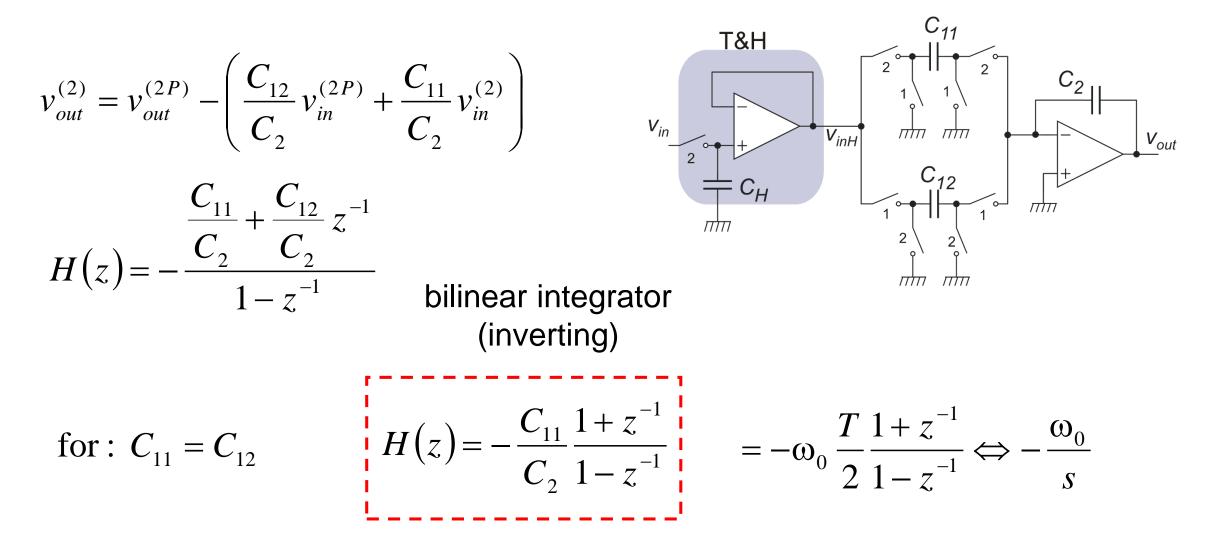
$$H(z) = \frac{b_2 z^{-2} + (-b_1 - 2b_2) z^{-1} + (b_0 + b_1 + b_2)}{z^{-2} + (-a_1 - 2) z^{-1} + (a_0 + a_1 + 1)}$$

The cascade of two simple SC integrators can implement an arbitrary 2nd order transfer function with a proper choice of the integrator coefficients (i.e. capacitance ratios).

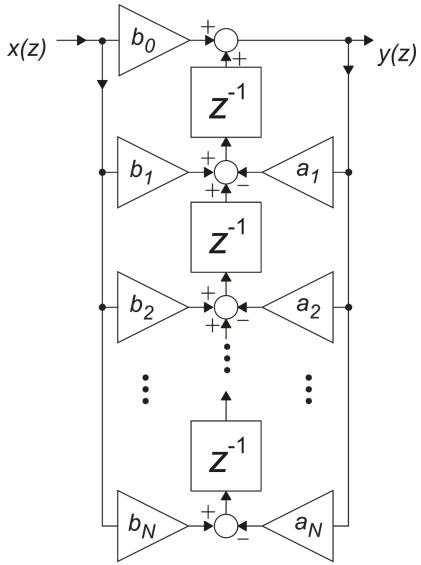
Example: synthesis of a bilinear integrator



Example: synthesis of a bilinear integrator



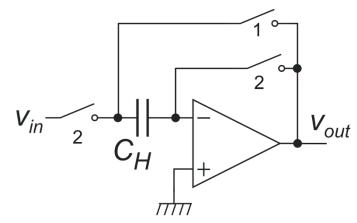
Direct synthesis



$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_M z^{-M}}{1 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_N z^{-M}}$$

