

# Delta-Sigma Analog-to-Digital Converters

*From System Architecture to Transistor-level  
Design*

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# Delta-Sigma Analog-to-Digital Converters

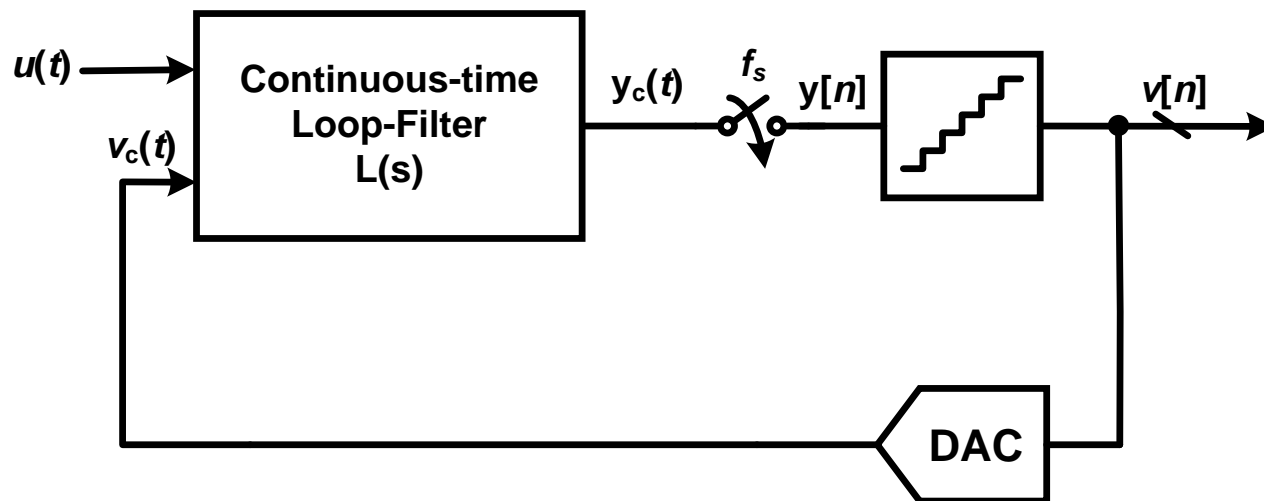
*Session II – Continuous-Time  $\Delta\Sigma$  ADCs*

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# Agenda

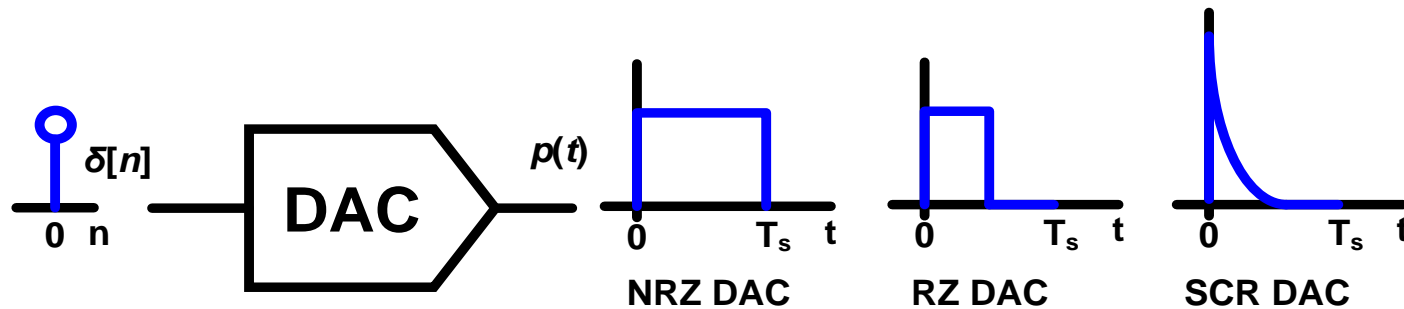
- Fundamentals of continuous-time (CT) ADCs
  - impulse invariance and CT loop-filter mapping
- Inherent anti-alias filtering in CT  $\Delta\Sigma$  ADCs
  - feedback vs. feedforward architectures.
- CT  $\Delta\Sigma$  ADC non-idealities:
  - Quantizer Excess loop-delay (ELD)
  - Clock jitter sensitivity
  - RC time-constant variation
- CT-  $\Delta\Sigma$  Design techniques:
  - NRZ, RZ and switched-capacitor DACs
  - Active-RC and gm-C implementations
  - ELD compensation techniques

# Continuous-Time $\Delta\Sigma$ Modulators



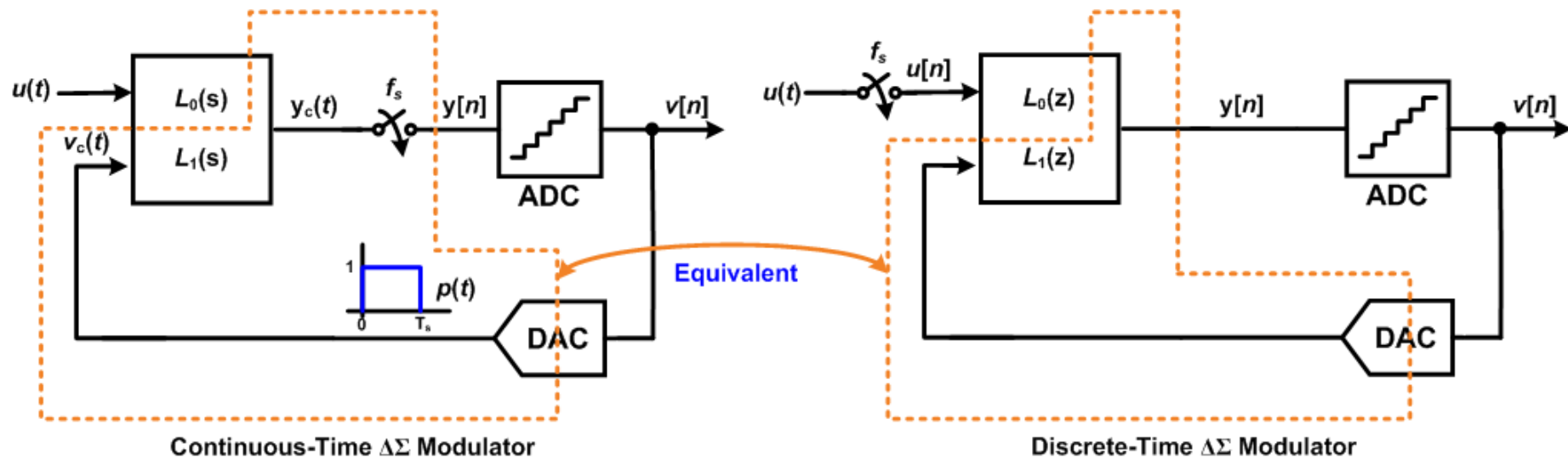
- The loop-filter ( $L(s)$ ) is implemented using continuous-time circuitry
  - The modulator is still a discrete-time system
- How to design the hybrid continuous- and discrete-time loop?

# Feedback DAC



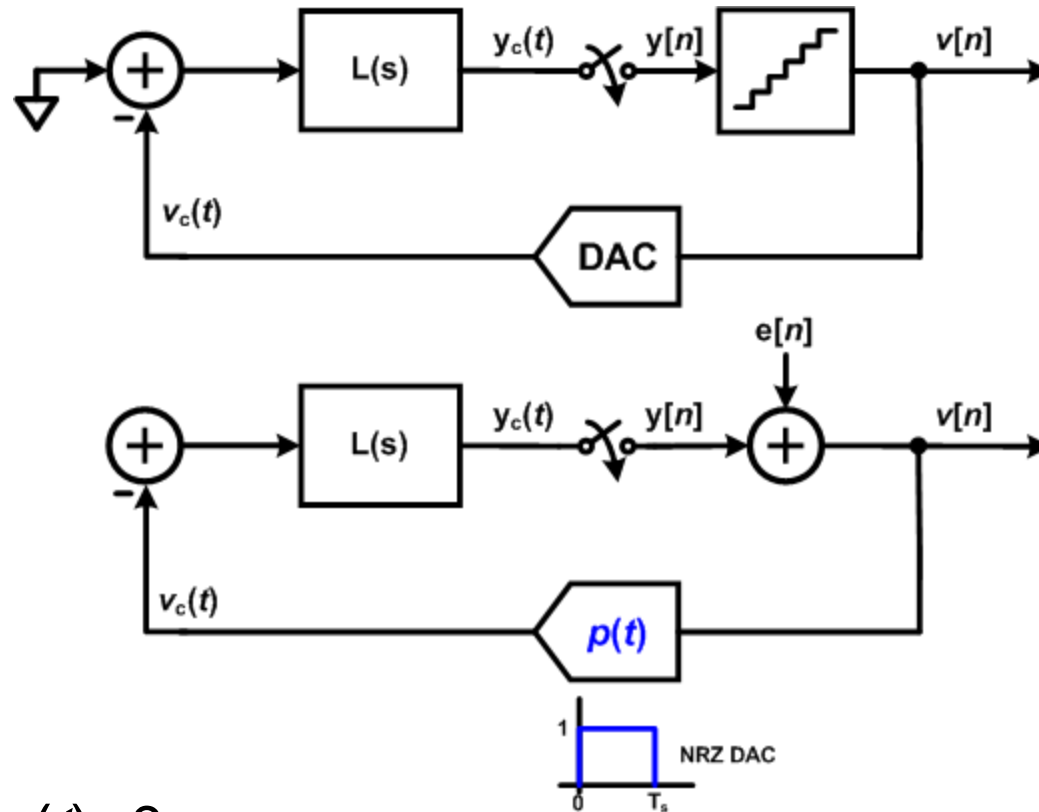
- ❑ The input to the DAC is a digital code with time-period  $T_s$
- ❑ The DAC output is an analog waveform
- ❑ The response of the DAC to the DT impulse,  $\delta[n]$ , is called the pulse-shape  $p(t)$
- ❑ Several choices for  $p(t)$  –
  - Return to zero (RZ)
  - Non-Return to zero (NRZ)
  - Switched capacitor Resistor (SCR), etc.

# Loop Modeling



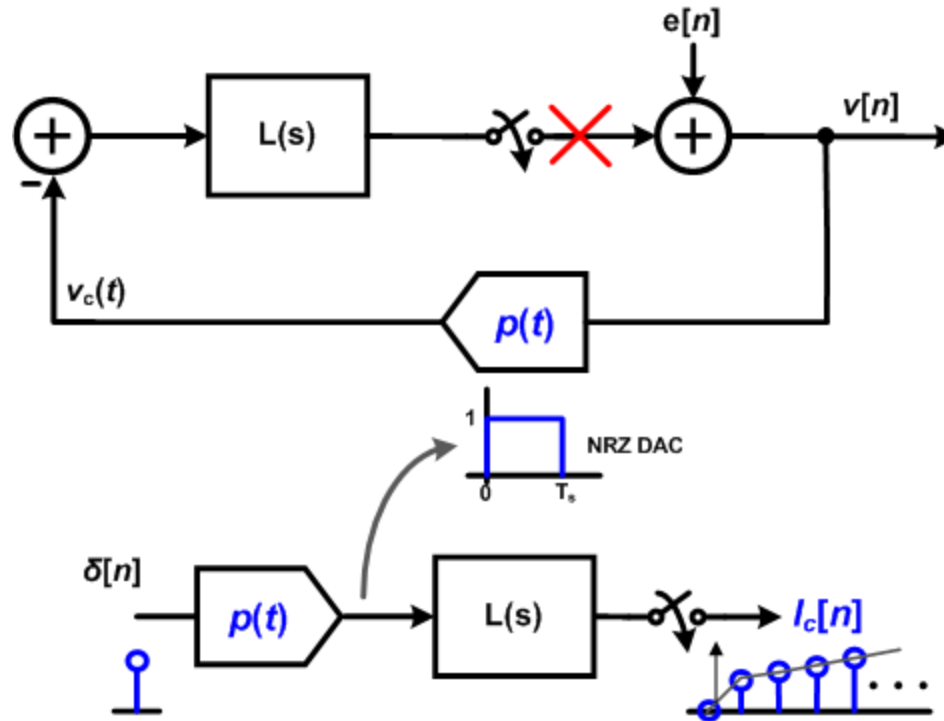
- ❑ In general loop-filter is a 2-input, 1-output LTI system
- ❑  $L_0(s)$  is the loop-filter seen by the input signal
- ❑  $L_1(s)=L(s)$ , is the loop-filter seen by the feedback signal
- ❑ The DT loop-response around the ADC-DAC block should be preserved in the CT loop

# Loop Modeling



- ❑ Set input  $u(t)=0$
- ❑ Replace ADC-DAC with the linear additive quantization noise model
- ❑ Model DAC as a filter with impulse response  $p(t)$

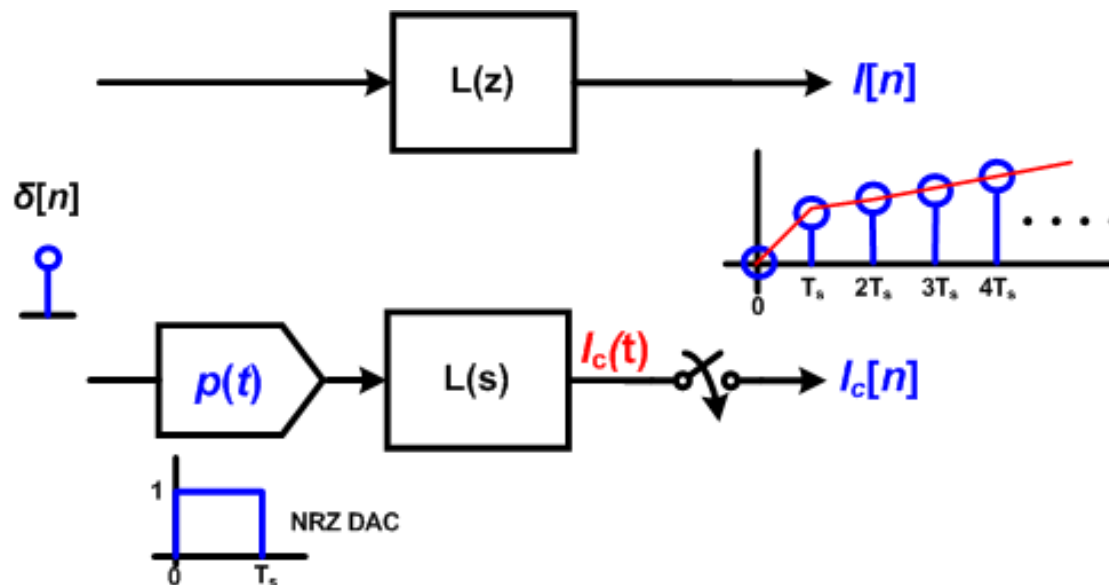
# Loop Modeling



- ❑ Break the loop after the sampler
- ❑ Apply a DT impulse
- ❑ The sampled output is  $l_c[n] = p(t) \otimes l(t) |_{nT_s}$
- ❑ The output sequence ( $l_c[n]$ ), should be identical to the DT loop-filter response  $l[n] \xrightarrow{Z} L(z)$

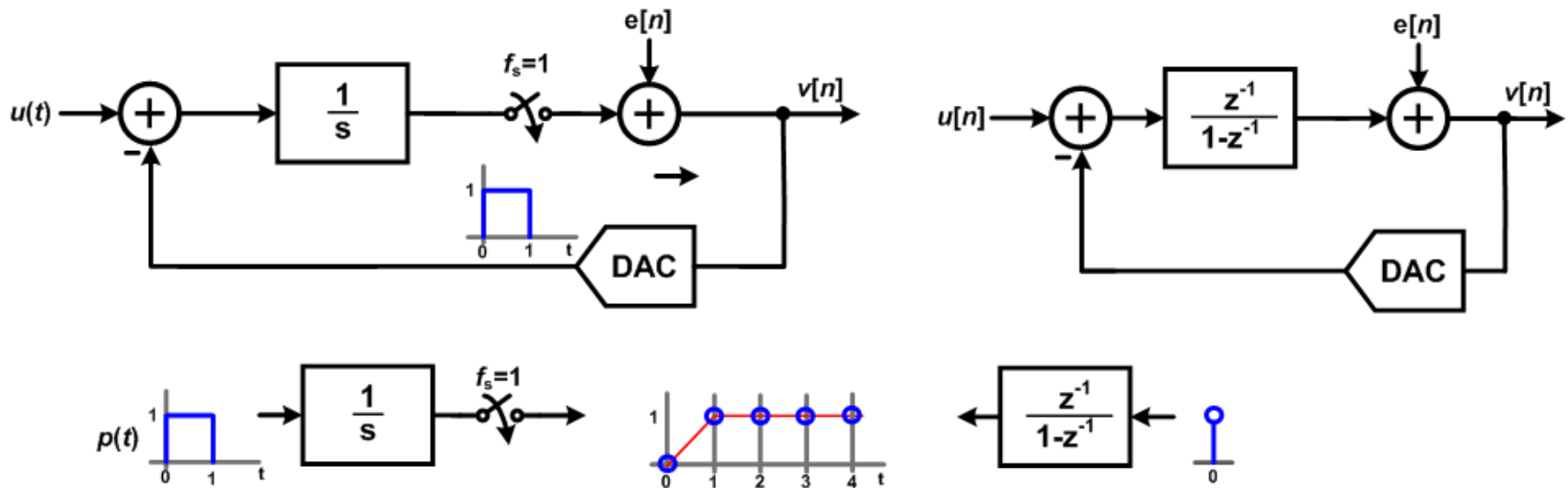


# Synthesis of CT Loop-Filter



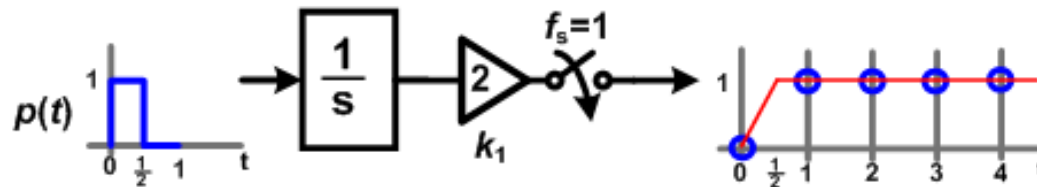
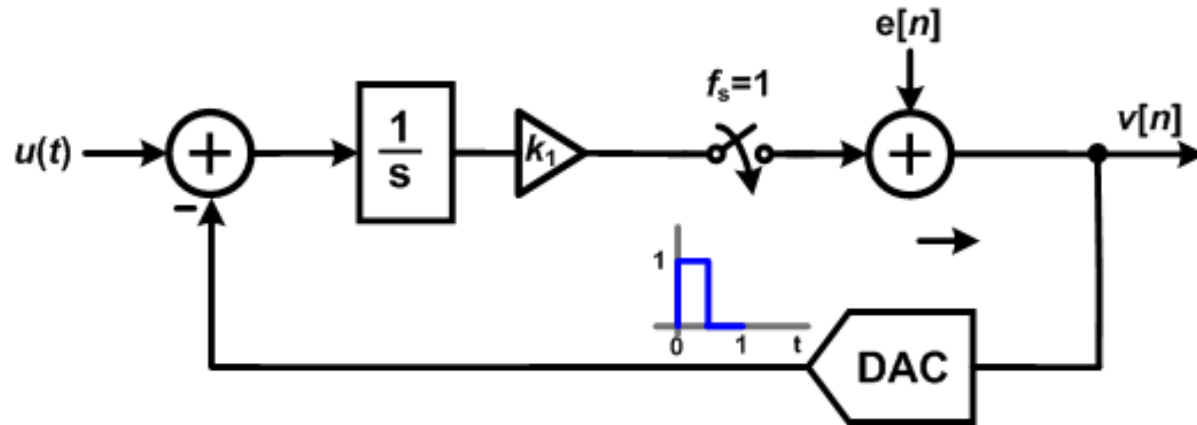
- Map the sampled CT loop-response to its DT equivalent
  - Called Impulse Invariant Transformation (IIT)
  - $l_c[n] = p(t) \otimes l(t) |_{nT_s} \square l[n]$
- $NTF(z)$  is preserved in the DT-to-CT if the transformation is correctly performed

# First-order Example – NRZ DAC



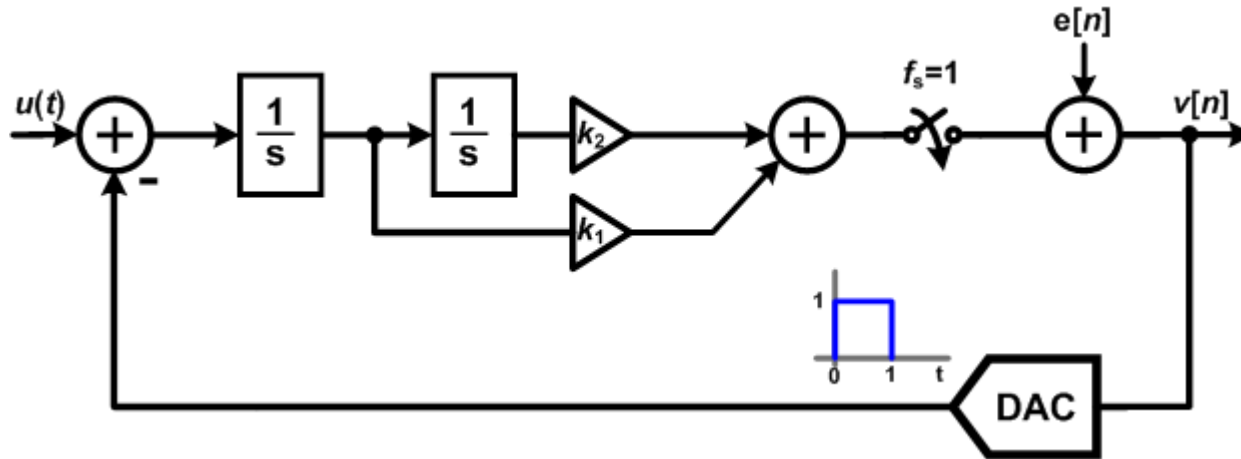
- Without loss of generality, let  $f_s=1$  Hz,  $T_s=1$  s
- Sampled loop-response  $l_c[n]=\{0,1,1,1,1,\dots\}$ 
  - Identical to the DT loop-response
- $$L(z) = \frac{z^{-1}}{1-z^{-1}}$$
- $$NTF(z) = \frac{1}{1+L(z)} = 1-z^{-1}$$

# First-order Example – RZ DAC



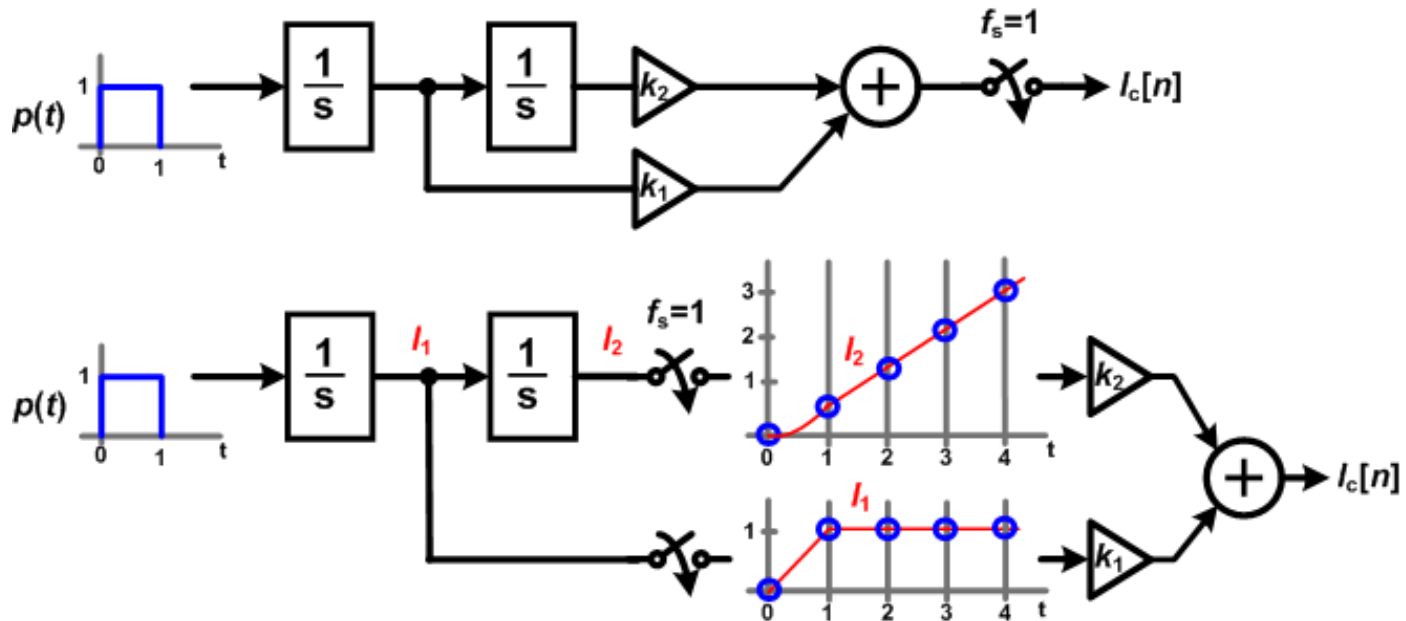
- With RZ DAC, scaling by a factor of  $k_1=2$  is needed to restore the loop-response
- Can have any number of CT responses to result in the same DT loop-response
  - The procedure can be conceptually applied to any other DAC pulse shape

# Second-order Example



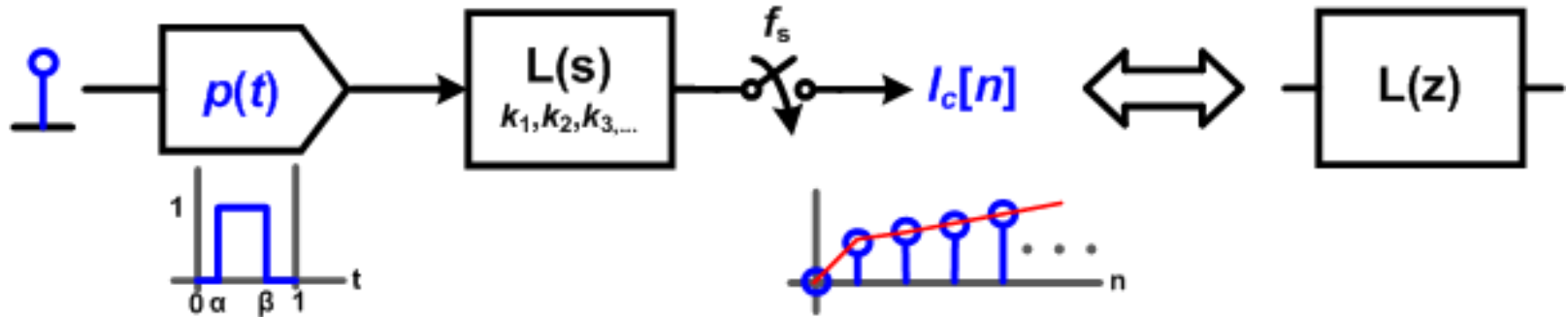
- $NTF(z) = (1 - z^{-1})^2$
- $L(z) = \frac{1}{NTF(z)} - 1 = \frac{1}{(1 - z^{-1})^2} - 1 = \frac{z^{-1}}{1 - z^{-1}} + \frac{z^{-1}}{(1 - z^{-1})^2}$
- $\uparrow[n] = \{0, 1, 1, \dots\} + \{0, 1, 2, 3, \dots\} = \{0, 2, 3, 4, 5, \dots\}$
- The loop-response is implemented using a cascade of two CT integrators
  - Need to find suitable loop-filter coefficients  $k_1$  and  $k_2$  using IIT

# Second-order Example



- $\lceil n \rceil = \{0, 2, 3, 4, 5, \dots\}$
- $l_c[n] = k_1 \cdot \{0, 1, 1, 1, \dots\} + k_2 \cdot \{0, 1.5, 2.5, 3.5, \dots\}$
- Solving for  $l_c[n] = \lceil n \rceil$ :  $k_1 = 1.5$  and  $k_2 = 1$
- For NRZ DAC, CT loop-filter,  $L(s) = \frac{1.5}{s} + \frac{1}{s^2}$
- Show that for RZ DAC,  $k_1 = 2.5$  and  $k_2 = 2$

# Impulse-Invariant Transformation



- Impulse-invariant transformation maps DT modulator to CT
  - $l[n] = p(t) \otimes l(t) \big|_{t=nT_s}$
  - $\mathbf{Z}^{-1}\{L(z)\} = \mathbf{L}^{-1}\{P(s) \cdot L(s)\} \big|_{t=nT_s}$
- These relations have been derived for standard z-domain poles
  - Uses generic rectangular DAC pulse defined by  $(\alpha, \beta)$ ,  $0 \leq \alpha < \beta \leq 1$
  - Tables available for z-domain to s-domain pole equivalence and vice versa [Cherry]
  - A DT pole at  $z=z_k$  transforms to a CT pole at  $s_k = \ln(z_k)$  with the same multiplicity ( $l$ )

# z-Domain to s-Domain Loop Filter Poles

s-DOMAIN EQUIVALENCES FOR z-DOMAIN LOOP FILTER POLES

| z-domain pole         | s-domain equivalent   | Limit for $z_k = 1$   |
|-----------------------|---|---|
| $\frac{1}{z-z_k}$     | $\frac{r_0}{s-s_k} \times \frac{1}{z_k^{1-\alpha} - z_k^{1-\beta}}$<br>$r_0 = s_k$  | $\frac{r_0}{s-s_k}$<br>$r_0 = \frac{1}{\beta-\alpha}$   |
| $\frac{1}{(z-z_k)^2}$ | $\frac{r_1 s + r_0}{(s-s_k)^2} \times \frac{1}{z_k(z_k^{1-\alpha} - z_k^{1-\beta})^2}$<br>$r_1 = q_1 s_k + q_0$<br>$r_0 = q_1 s_k^2$<br>$q_1 = z_k^{1-\beta}(1-\beta) - z_k^{1-\alpha}(1-\alpha)$<br>$q_0 = z_k^{1-\alpha} - z_k^{1-\beta}$   | $\frac{r_1 s + r_0}{(s-s_k)^2}$<br>$r_1 = \frac{1}{2} \frac{\alpha + \beta - 2}{\beta - \alpha}$<br>$r_0 = \frac{1}{\beta - \alpha}$  |
| $\frac{1}{(z-z_k)^3}$ | $\frac{r_2 s^2 + r_1 s + r_0}{(s-s_k)^3} \times \frac{1}{z_k^2(z_k^{1-\alpha} - z_k^{1-\beta})^3}$<br>$r_2 = \frac{1}{2} q_2 s_k - q_1$<br>$r_1 = -q_2 s_k^2 + q_1 s_k + q_0$<br>$r_0 = \frac{1}{2} q_2 s_k^3$<br>$q_2 = (1-\beta)(2-\beta)(z_k^{1-\beta})^2 + (1-\alpha)(2-\alpha)(z_k^{1-\alpha})^2 + [\beta(\beta+3) + \alpha(\alpha+3) - 4(1+\alpha\beta)] z_k^{1-\alpha} z_k^{1-\beta}$<br>$q_1 = (\frac{3}{2} - \beta)(z_k^{1-\beta})^2 + (\frac{3}{2} - \alpha)(z_k^{1-\alpha})^2 + (\alpha + \beta - 3) z_k^{1-\alpha} z_k^{1-\beta}$<br>$q_0 = (z_k^{1-\alpha} - z_k^{1-\beta})^2$ | $\frac{r_2 s^2 + r_1 s + r_0}{(s-s_k)^3}$<br>$r_2 = \frac{1}{12} \frac{1}{\beta - \alpha} [\beta(\beta - 9) + \alpha(\alpha - 9) + 4\alpha\beta + 12]$<br>$r_1 = \frac{1}{2} \frac{\alpha + \beta - 3}{\beta - \alpha}$<br>$r_0 = \frac{1}{\beta - \alpha}$ |

# s-Domain to z-Domain Loop Filter Poles

z-DOMAIN EQUIVALENCES FOR s-DOMAIN LOOP FILTER POLES

| s-domain pole         | z-domain equivalent  | Limit for $s_k = 0$  |
|-----------------------|--|--|
| $\frac{1}{s-s_k}$     | $\frac{y_0}{z-z_k} \times \frac{1}{s_k}$<br>$y_0 = z_k^{1-\alpha} - z_k^{1-\beta}$   | $\frac{y_0}{z-z_k}$<br>$y_0 = \beta - \alpha$  |
| $\frac{1}{(s-s_k)^2}$ | $\frac{y_1 z + y_0}{(z-z_k)^2} \times \frac{1}{s_k^2}$<br>$y_1 = z_k^{1-\beta}[1 - s_k(1 - \beta)]$<br>$- z_k^{1-\alpha}[1 - s_k(1 - \alpha)]$<br>$y_0 = z_k^{2-\alpha}(1 + s_k \alpha)$<br>$- z_k^{2-\beta}(1 + s_k \beta)$   | $\frac{y_1 z + y_0}{(z-z_k)^2}$<br>$y_1 = \frac{1}{2}[\beta(2 - \beta) - \alpha(2 - \alpha)]$<br>$y_0 = \frac{1}{2}(\beta^2 - \alpha^2)$   |
| $\frac{1}{(s-s_k)^3}$ | $\frac{y_2 z^2 + y_1 z + y_0}{(z-z_k)^3} \times \frac{1}{s_k^3}$<br>$y_2 = z_k^{1-\beta}[-1 + s_k(1 - \beta) + \frac{s_k^2}{2}(1 - \beta)^2]$<br>$- z_k^{1-\alpha}[-1 + s_k(1 - \alpha) + \frac{s_k^2}{2}(1 - \alpha)^2]$<br>$y_1 = z_k^{2-\beta}[2 - s_k(1 - 2\beta)$<br>$+ \frac{s_k^2}{2}(-1 - 2\beta + 2\beta^2)]$<br>$+ z_k^{2-\alpha}[2 - s_k(1 - 2\alpha)$<br>$+ \frac{s_k^2}{2}(-1 - 2\alpha + 2\alpha^2)]$<br>$y_0 = z_k^{3-\alpha}(1 + s_k \alpha + \frac{s_k^2}{2}\alpha^2)$<br>$- z_k^{3-\beta}(1 + s_k \beta + \frac{s_k^2}{2}\beta^2)$ | $\frac{y_2 z^2 + y_1 z + y_0}{(z-z_k)^3}$<br>$y_2 = \frac{1}{6}(\beta^3 - \alpha^3)$<br>$- \frac{1}{2}(\beta^2 - \alpha^2) + \frac{1}{2}(\beta - \alpha)$<br>$y_1 = \frac{1}{3}(\beta^3 - \alpha^3)$<br>$- \frac{1}{2}(\beta^2 - \alpha^2) - \frac{1}{2}(\beta - \alpha)$<br>$y_0 = \frac{1}{6}(\beta^3 - \alpha^3)$ |

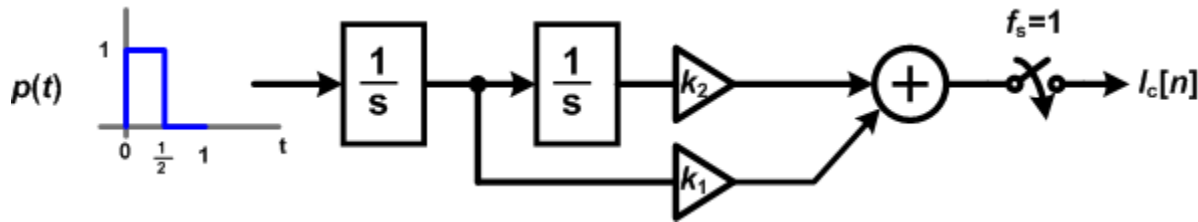


# IIT Example: 2<sup>nd</sup>-order NTF, RZ DAC

- Expand  $L(z)$  as pole-form partial fractions

$$NTF(z) = (1 - z^{-1})^2$$

$$L(z) = \frac{2}{z-1} + \frac{1}{(z-1)^2}$$



$$p(t) \square [\alpha, \beta] = [0, 0.5]$$

- From the equivalence tables

$$\frac{1}{z-1} \xrightarrow{IIT} \frac{1}{(\beta-\alpha)} \cdot \frac{1}{s-s_k} = \frac{2}{s}$$

$$\frac{1}{(z-1)^2} \xrightarrow{IIT} = \frac{\frac{1}{2} \left( \frac{\alpha+\beta-2}{\beta-\alpha} \right) s + \frac{1}{(\beta-\alpha)}}{s^2} = \frac{(2-1.5s)}{s^2}$$

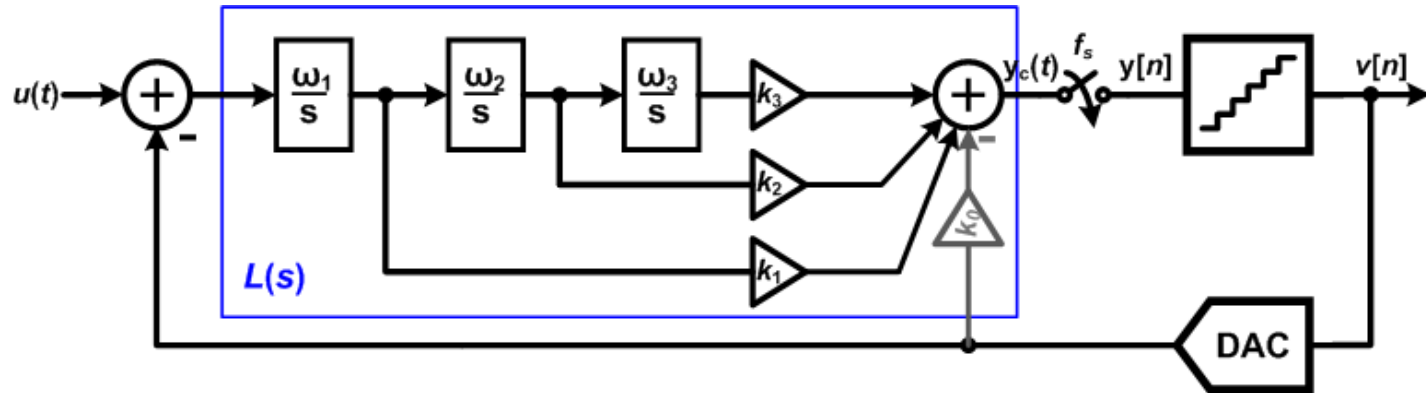
- Combining the terms, we get

$$\Rightarrow L(s) = \frac{2}{s} + \frac{2-1.5s}{s^2} = \frac{2+2.5s}{s^2}$$

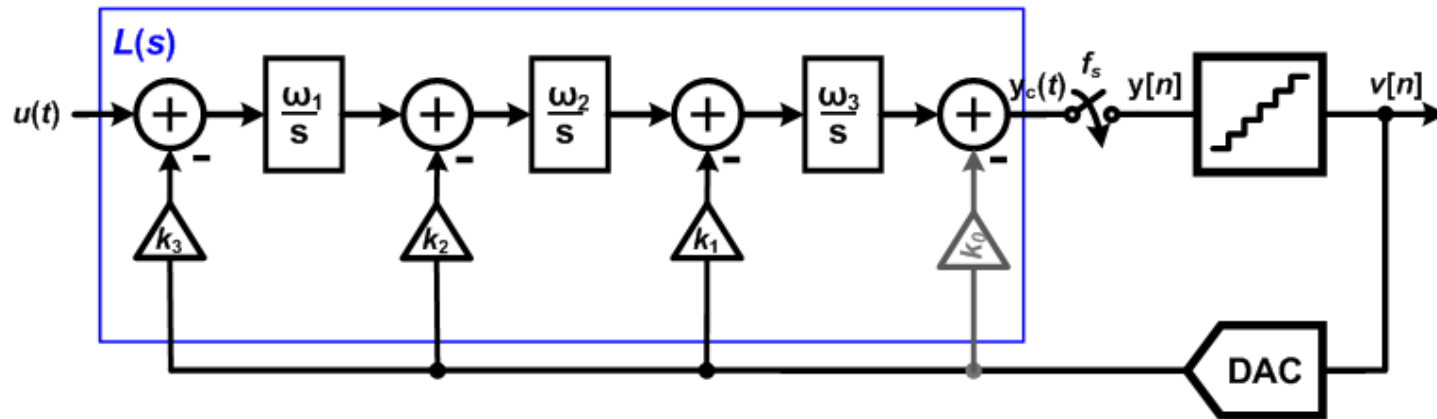
# Continuous-Time $\Delta\Sigma$ Summary