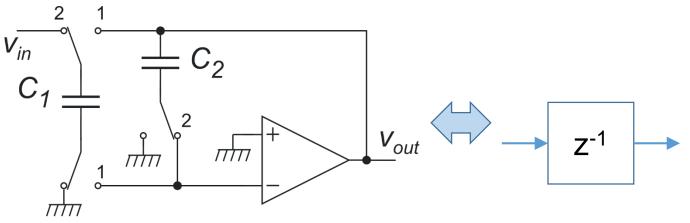
Delay lines and adders (summing amplifiers) are necessary

Direct synthesis: analog DT delay lines

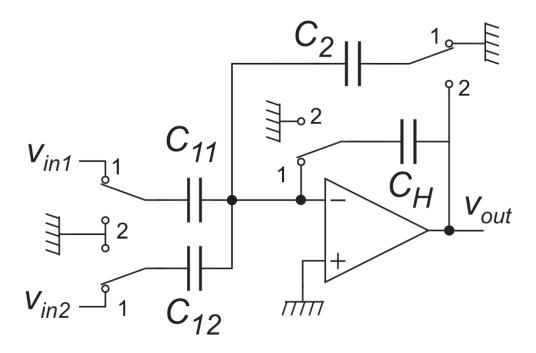


Half-period delay line with return to zero in phase 2

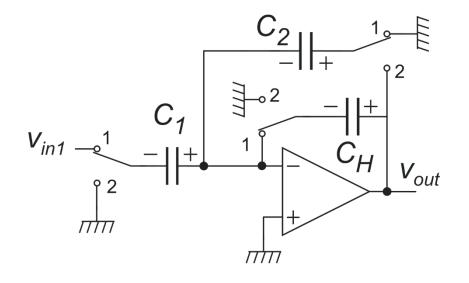
Sample and Hold (zero tracking time)