- ❑ A DT loop can be emulated using a CT loop-filter
- ❑ The DT-to-CT transformation depends on the DAC pulse shape
- ❑ The transformation can be performed using analytical as well as numerical methods
- ❑ This technique can be extended to higher-order NTFs
  - From the desired NTF(z), find L(z)
  - Convert L(z) into L(s) using the DAC pulse shape
  - Schreier  $\Delta\Sigma$  Toolbox function **realizeNTF\_ct** can be used for rectangular DAC pulses
  - Implement L(s) using a suitable loop-filter topology
- ❑ A CT  $\Delta\Sigma$  modulator has significant advantages over its DT implementation.... stay tuned

# Loop-Filter Architectures

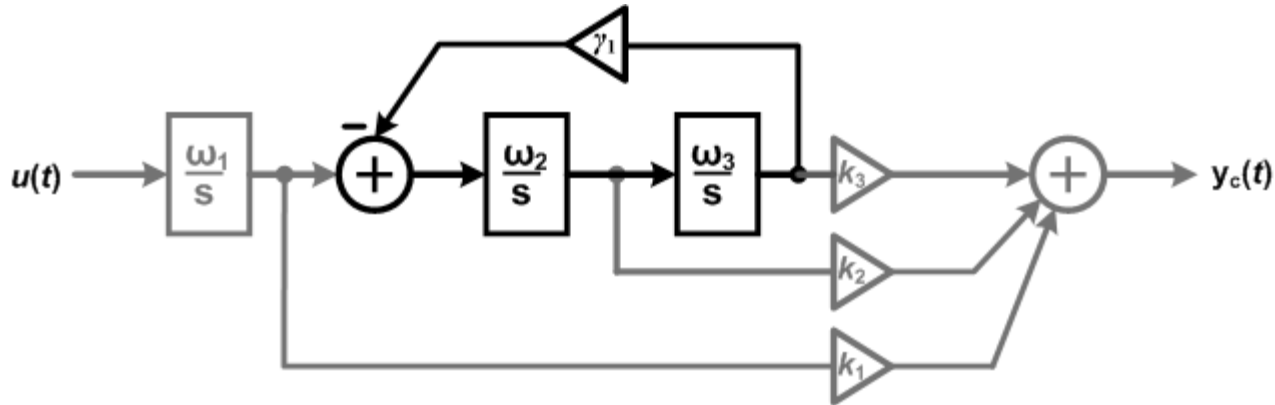


Cascade of Integrators with Feedforward Summation (**CIFF**)



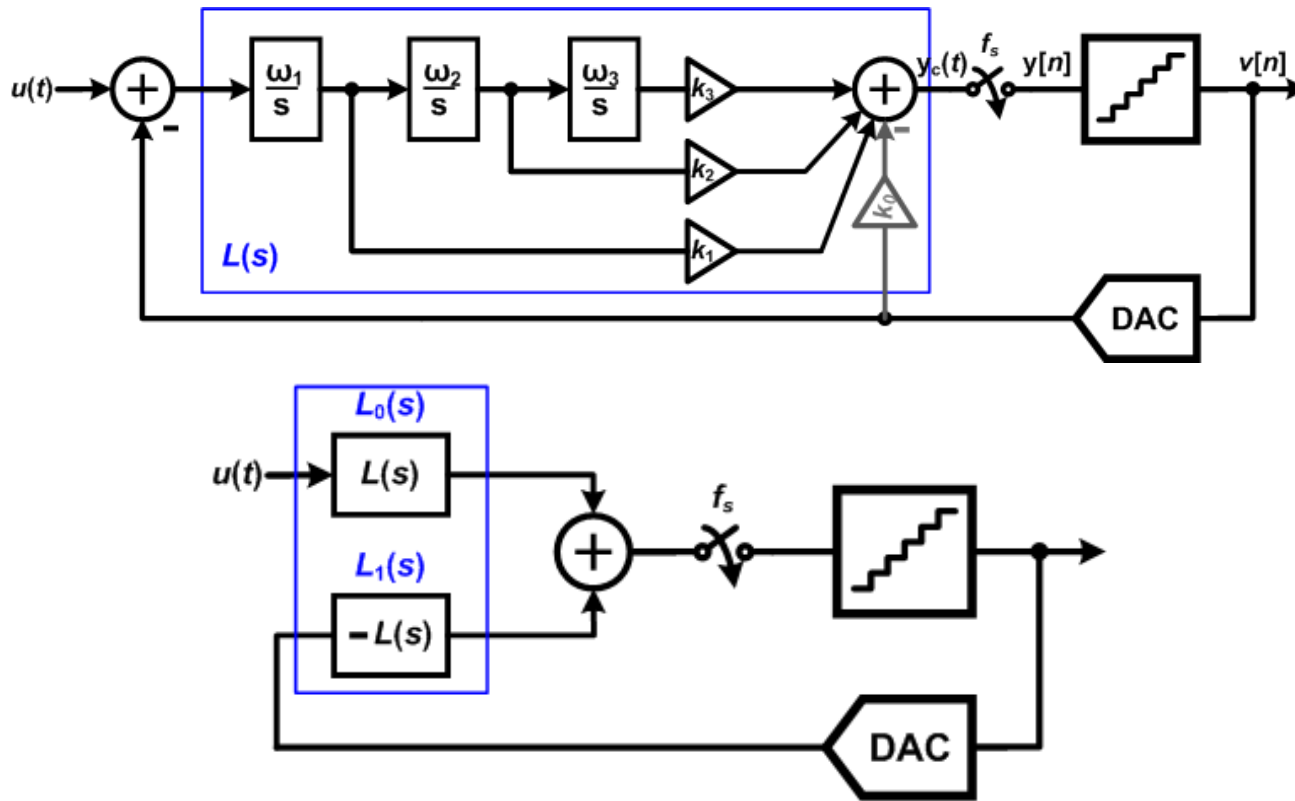
Cascade of Integrators with Distributed Feedback (**CIFB**)

# Resonators for Complex NTF Zeros



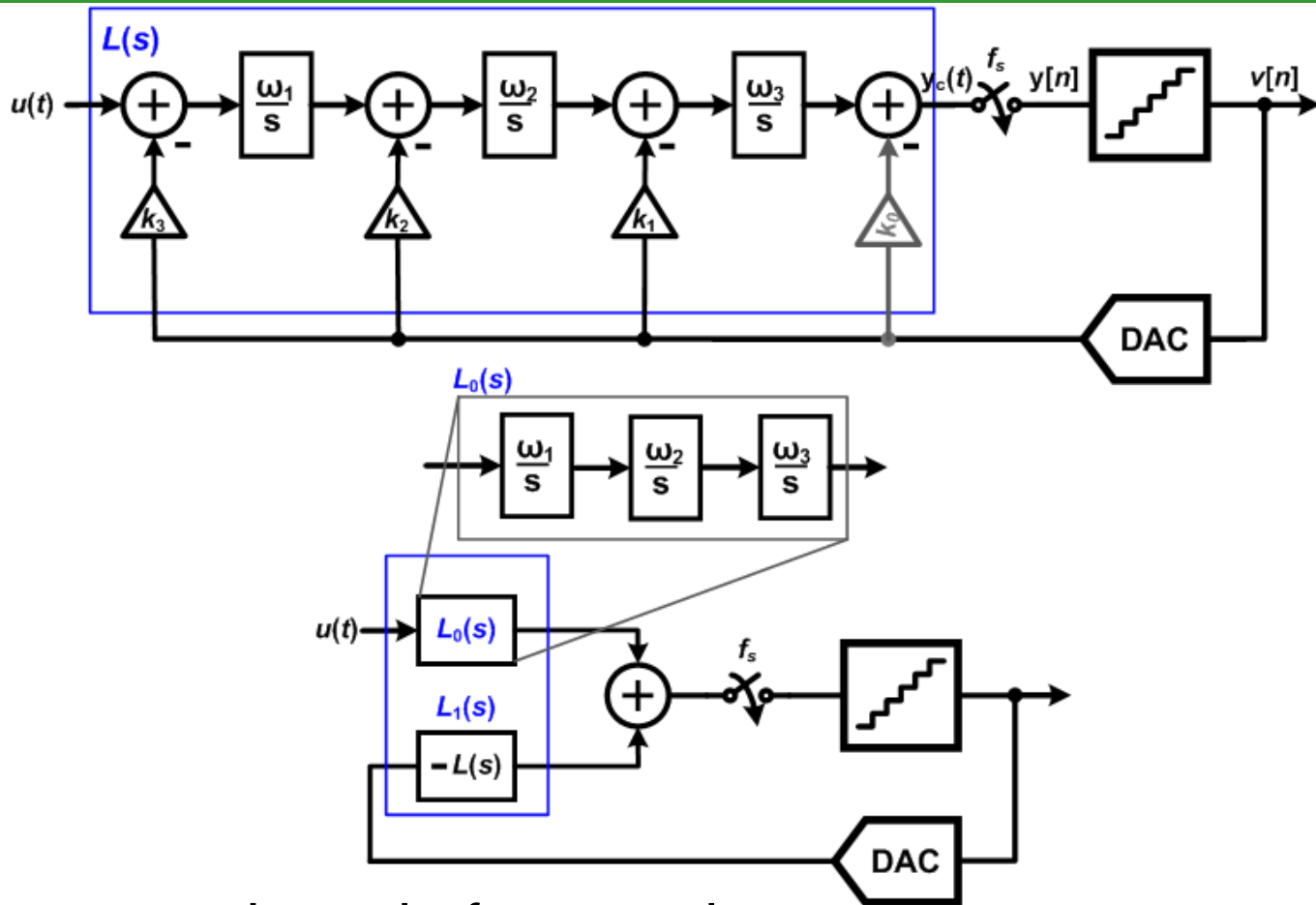
- Complex NTF( $z$ ) zeroes,  $z_k = e^{\pm j\alpha}$  transform into conjugate loop-filter poles at  $s_k = \pm |\angle z_k| = \pm j\alpha$
- Here,  $\alpha = \sqrt{\gamma_1 \omega_2 \omega_3}$

# CIFF Loop-Filter



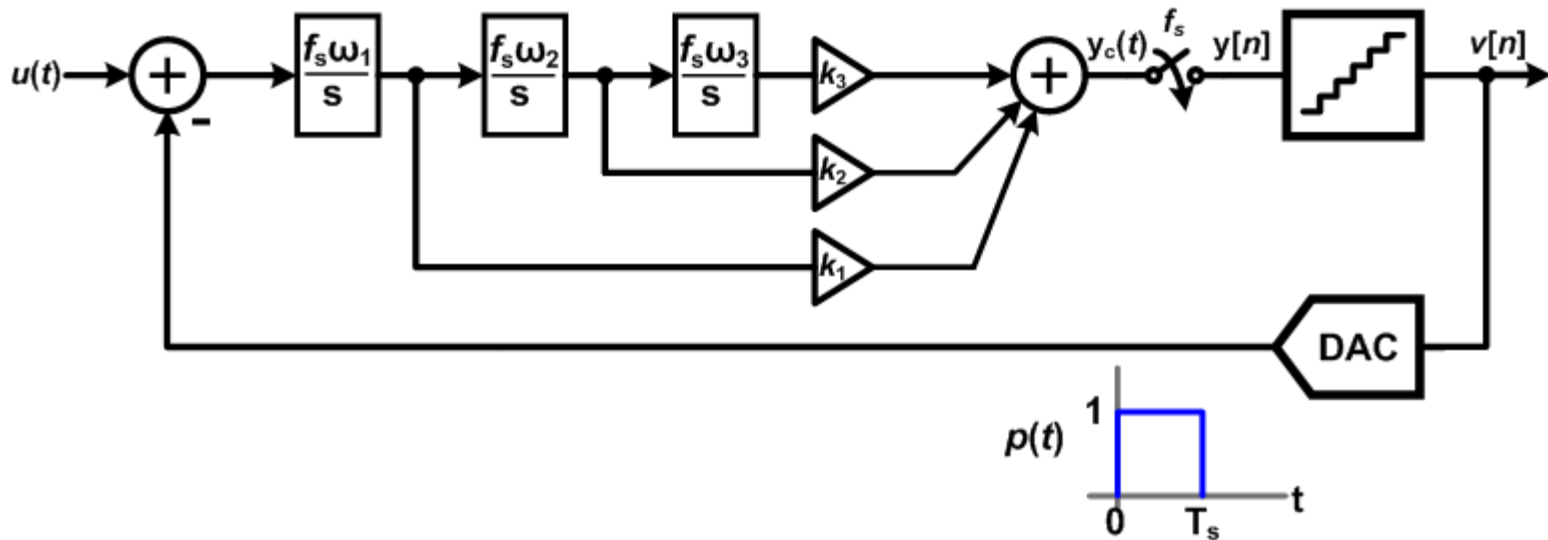
- Integrator unity-gain frequencies  $\omega_1 > \omega_2 > \omega_3$ 
  - First integrator is the fastest while the last is slowest
- Since the first integrator needs to be fastest for noise and linearity considerations, results in lower power loop-filter

# CIFB Loop-Filter



- ❑ Integrator unity-gain frequencies  $\omega_1 < \omega_2 < \omega_3$ 
  - ❑ First integrator is the slowest while the last is fastest
  - ❑ Not power efficient compared to CIFF

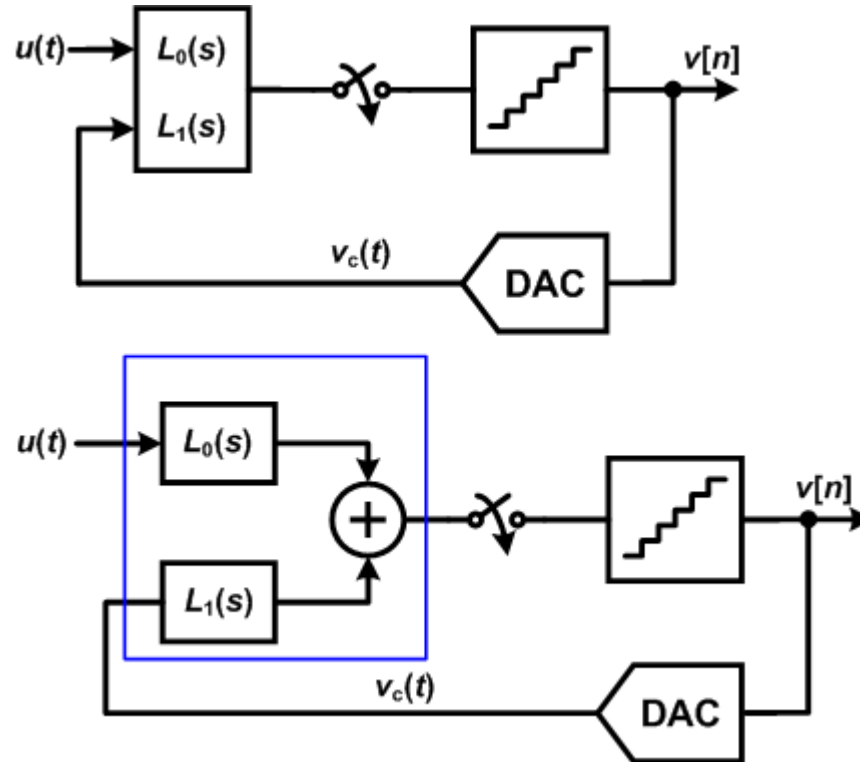
# Loop-Filter Time-Scaling



- ❑ Changing the sampling rate for the normalized design from 1Hz to  $f_s$
- ❑ DAC pulse-shape stretches by  $T_s$
- ❑ loop-filter scales as

$$\frac{1}{s} \rightarrow \frac{f_s}{s} = \frac{1}{sT_s}$$

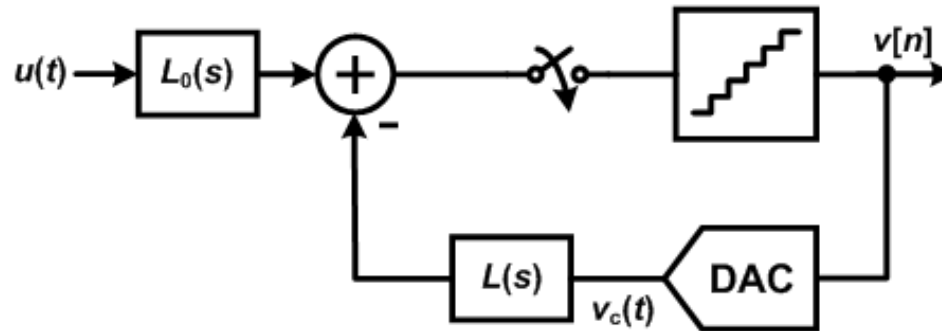
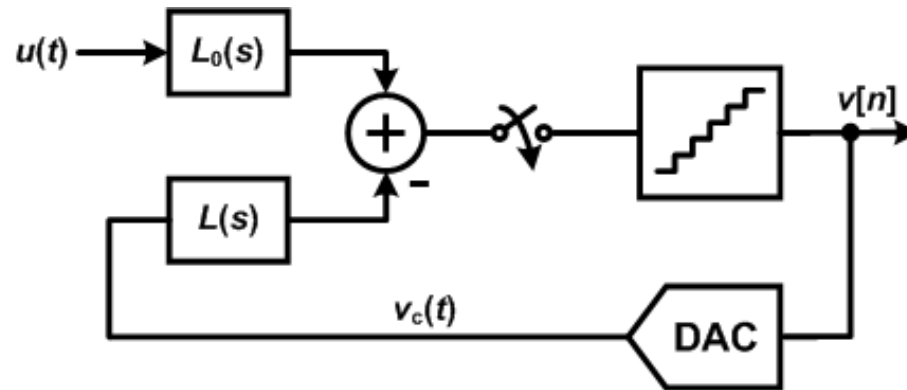
# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators



- Recall that the loop-filter may appear different to the input signal  $u(t)$ , and the feedback signal,  $v_c(t)$ .

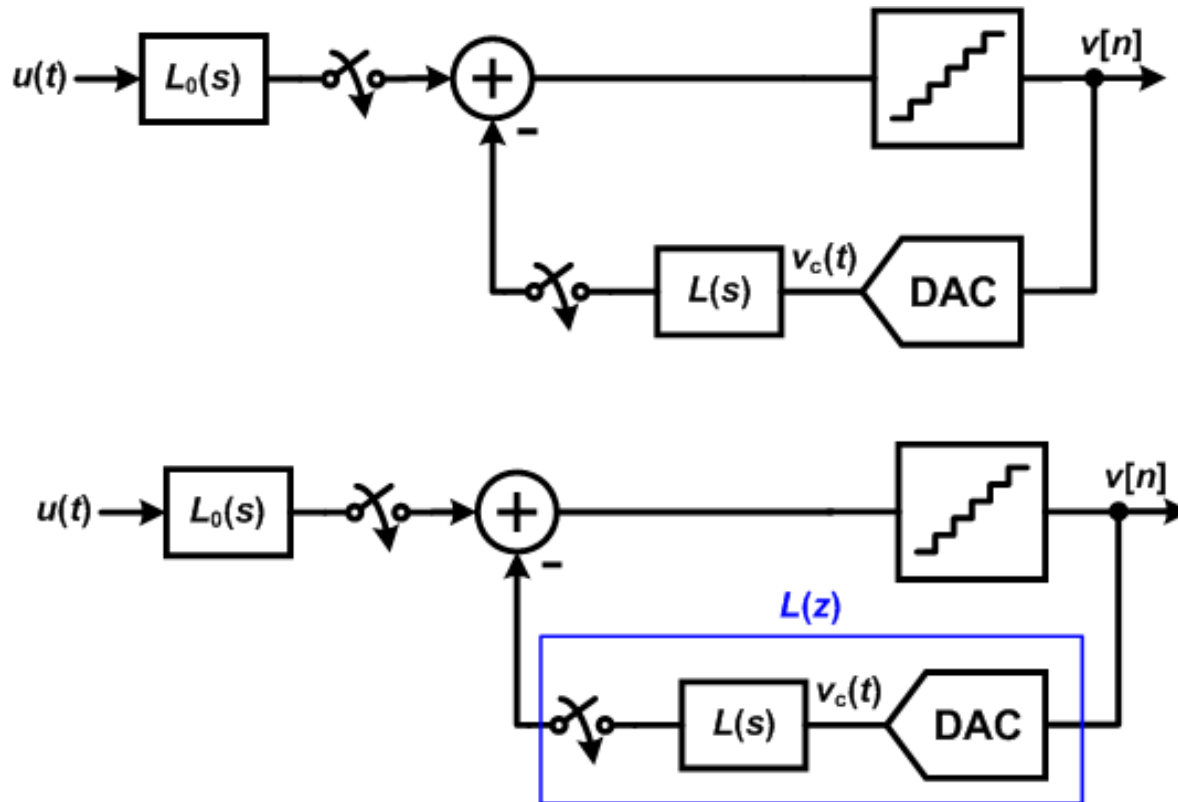


# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators



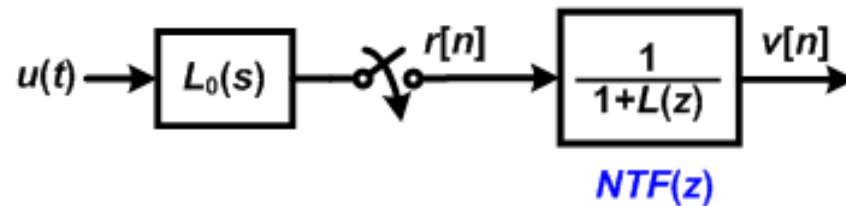
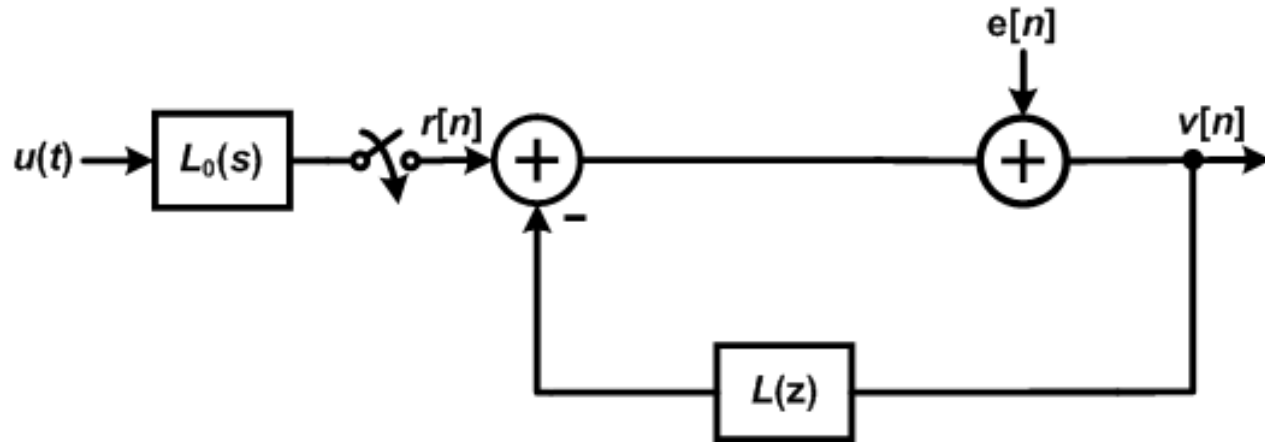
- Re-arrange the blocks
  - Recall that  $L_1(s) = -L(s)$  due to the negative feedback inversion

# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators



- Move the sampler outside the loop
  - Apply impulse-invariant transformation on the feedback path
- $$\mathbf{Z}^{-1}\{L(z)\} = \mathbf{L}^{-1}\{P(s) \cdot L(s)\} \Big|_{t=nT_s}$$

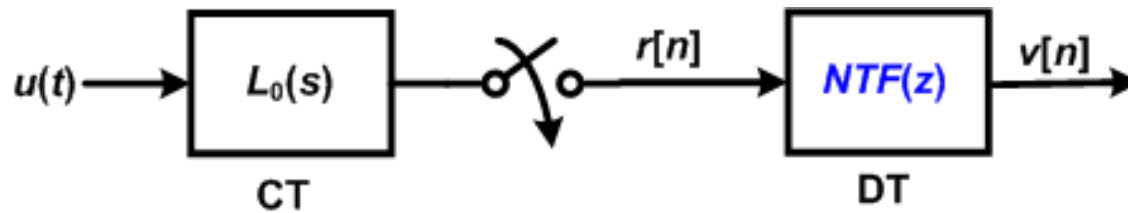
# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators



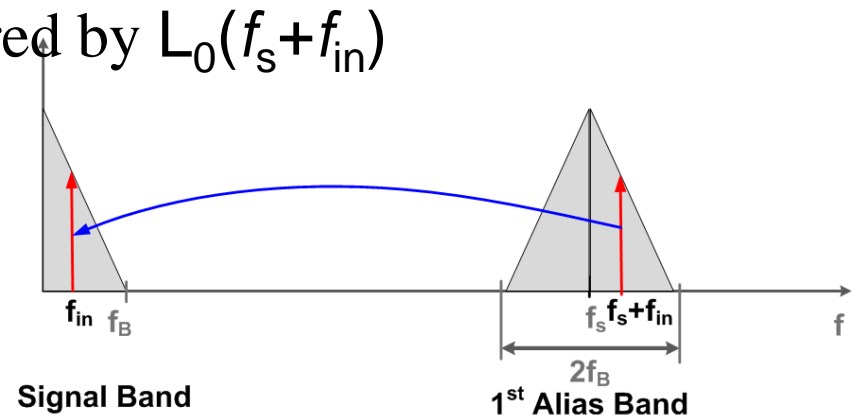
- Transfer function from  $r$  to  $v$  is given by

$$NTF(z) = \frac{1}{1+L(z)}$$

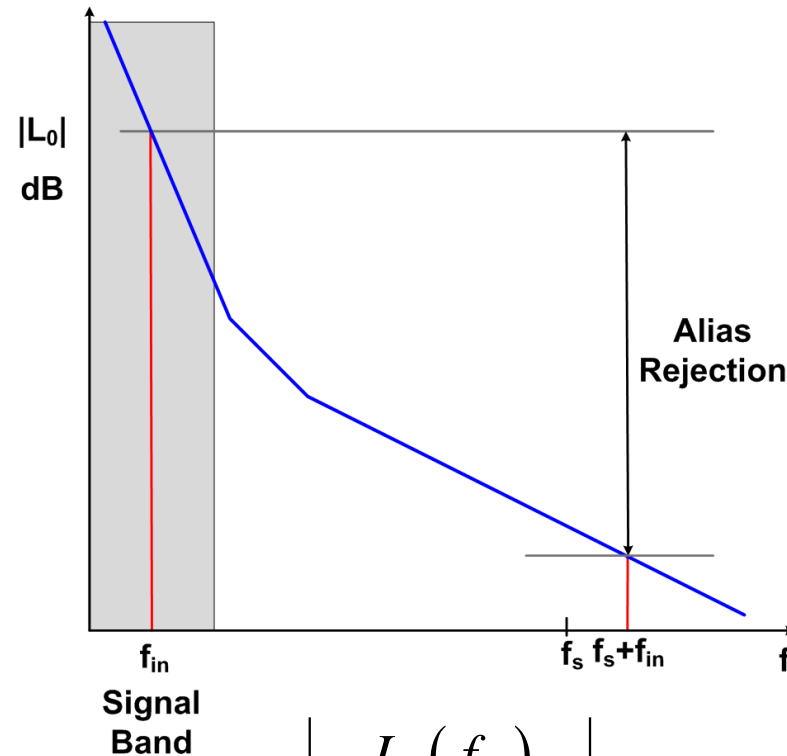
# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators



- The input signal is pre-filtered by a CT LPF,  $L_0(s)$ , before sampling
- For an input tone at frequency  $f_{in}$  in the signal band
- Pre-filtered output before sampling is given by  $L_0(f_{in})$
- Due to sampling, a tone at  $f_s + f_{in}$  can alias at  $f_{in}$ 
  - i.e. the first alias-band  $[f_s - f_B, f_s + f_B]$
- In CT  $\Delta\Sigma$ , the alias tone is filtered by  $L_0(f_s + f_{in})$

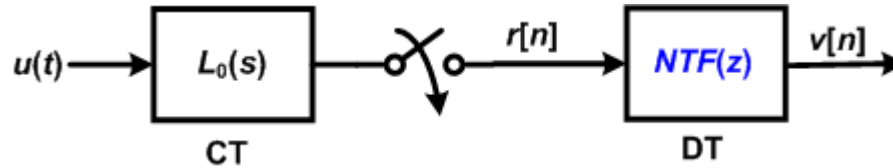


# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators



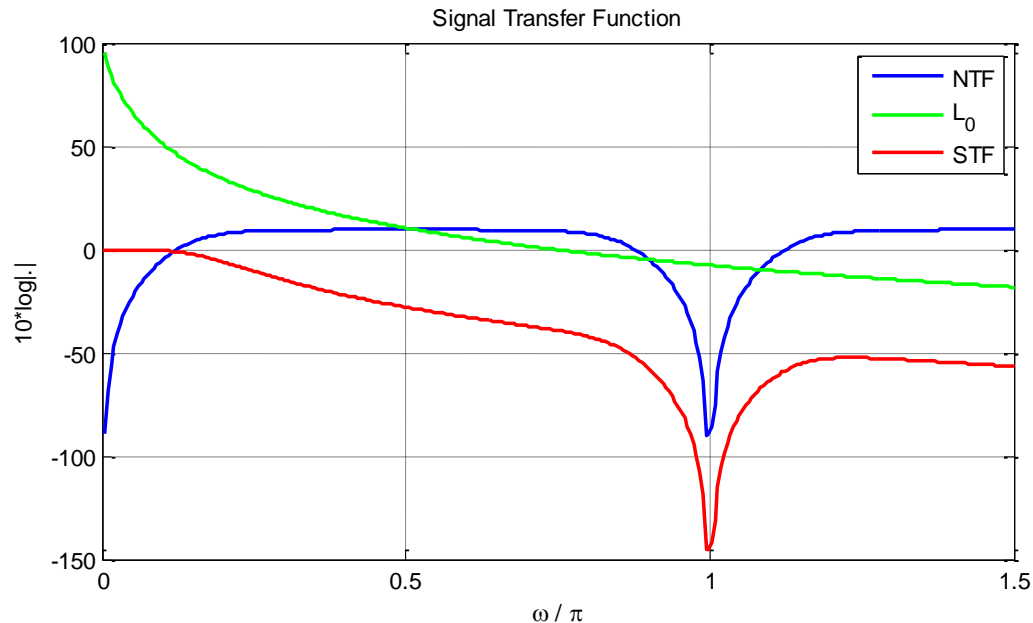
- Alias rejection is given by  $\left| \frac{L_0(f_{in})}{L_0(f_s + f_{in})} \right|$
- Implicit anti-aliasing due to pre-filtering by  $L_0$ 
  - Explicit AAF can be eliminated!
- Important distinguishing feature of CT  $\Delta\Sigma$

# Implicit Anti-Aliasing in CT $\Delta\Sigma$ Modulators

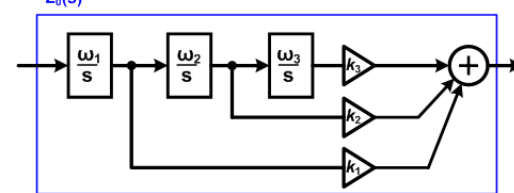
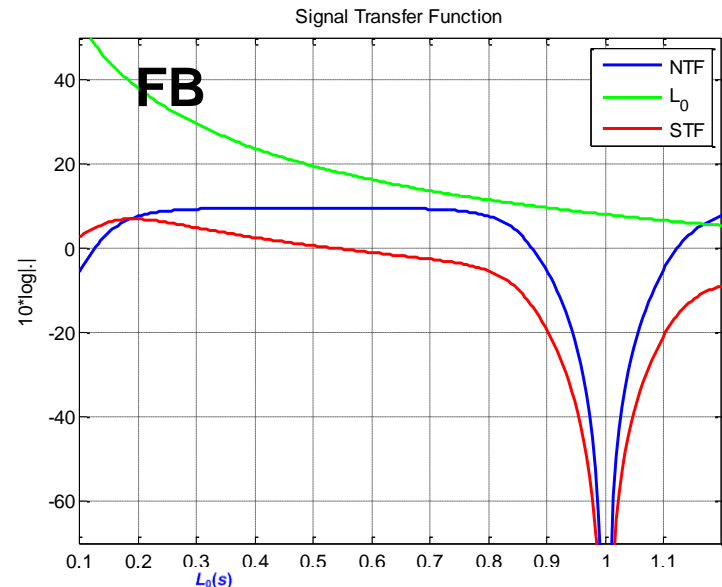
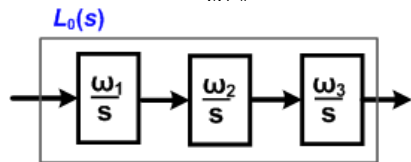
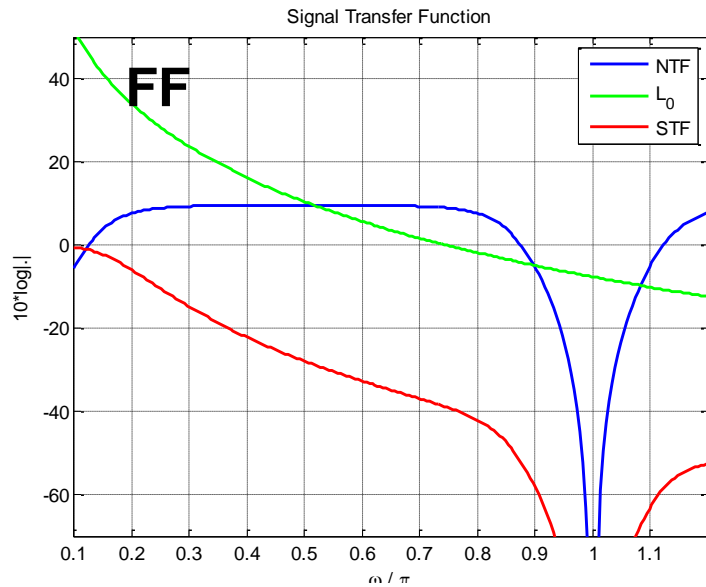


- Overall signal transfer function (STF) is given by

$$STF(j\omega) = L_0(j\omega) \cdot NTF(e^{j\omega})$$

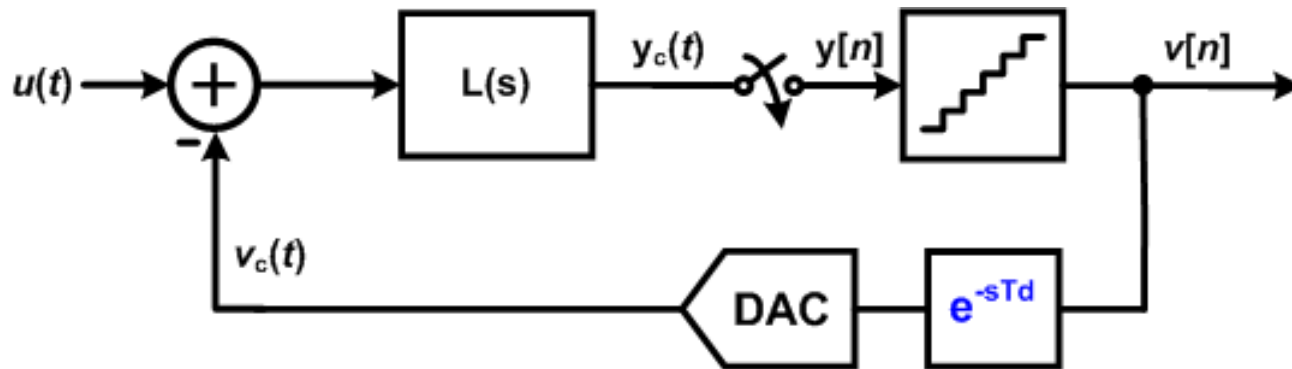


# Implicit AAF – FF vs FB Loop-Filter



- ❑ Feedforward loop-filter exhibits STF peaking
- ❑ STF peaking due to zeroes in  $L_0(s)$ 
  - Zeros at higher frequency flatten  $|L_0(j\omega)|$  response
- ❑ For FB,  $L_0(s)$  has all-pole form
  - Best STF performance (blocker rejection)

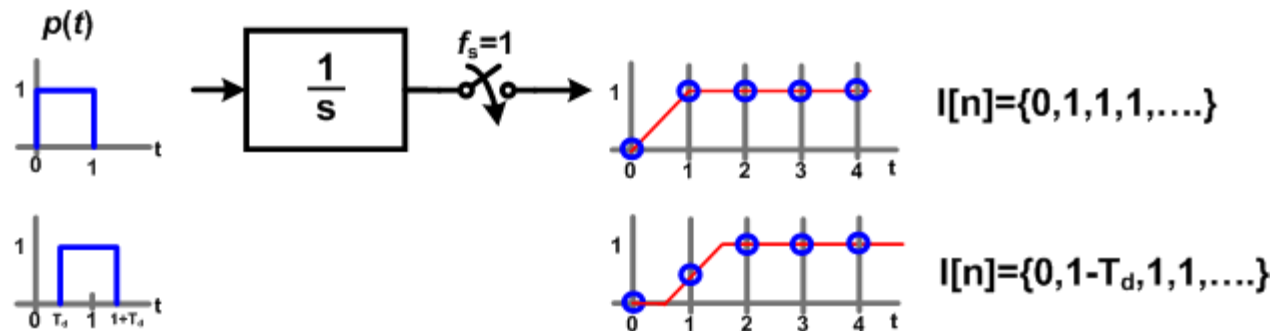
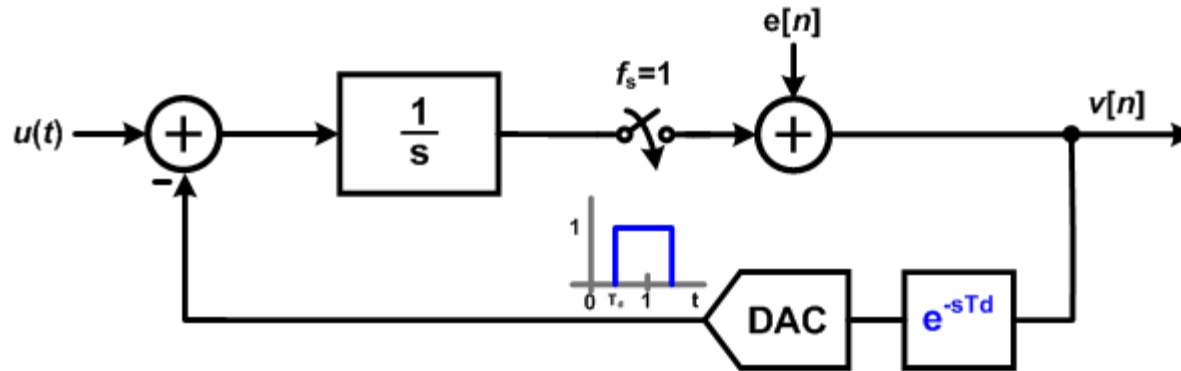
# Excess Loop-Delay in CT $\Delta\Sigma$



- Real circuits blocks introduce excess-loop delays (ELD)
  - Quantizer requires finite regeneration time
  - ELD due to finite gain-bandwidth of opamps in the loop-filter
  - DEM/DWA logic in multi-bit  $\Delta\Sigma$  adds finite delay
  - Finite DAC rise/fall time (relatively small)
- Important concern in high-sampling rate  $\Delta\Sigma$  converters
- What effect does ELD have on the loop-response?