Summing amplifier

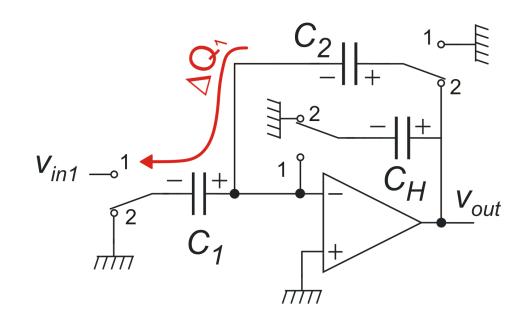


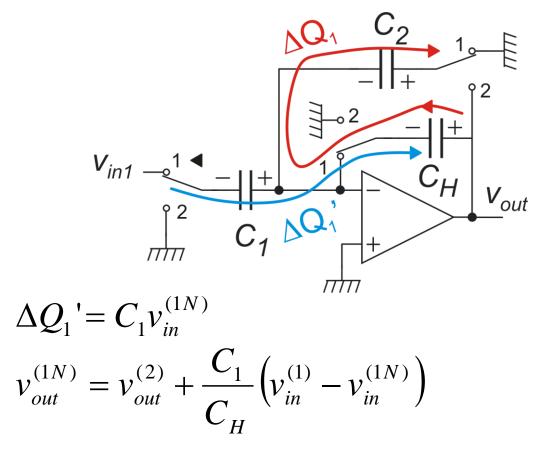
Simplified case: single input



Phase 1

Summing amplifier: analysis



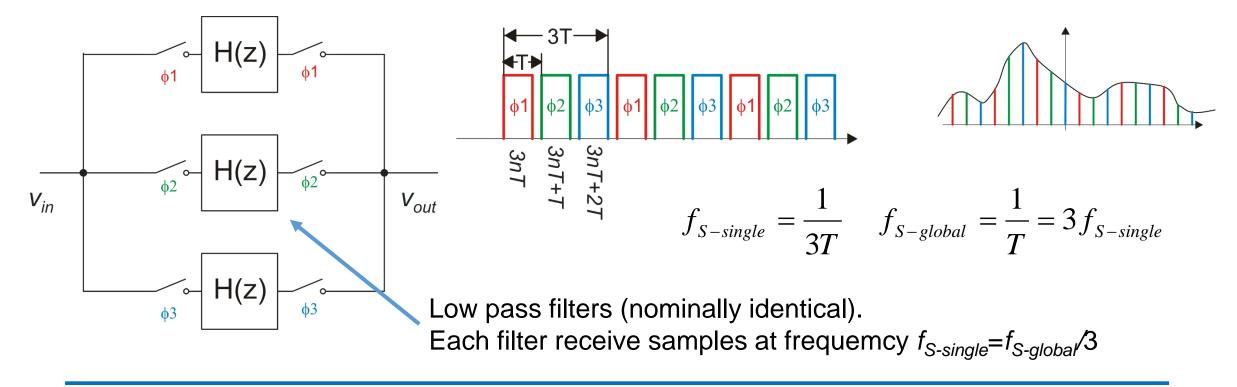


$$\Delta Q_1 = C_1 v_{in}^{(1)}$$
$$v_{out}^{(2)} = \frac{C_1}{C_2} v_{in}^{(1)}$$

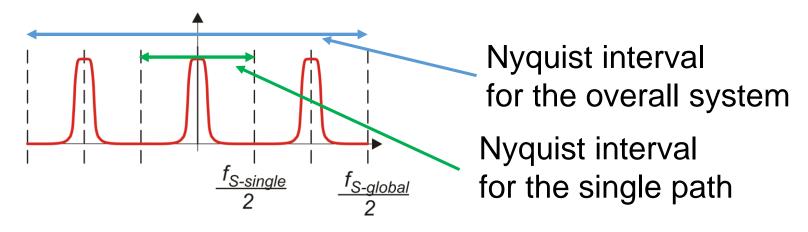
If v_{in} does not change much across a period, the output voltage is maintained in phase 1

Multipath filters

- > The target is obtaining a band-pass filter with a very narrow band (i.e. an high $Q=B/f_0$) i.e. a very selective filters.
- Synthesis of very selective Band-Pass filters by means of traditional techniques is very difficult due to component inaccuracy and active element non-idealities (e.g. amplifier gain)
- Multipath filters uses N low pass filters (in this example N=3) fed with decimated sample sequences, in order to explicitly produce aliasing.

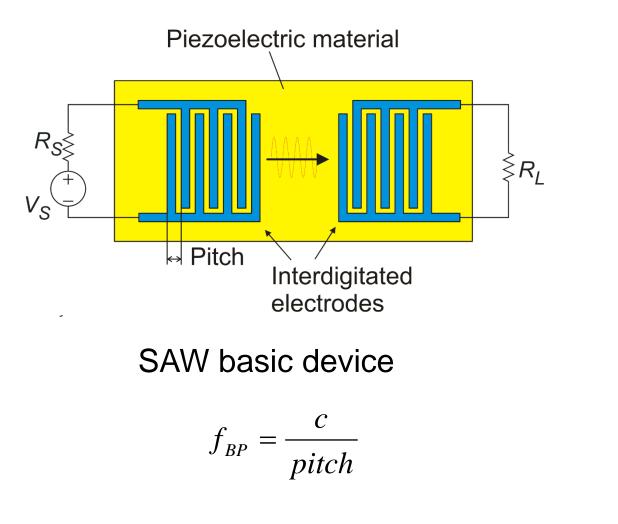


Multipath Filters

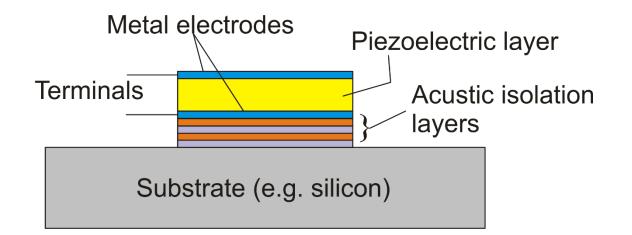


- Due to aliasing, the low pass response is duplicated around $f_{S-single}$.
- This would be meaningless for a single filter, since signals around f_{S-single} are beyond the Nyquist limit
- Using all the three filters together with delayed phases is equivalent to sampling at $f_{S-global}$. Now, signals at $f_{S-single}$ are within the Nyquist limit
- The replica of the response around $f_{S-single}$ can be made very narrow, by simply reducing the bandwidth of the individual low pass filters.