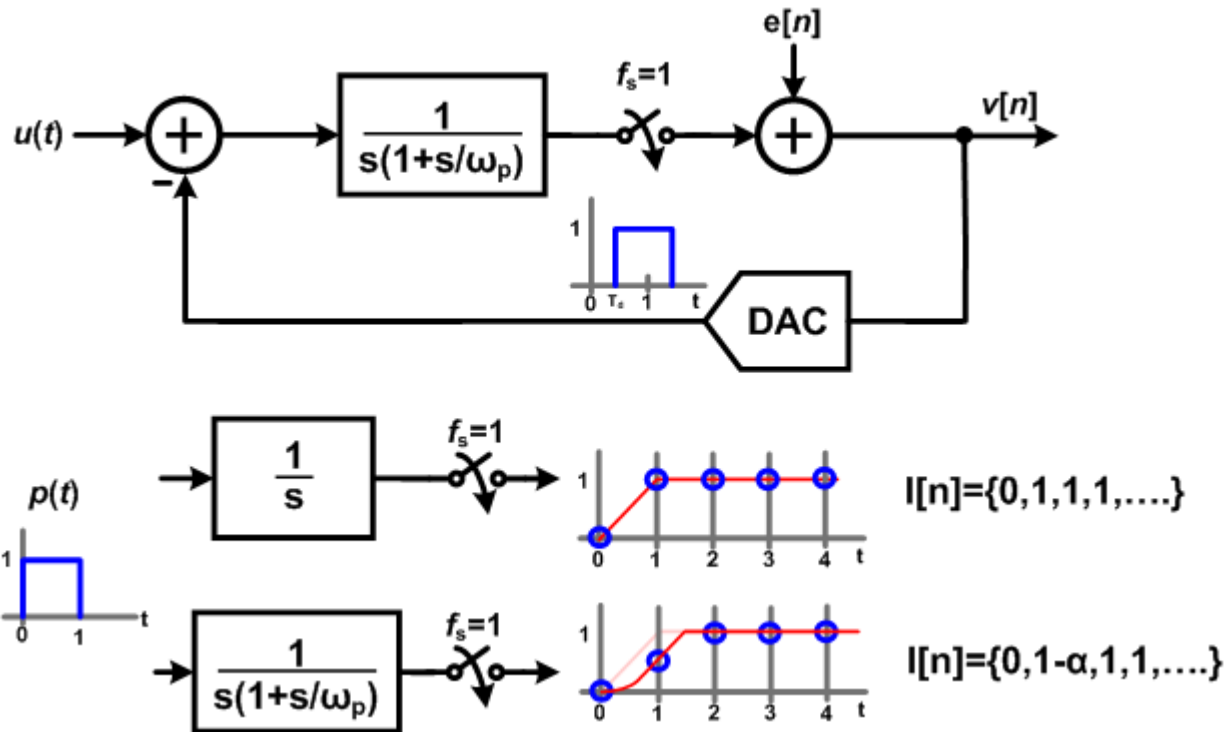
# ELD – First-order Example



- $f_s = 1$  Hz, ELD due to quantizer delay:  $0 < T_d < 1$
- First sample after unit delay is affected ( $1 - T_d$ )
- Resulting DT equivalent loop-response

$$L(z) = \frac{z^{-1}}{1 - z^{-1}} - T_d z^{-1}$$

# ELD – First-order Example



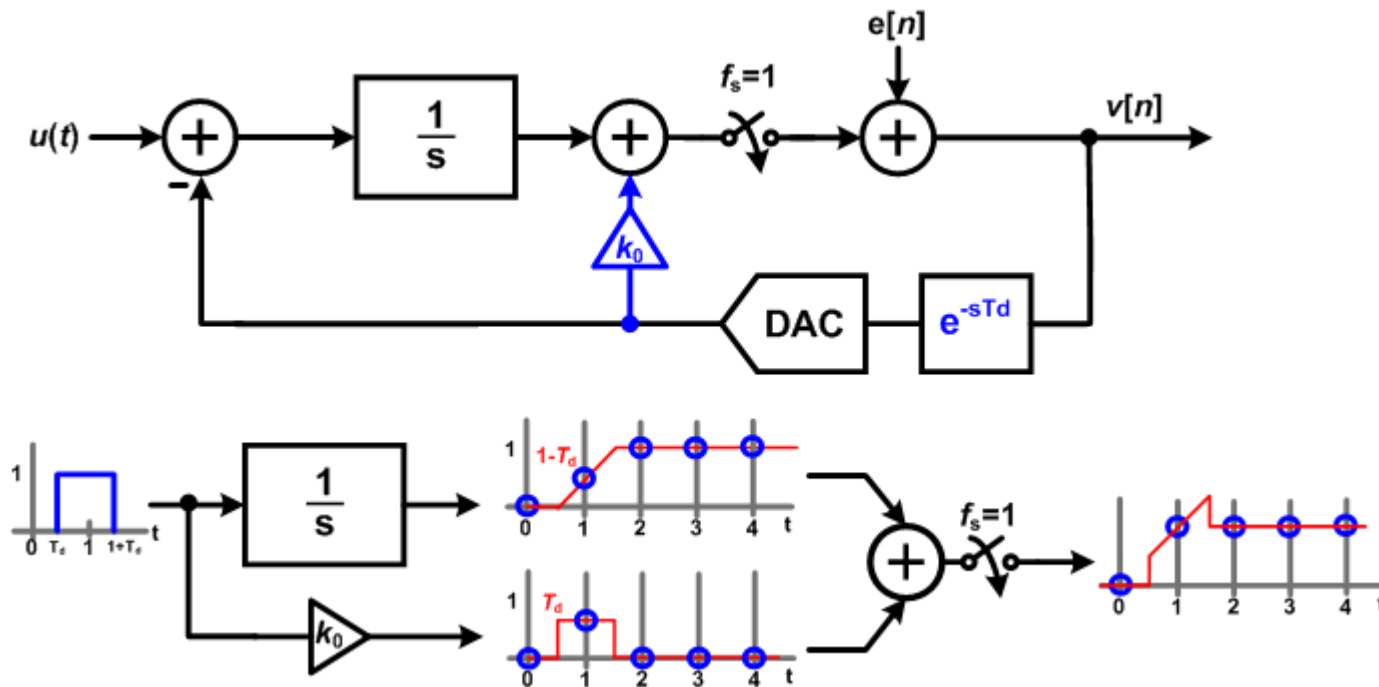
- ❑  $f_s=1$  Hz
- ❑ Sampled loop-response is delayed due to the finite gain-bandwidth of the opamp
- ❑ First sample after unit delay is affected  $(1-T_d)$

# Excess Loop-Delay in CT $\Delta\Sigma$

□ NTF with ELD 
$$NTF(z) = \frac{(1 - z^{-1})}{1 - T_d z^{-1} + T_d z^{-2}}$$

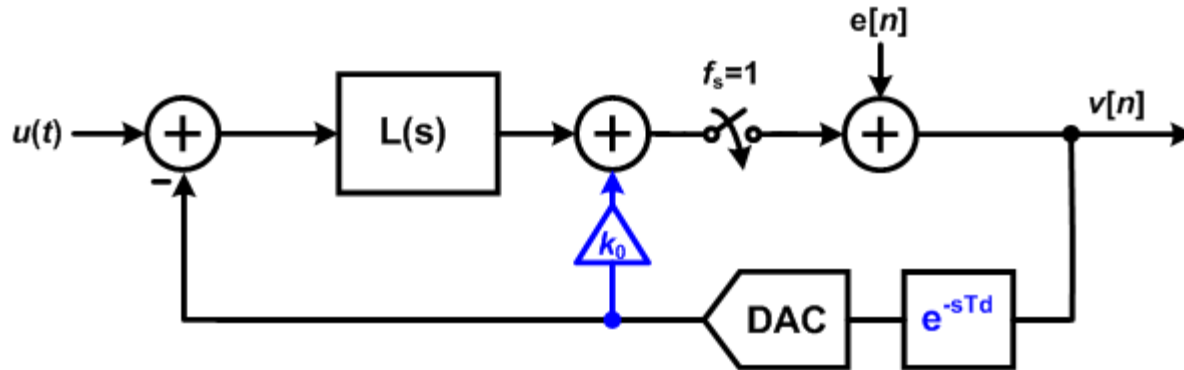
- Order of the loop is increased due to the extra delay
- Loop gets unstable at  $T_d=1$
- NTF peaking as  $T_d$  increases, leading to modulator becoming more sensitive and prone to instability

# ELD Compensation– Basic Idea



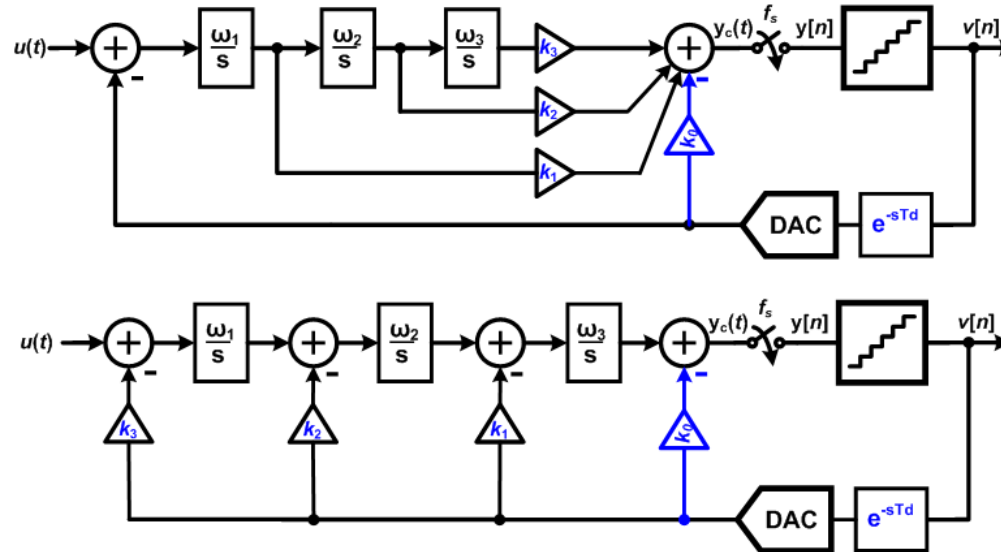
- ❑ Add a direct path ( $k_0$ ) around the quantizer to provide the first sample  $l[1]$
- ❑ Here,  $k_0 = T_d$  restores  $L(z)$ 
  - modulator NTF( $z$ ) is restored!
- ❑ Several other solutions are possible to restore  $L(z)$

# ELD Compensation



- ❑ Normalized ELD due to quantizer delay  $= T_d/T_s = \tau$
- ❑ A direct path around the quantizer ( $k_0$ ) is used to restore the DT loop-response  $L(z)$
- ❑ Approach valid for higher-order modulators
  - For order  $> 1$ ,  $L(s)$  is also modified (requires coefficient tuning)
- ❑ Schreier's  $\Delta\Sigma$  Toolbox function `realizeNTF_ct`
  - can synthesize CT loop-filters with  $ELD < 1$  and rectangular DAC pulse response.

# ELD Compensation



- For  $n^{\text{th}}$  order modulator, the loop-filter coefficients also need to be tuned. For NRZ DAC:

$$k'_0 = k_1\tau + \frac{k_1}{2!}\tau^2 + \dots + \frac{k_n}{n!}\tau^n = \sum_{i=1}^n \frac{k_i}{i!}\tau^i$$

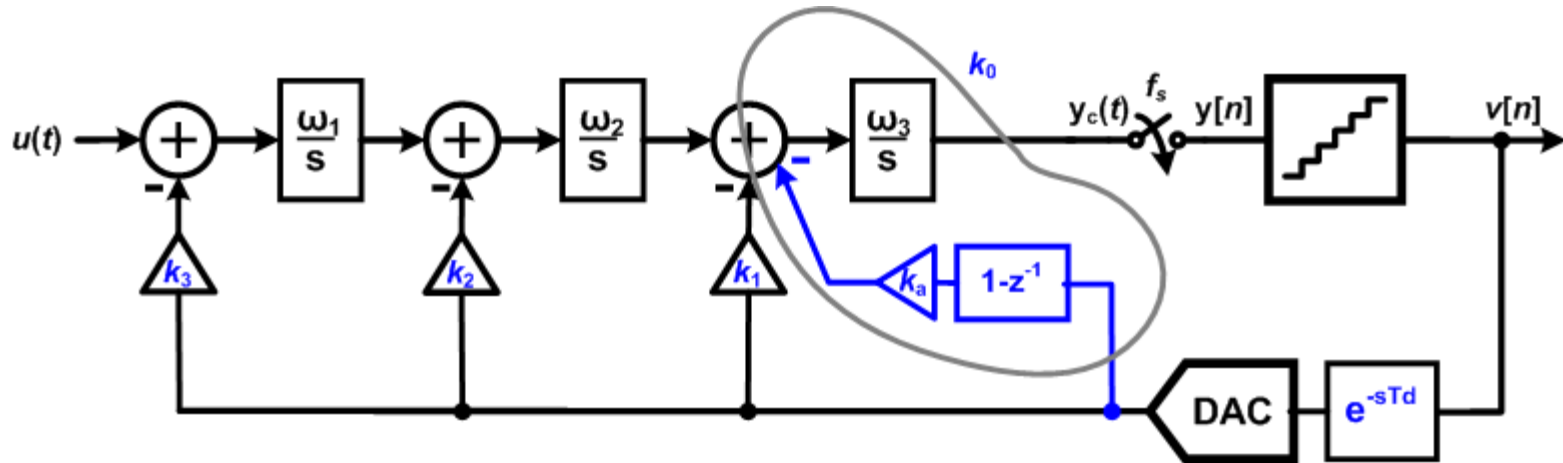
$$k'_1 = k_1 + k_2\tau + \dots + \frac{k_n}{n-1!}\tau^{n-1} = \sum_{i=1}^n \frac{k_i}{(i-1)!}\tau^i$$

⋮

$$k'_n = k_n$$

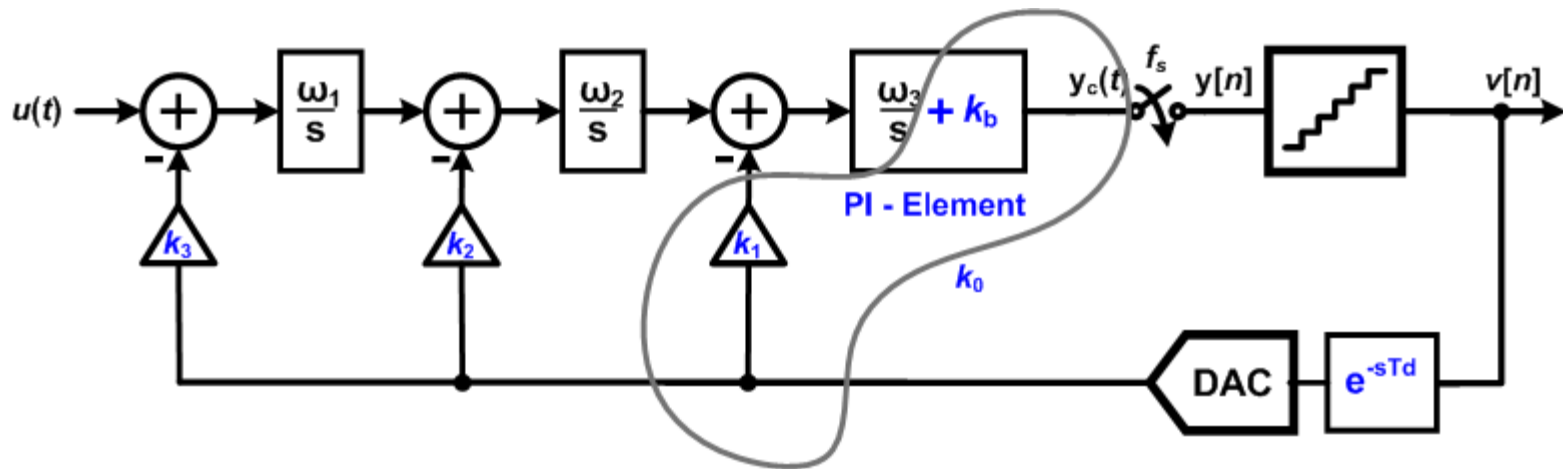
- Coefficient tuning is better done numerically.

# ELD Comp. using DT Differentiator



- ❑ Proportional path = CT integrator + DT differentiator
- ❑ Avoids the extra summer at the expense of a register
- ❑ Scaling factor  $k_a = k_0 / \omega_3$
- ❑ Other variations include
  - Using a  $(1-z^{-0.5})$  differentiator,  $k_a = k_0 / 2\omega_3$
  - Can tap first integrator output and perform CT differentiation

# ELD Comp. using a PI-Element



- ❑ Combine the two inner loops into a single PI-element
- ❑ Note that  $k_b$  affects other inner loops too
  - Other coefficients need to be tuned accordingly
- ❑ Can aggravate STF peaking due to additional zero

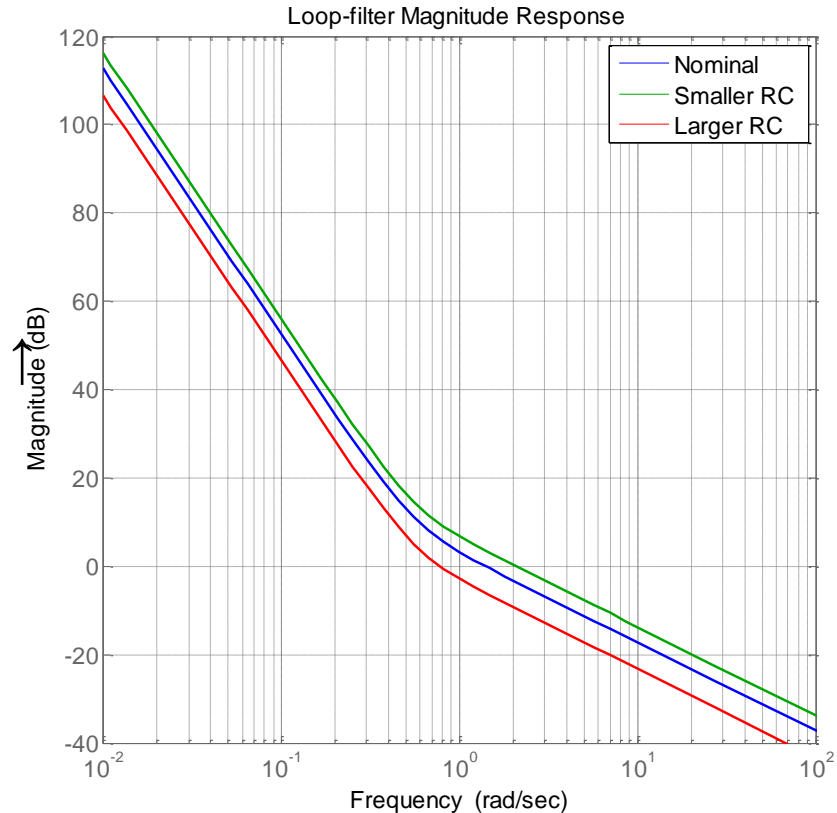


# Effect of RC Variation

- CT- $\Delta\Sigma$  loop-filter coefficients  $k_i$  are implemented as  $1/RC$  (or  $g_m/C$ )
- The RC time-constants can vary with process and temperature by
  - $\pm 1\%$  on a single die
  - $\pm 20\%$  with from die-to-die
- DT- $\Delta\Sigma$  employs implements  $k_i$  using ratio of capacitors
  - Much tightly controlled on-chip and hence accurate
- What is the impact of RC time-constant variation on CT- $\Delta\Sigma$  modulator performance?

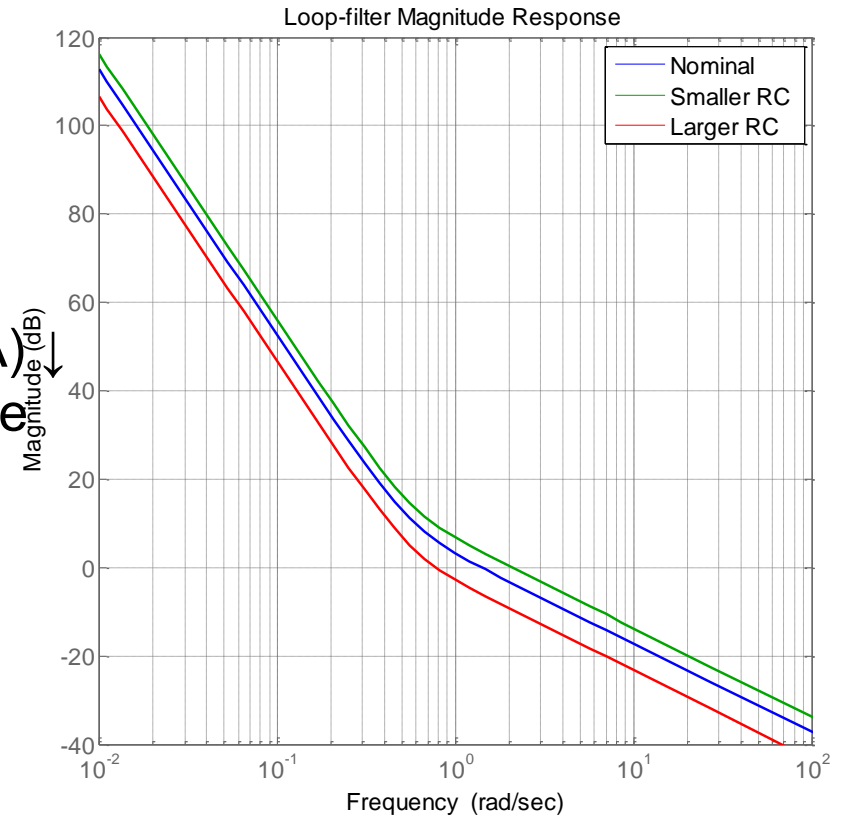
# RC Time-Constant Variation

- With an increase in RC time-constant
  - Coefficients  $k_i$  decrease (↓)
  - Loop-filter bandwidth ↓
  - In-band loop-gain ↓
  - NTF OBG ↓
  - In-band quantization noise ↑
  - Maximum stable amplitude (MSA) ↑
  - “More” stable but degraded NTF performance
  - Slower design



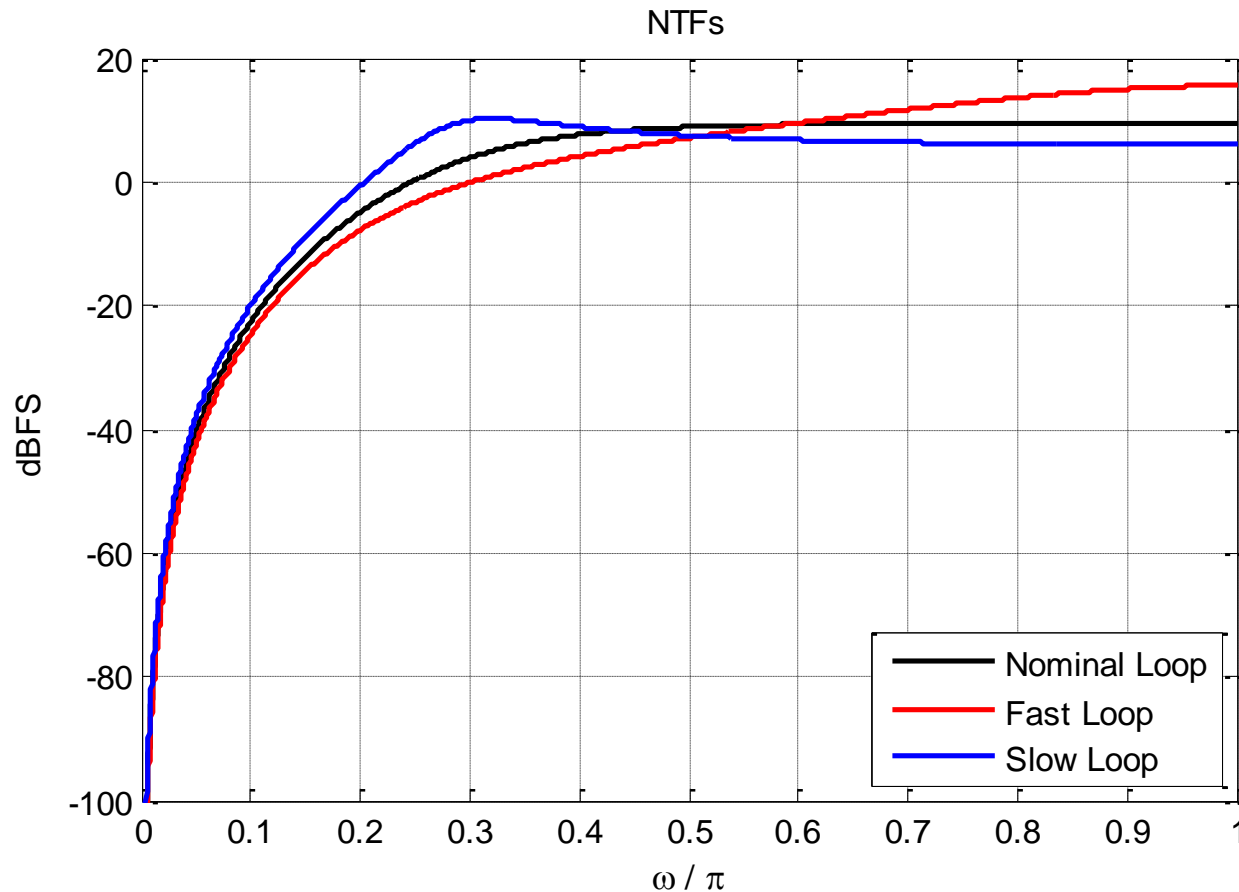
# RC Time-Constant Variation

- With a decrease in RC time-constant
  - Coefficients  $k_i$  increase ( $\uparrow$ )
  - Loop-filter bandwidth  $\uparrow$
  - In-band loop-gain  $\uparrow$
  - NTF OBG  $\uparrow$
  - In-band quantization noise  $\downarrow$
  - Maximum stable amplitude (MSA)  $\downarrow$
  - “Less” stable but more aggressive NTF performance
  - Faster design



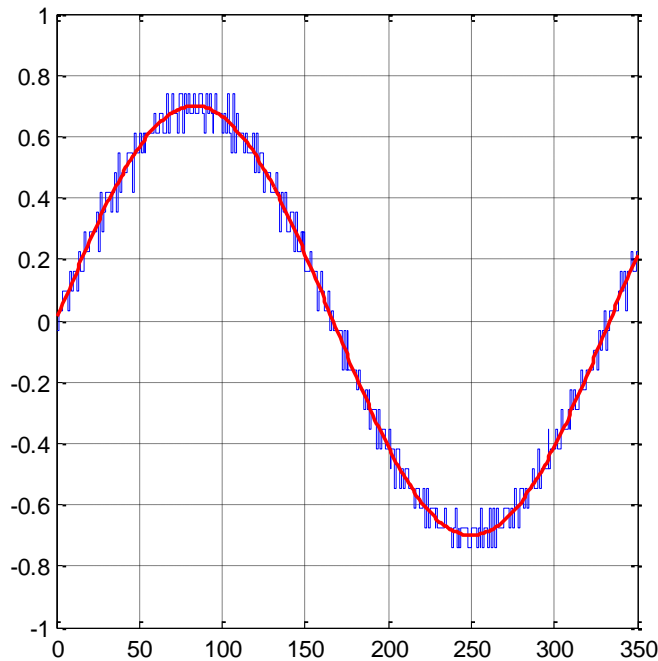
# RC Time-Constant Variation

- Maximally flat 3<sup>rd</sup>-order NTF with nominal OBG=3
  - Modulators with  $\pm 30\%$  RC variation

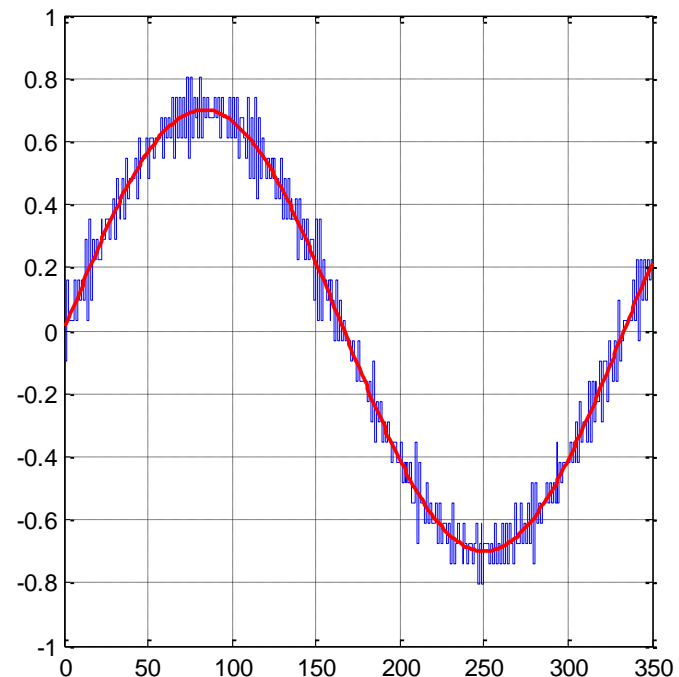


# RC Time-Constant Variation

- Maximally flat 3<sup>rd</sup>-order NTF with nominal OBG=3, nLev=2<sup>3</sup>
  - Modulators with  $\pm 30\%$  RC variation

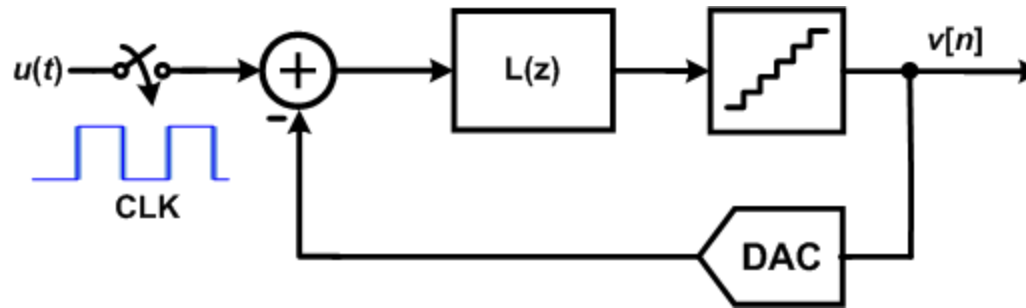


Nominal Loop



Fast Loop

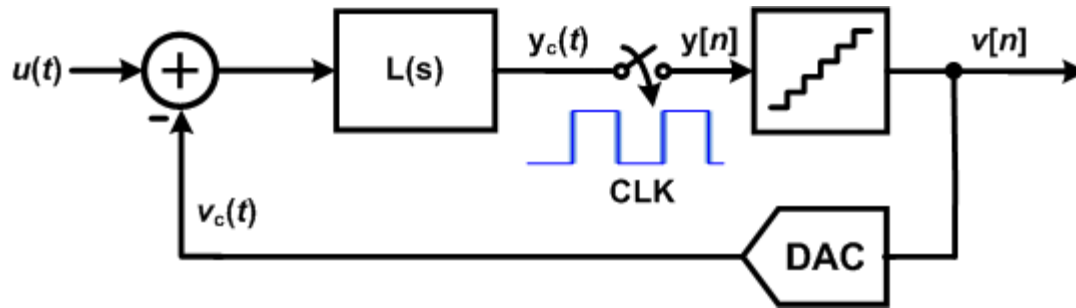
# Clock Jitter in DT- $\Delta\Sigma$



- DT- $\Delta\Sigma$  the input is sampled before the modulator loop
- Assume white clock jitter with RMS value  $\sigma_j$
- RMS jitter noise in the signal band:  $\frac{\sigma_j A \omega_{in}}{OSR \sqrt{2}}$
- Clock jitter limited SNR:

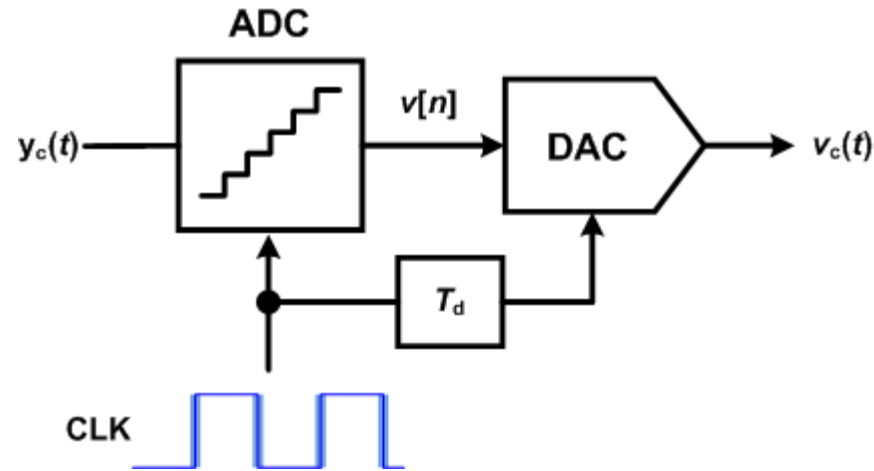
$$SNR_j = -20 \cdot \log_{10} (2\pi f_{in} \sigma_j OSR)$$

# Clock Jitter in CT $\Delta\Sigma$



- ❑ The input signal is sampled inside the modulator loop
- ❑ Clock jitter affects both ADC decisions as well as the DAC output

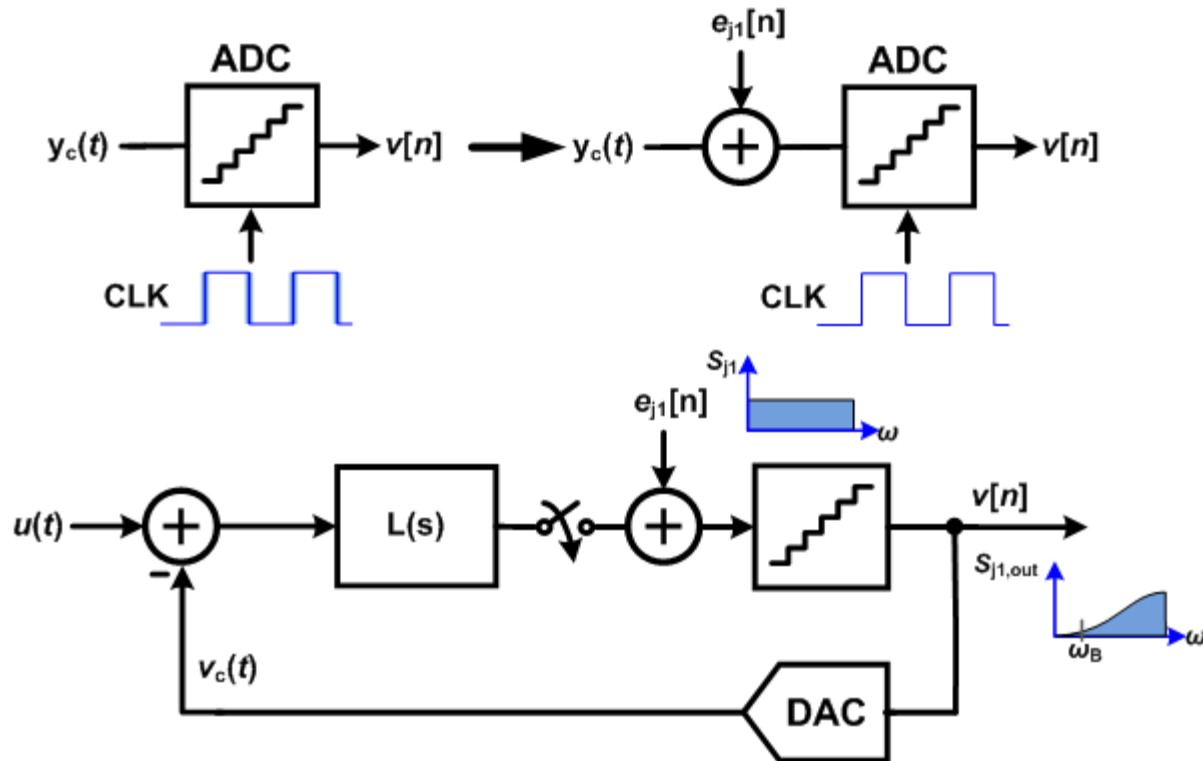
# Real Quantizer with Jittery Clock



- ❑ ADC samples and quantizes CT input  $y_c$  digital output  $v[n]$ 
  - ADC regeneration time -  $T_d$
- ❑ DAC reconstructs samples of  $v[n]$  into CT output –  $v_c$
- ❑ DAC is clocked with a delay to capture the ADC output
- ❑ Both ADC and DAC clocks have jitter

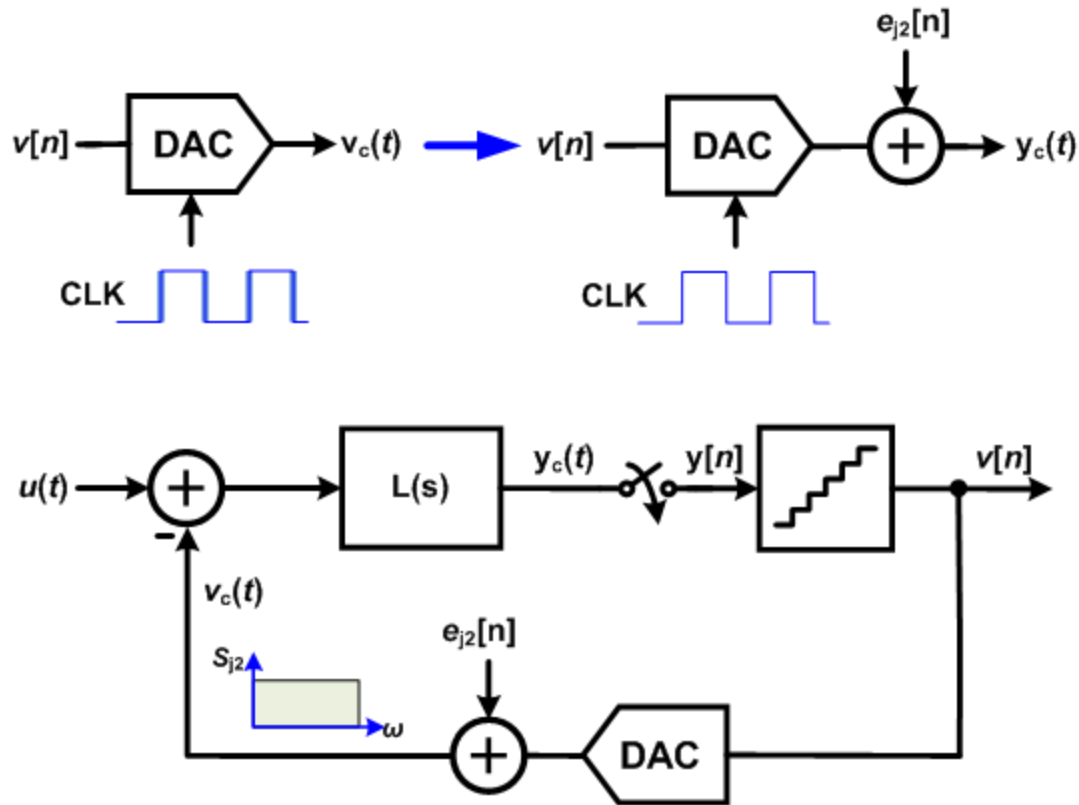


# ADC Sampling Jitter



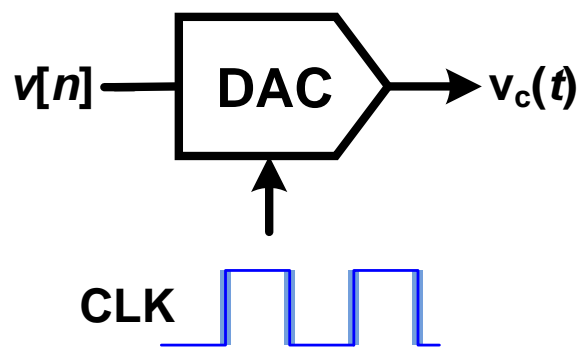
- ❑ ADC sampling jitter is modeled as AWGN
- ❑ Noise due to sampling jitter is shaped by the loop (NTF)

# DAC Reconstruction Jitter

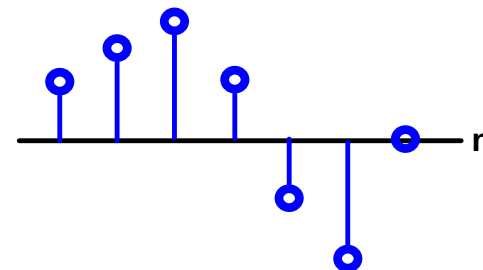


- ❑ DAC reconstruction error ( $e_{j2}$ ) follows the DAC
- ❑ Directly adds to the input
- ❑ Major contributor of jitter induced noise