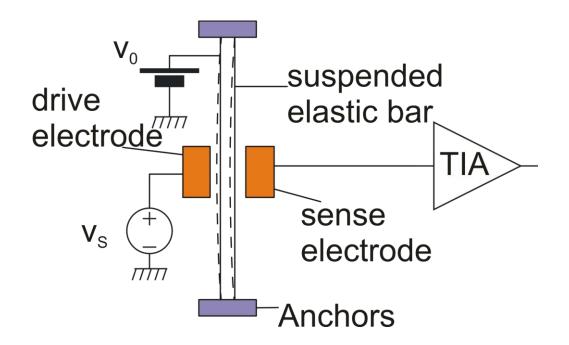
SAW and BAW filters

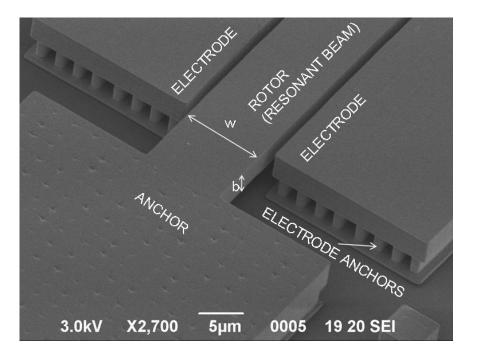


BAW device with Bragg reflector FBAR (Film Bulk Acoustic Resonator)



MEMS resonators: examples





MEMS with capacitive actuation: principle of operation

A resonator designed at the DII – Unipi (F. Pieri)

Examples of recently proposed integrated filters

The LP filter topology shown in Fig. 5.5.2 is derived from a 3rdorder, doubly terminated Butterworth LC ladder prototype using gyrator synthesis [1].

100M

50M 100M 200M

Max BW - - - - V_{DD}=0.7V ----

200M

500M

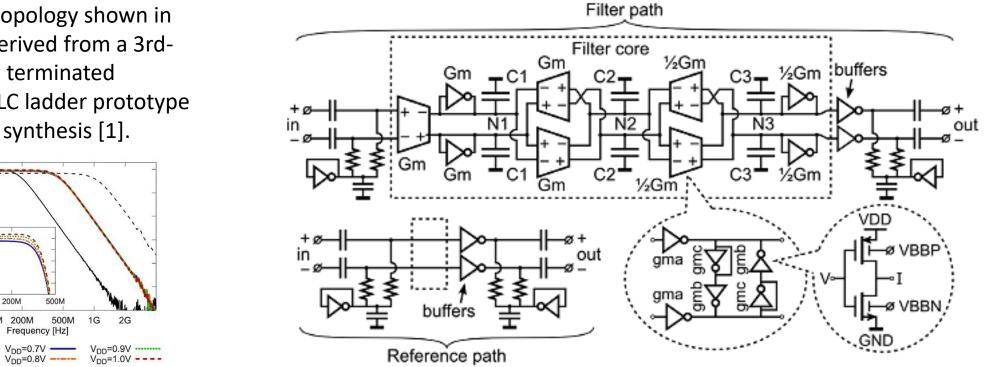
-10

-20 |S₂₁| [dB] -30

-40 -1.5 -2.5 -50

-60 20M 50M

Min BW ------



J. Lechevallier et. al, "A Forward-Body-Bias Tuned 450MHz GmC 3 rd-Order Low-Pass Filter in 28nm UTBB FD-SOI with >1dBVp IIP3 over a 0.7-to-1V Supply" ISSCC 2015