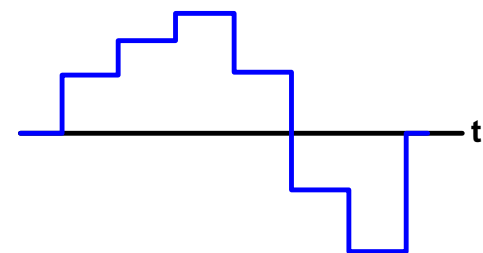
# DAC Output Waveforms



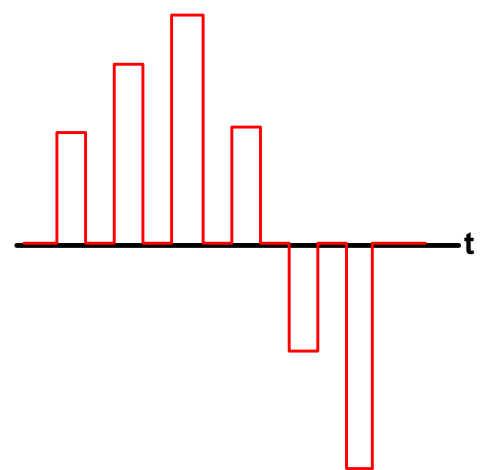
$v[n]$



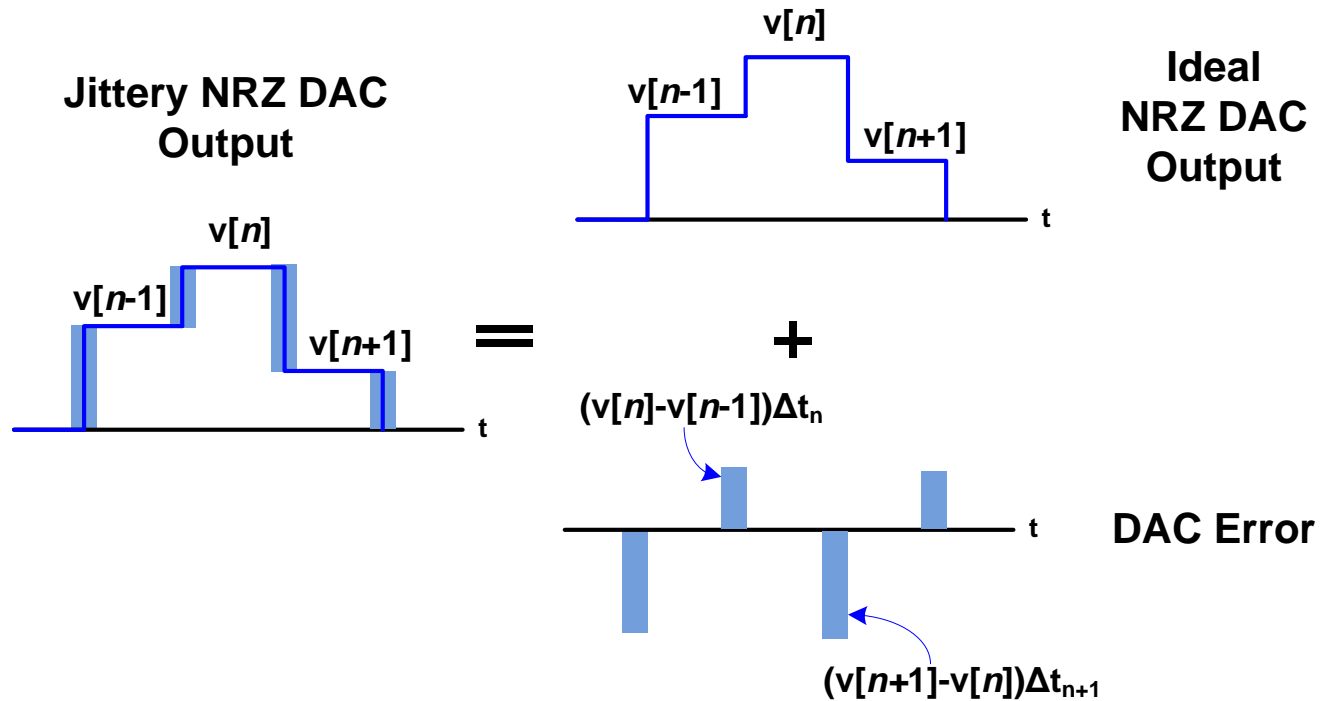
$v_c(t)$   
NRZ DAC



$v_c(t)$   
RZ DAC

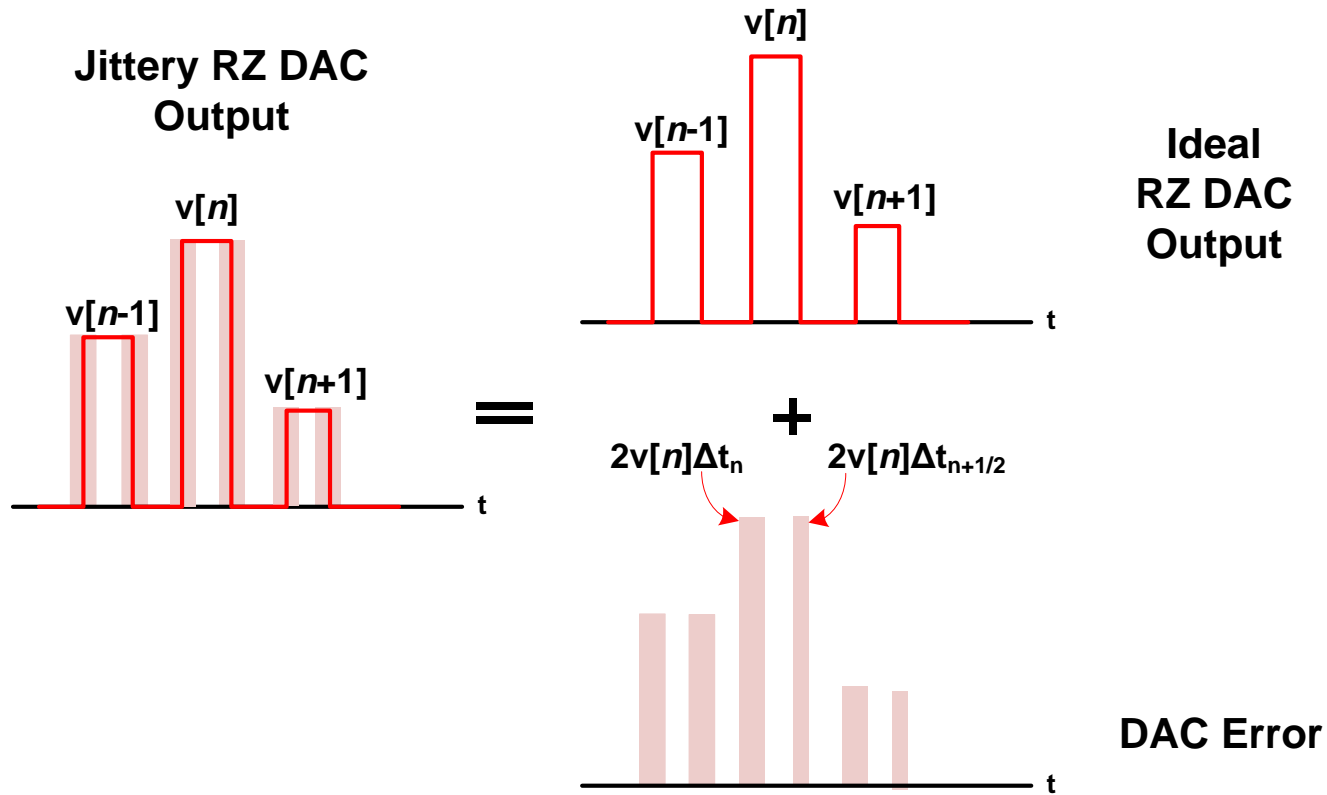


# Jitter Induced DAC Error- NRZ



- ❑ Error depends on the transition height  $v[n]-v[n-1]$ 
  - Granularity of LSB size
- ❑ Multi-bit NRZ DACs have lower jitter induced error

# Jitter Induced DAC Error- RZ



- ❑ Error depends on twice the pulse height  $2v[n]$ 
  - Two transitions per time period
- ❑ RZ DACs are much more sensitive to clock jitter

# Jitter Noise with NRZ DAC

- Error due to jitter  $e_j[n] = (v[n] - v[n-1]) \frac{\Delta t_n}{T} = \delta v[n] \frac{\Delta t_n}{T}$

$$\sigma_{ej}^2 = \sigma_{\delta v}^2 \cdot \frac{\sigma_{\Delta t}^2}{T_s^2}$$

- Modulator output variance (neglecting slowly varying input)

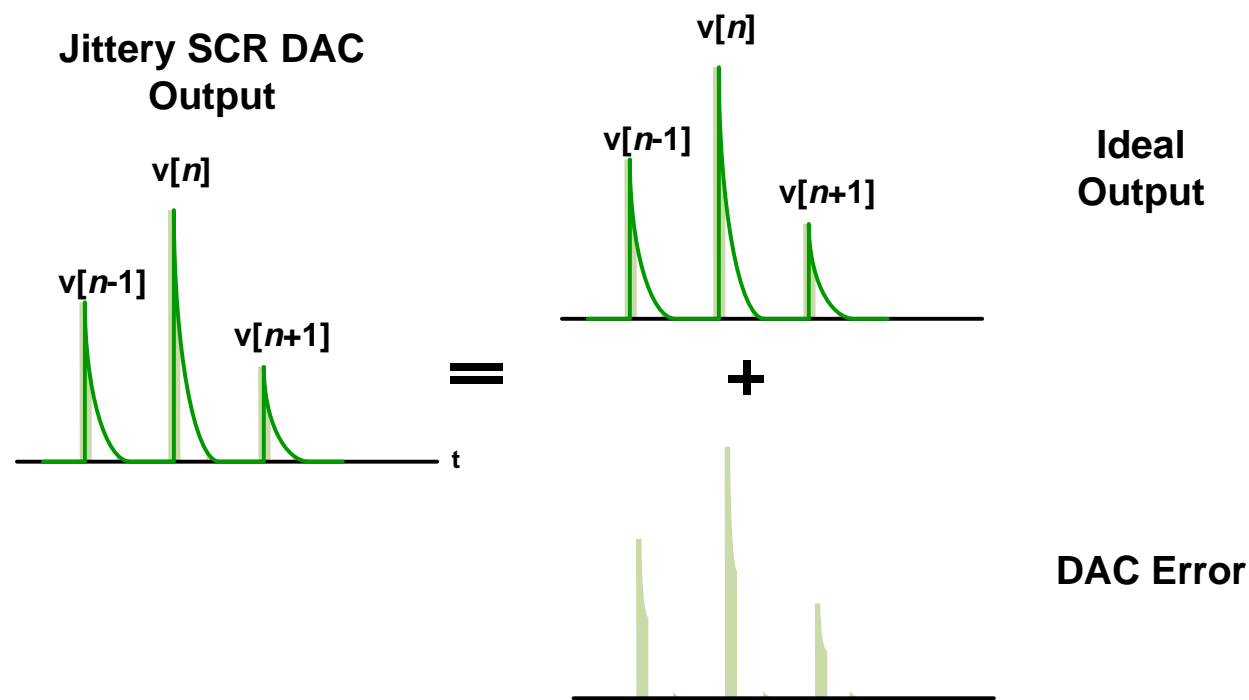
$$\sigma_{\delta v}^2 \square \left( \frac{\sigma_{LSB}^2}{\pi} \right) \int_0^{\pi} \left| (1 - e^{-j\omega}) NTF(e^{j\omega}) \right|^2 d\omega$$

- In-band jitter noise

$$J \square \frac{\sigma_{\Delta T_s}^2}{T_s^2} \cdot \frac{\sigma_{LSB}^2}{\pi \cdot OSR} \int_0^{\pi} \left| (1 - e^{-j\omega}) NTF(e^{j\omega}) \right|^2 d\omega$$

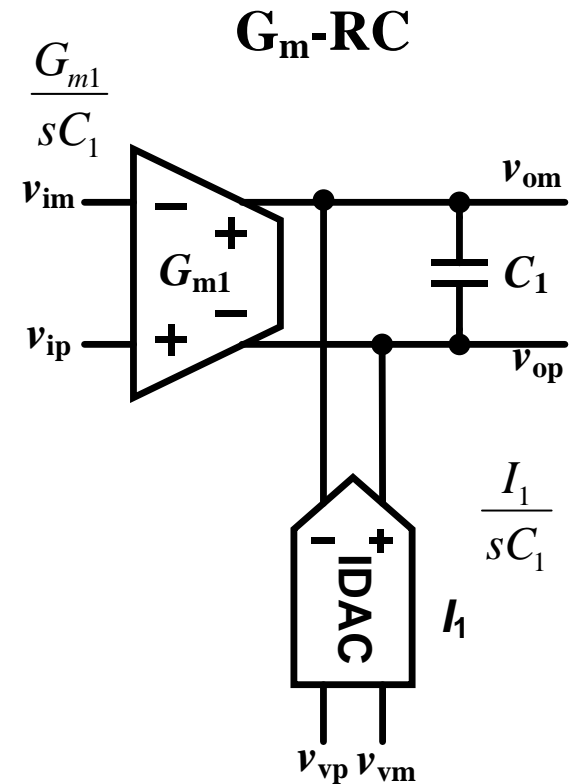
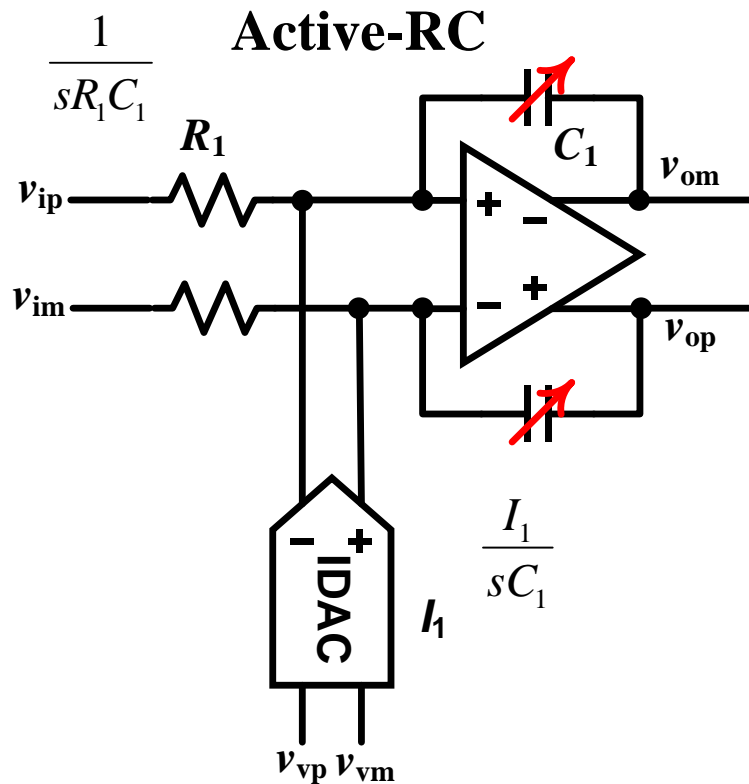
- depends upon the area of differentiated NTF response
- NTF behavior at higher frequencies determines most of  $J$
- To reduce  $J$ :  $OBG \downarrow$ ,  $V_{LSB} \downarrow$ ,  $\frac{\sigma_{\Delta T_s}}{T_s} \downarrow$ ,  $OSR \uparrow$

# SCR DAC for Lower Jitter Noise



- Use sloping DACs to reduce the error due to jitter
  - Switched-cap resistor DAC (SCR), cosine-shape, etc.
- More stringent requirements on the opamp performance
  - Possible linearity and AAF degradation

# CT $\Delta\Sigma$ Implementation



- CT Integrators used as the loop-filter stages
  - **Active-RC**,  **$G_m$ -C**, Active  $G_m$ -C, Active-MOS-C, Current-mode integrators, log-domain integrators
  - Active-passive Hybrid stages can alternate

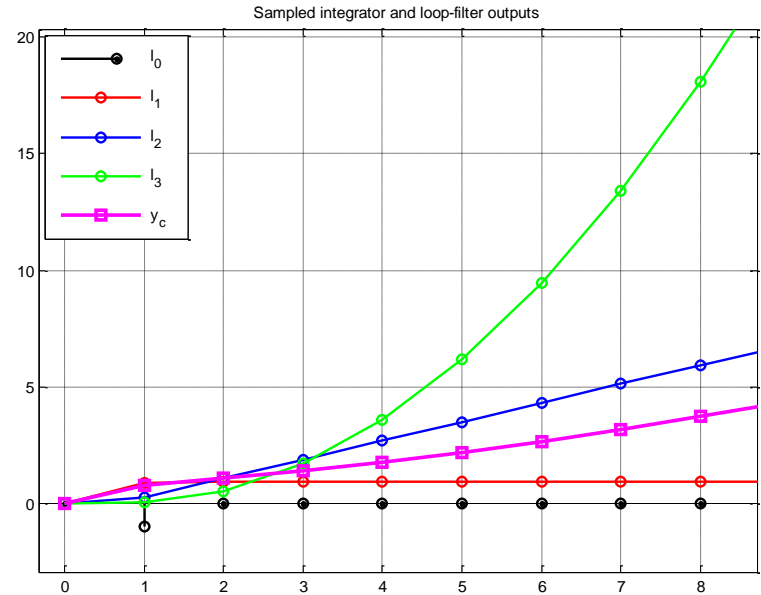
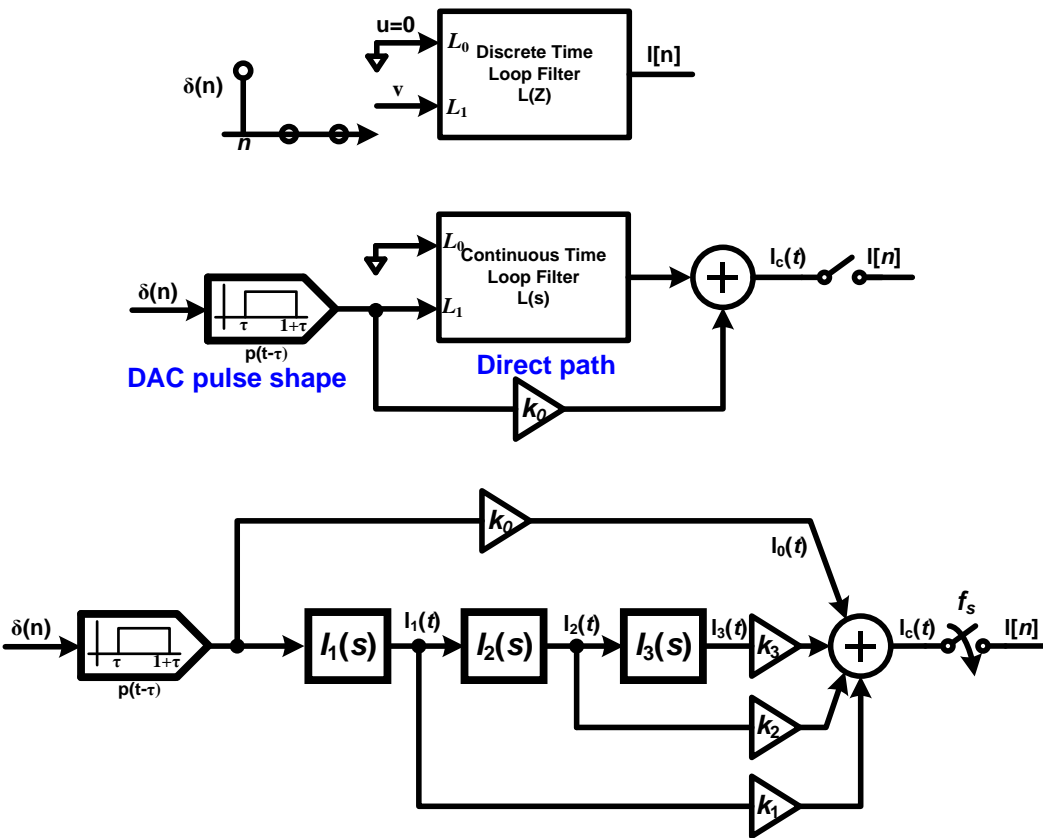


# CT $\Delta\Sigma$ Implementation

|                             | Active-RC | $G_m$ -C |
|-----------------------------|-----------|----------|
| Linearity                   | ✓         | ✗□       |
| Power consumption           | ✗□        | ✓        |
| Frequency range             | ✗□        | ✓        |
| Tunability                  | ✗□        | ✓        |
| Dynamic Range               | ✓         | ✗□       |
| Voltage headroom            | ✓         | ✗□       |
| Parasitic Sensitivity       | ✓         | ✗□       |
| Realization of FF Summation | ✗□        | ✓        |

- Overall Active-RC preferred for superior dynamic range in CT  $\Delta\Sigma$  designs
- Active-RC with two- or higher stage opamp for better linearity

# Loop-filter Coefficient Tuning



# Loop-filter Coefficient Tuning

- Loop-filter coefficients are typically determined using the Schreier's  $\Delta\Sigma$  Toolbox
  - using impulse-invariance transformation tables is unwieldy for higher-order modulators and with ELD

## Algorithm

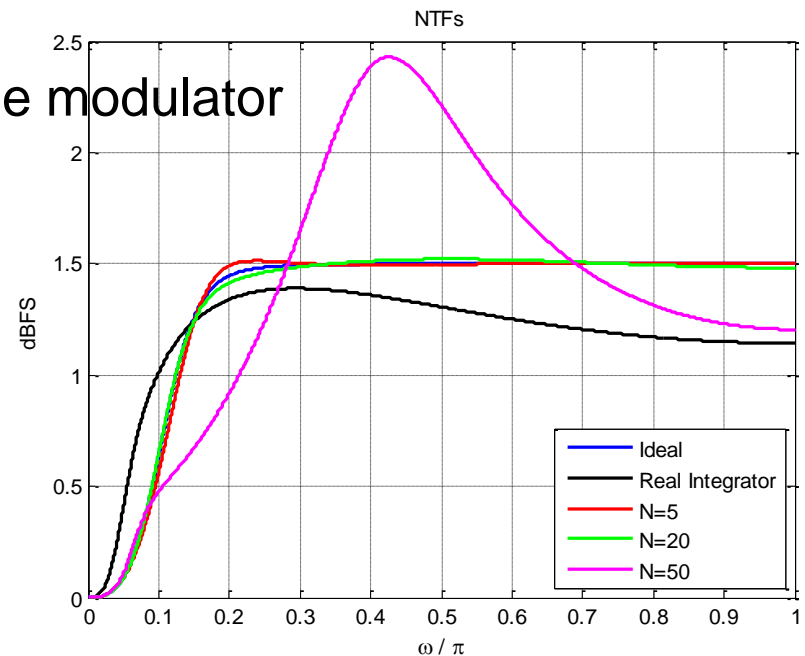
- If the sampled outputs of the direct path and the integrators are given by  $l_0[n]$ ,  $l_1[n]$ ,  $l_2[n]$  and  $l_3[n]$ , and the open-loop impulse response is  $l[n]$ . Here,  $Z(l[n]) = L(z) = \frac{1}{NTF(z)} - 1$ .
- The coefficients  $\mathbf{K} = [k_0 \ k_1 \ k_2 \ k_3]^T$  are determined by solving

$$[l_0[n] \ l_1[n] \ l_2[n] \ l_3[n]] \mathbf{K} = l[n]$$

- Solved using LMS data fitting for  $N$  samples (pseudo-inverse)

# Problems with Coefficient Tuning

- Practical integrators are implemented using opamps
  - finite opamp gain ( $A_{OL}$ ) and unity-gain bandwidth ( $f_{un}$ ), and with extra poles and zeros.
- The excess loop-delay due to finite  $f_{un}$  causes significant amount of gain peaking in the resulting NTF.
- Coefficient tuning yields different results depending on the number of samples used for LMS fitting.
  - tuning is numerically unstable
  - higher OBG leads to instability in the modulator



# Systematic Design Centering

- Instead of fitting the open-loop response, fit  $NTF(z)(1+L(z))$  to 1

$$\Rightarrow h[n] + h[n] \otimes l[n] = \delta[n] \qquad h[n] = Z^{-1}(NTF(z))$$

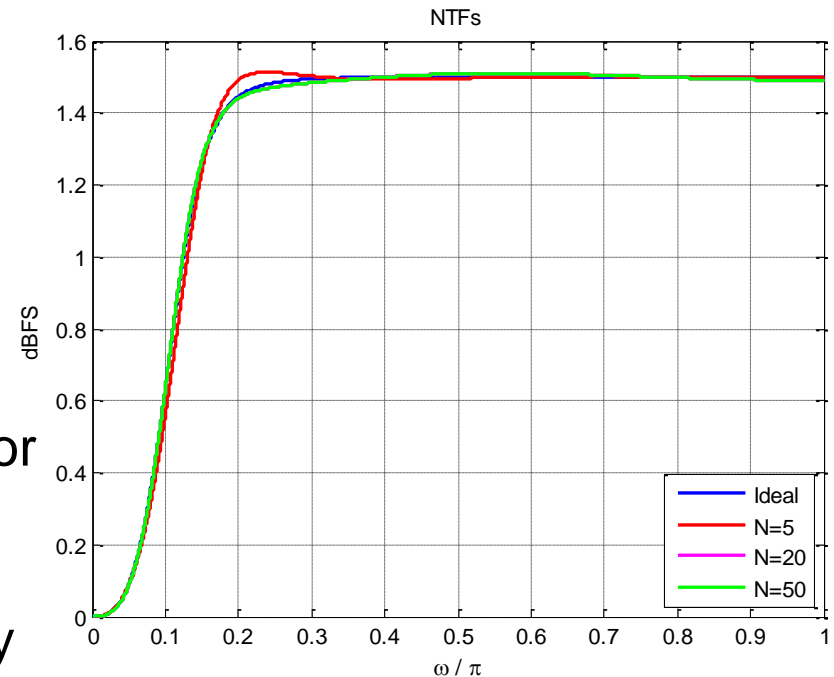
## Algorithm

- Let  $h_i[n] = l_i[n] \otimes h[n]$ , for  $i = 0, 1, \dots, 3$
- The coefficients  $\mathbf{K} = [k_0 \ k_1 \ k_2 \ k_3]^T$  are determined by solving

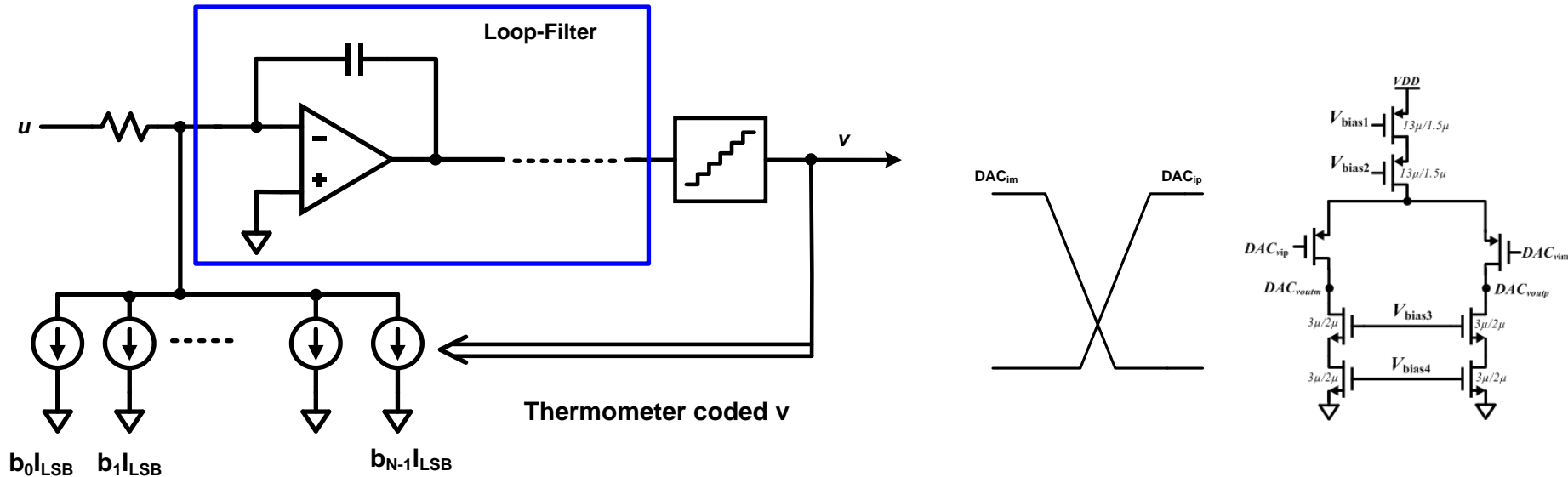
$$[h_0 \ h_1 \ h_2 \ h_3] \mathbf{K} = \delta[n] - h[n]$$

# Systematic Design Centering

- The loop-filter coefficients are tuned to compensate for the excess loop delay due to the op-amps and the quantizer delay
- NTF response with non-ideal integrators is close to the ideal NTF
- The NTFs are indistinguishable for any value of  $N=5,20,50$
- Coefficient tuning is numerically stable

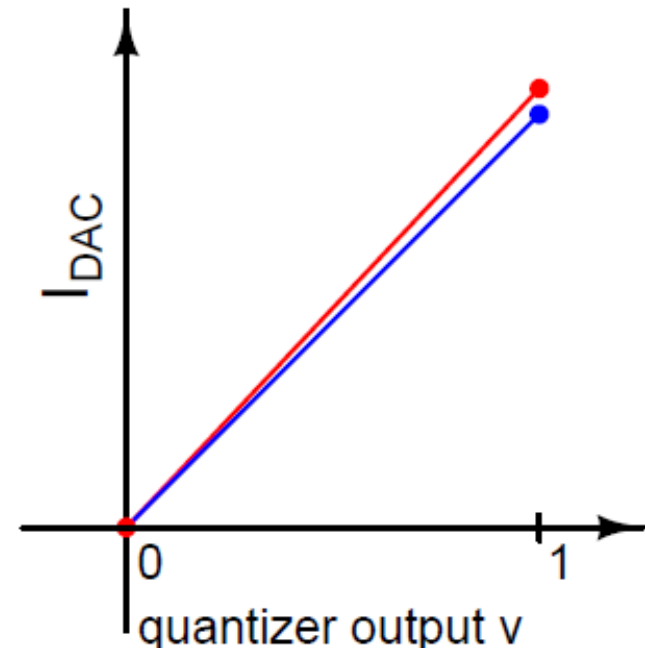
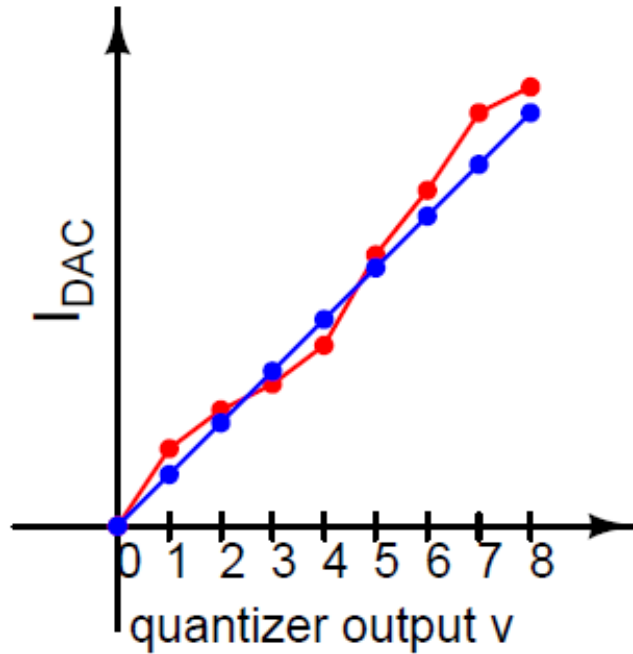


# Feedback DAC Architecture



- Array of unit-weighted current steering DACs to pull and push current from the opamp current summing node
- Use high crossover pre-drivers to reduce DAC glitching noise

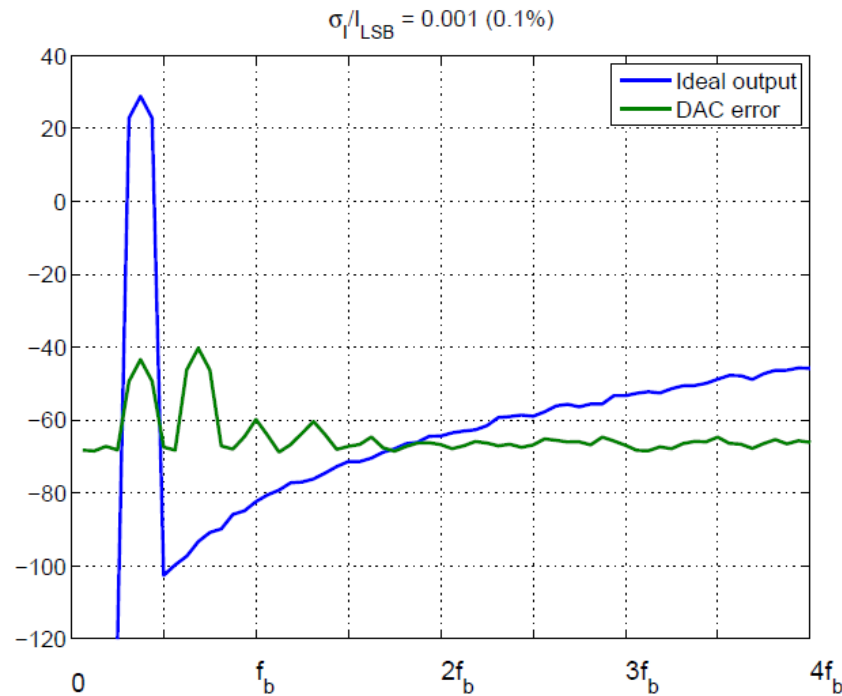
# Feedback DAC Nonlinearity



- In a multibit DAC, element mismatch leads to non-linearity
  - $v$  is related to the input ( $u$ ) by inverse non-linearity of the DAC
- A single-bit DAC is always linear

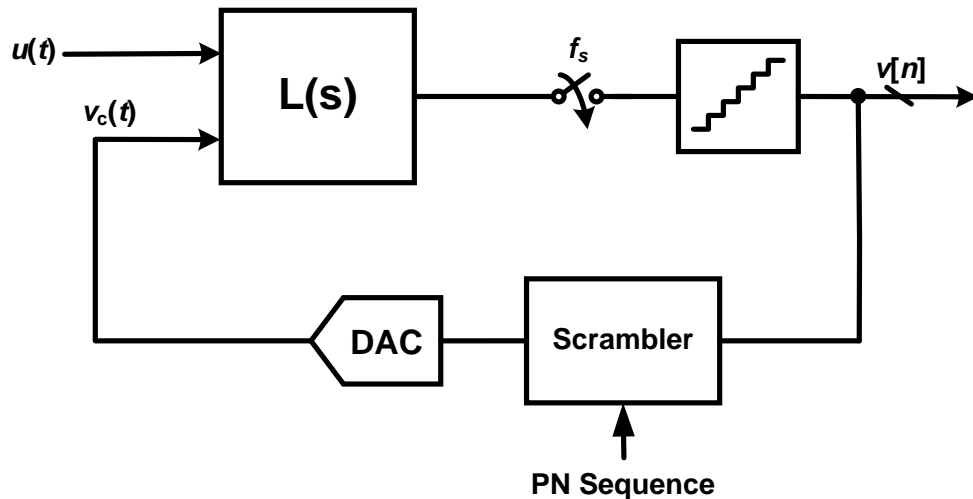


# Feedback DAC Nonlinearity

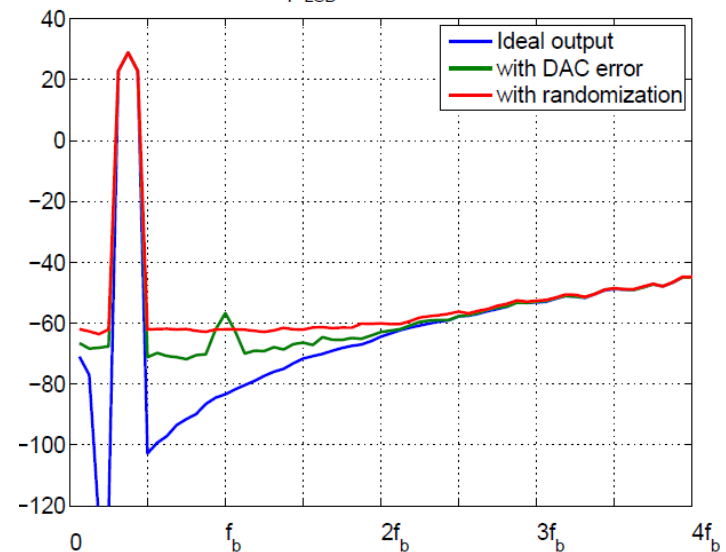
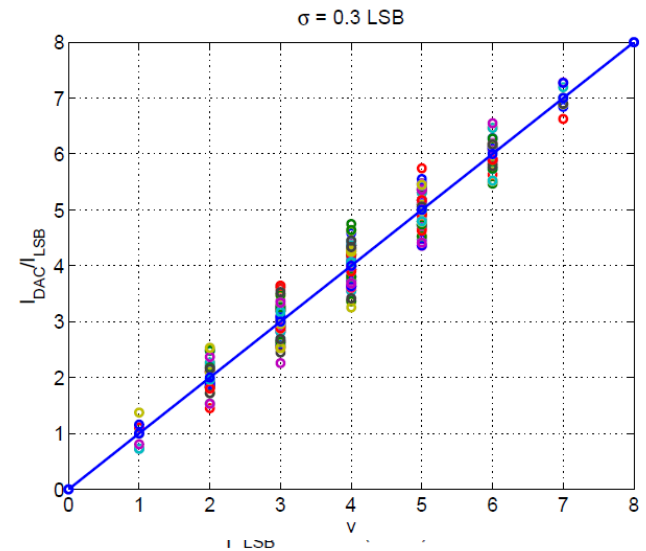


- Leads to distortion
- intermodulation of quantization noise into the signal band

# Dynamic Element Matching (DEM)

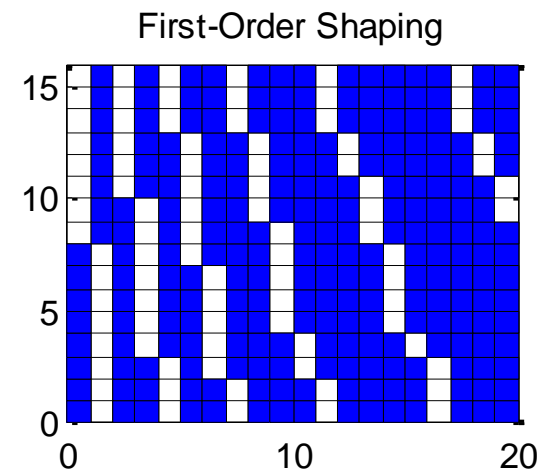
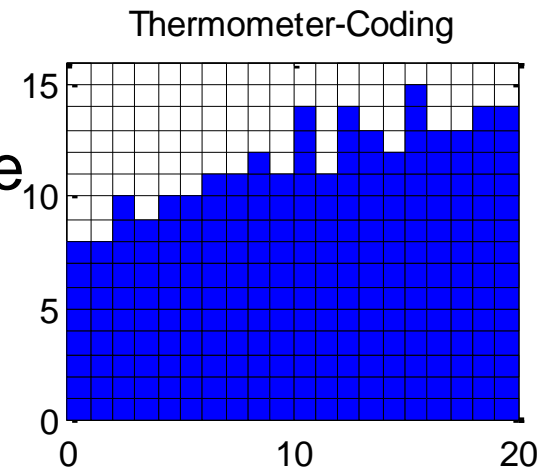
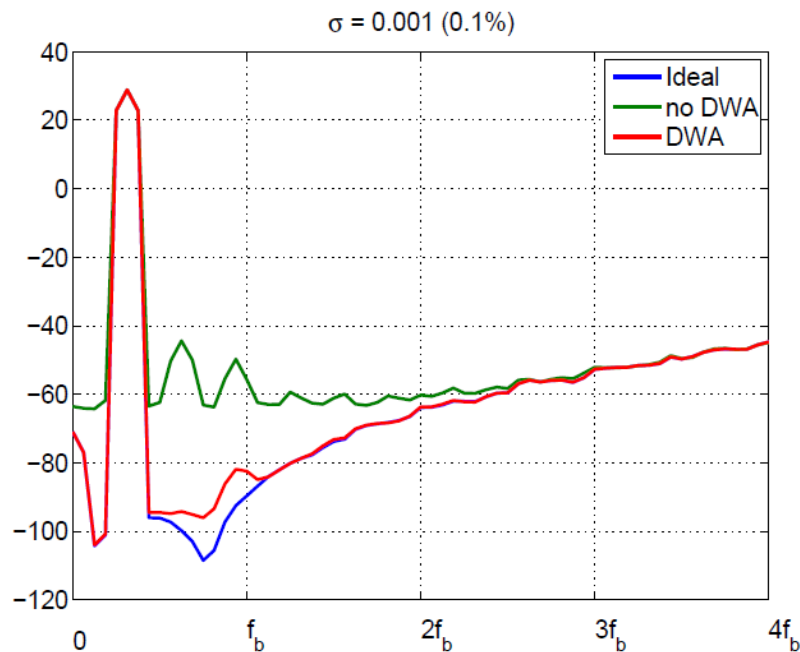


- Randomize the DAC elements
- Distortion components converted to noise
- Increased noise floor

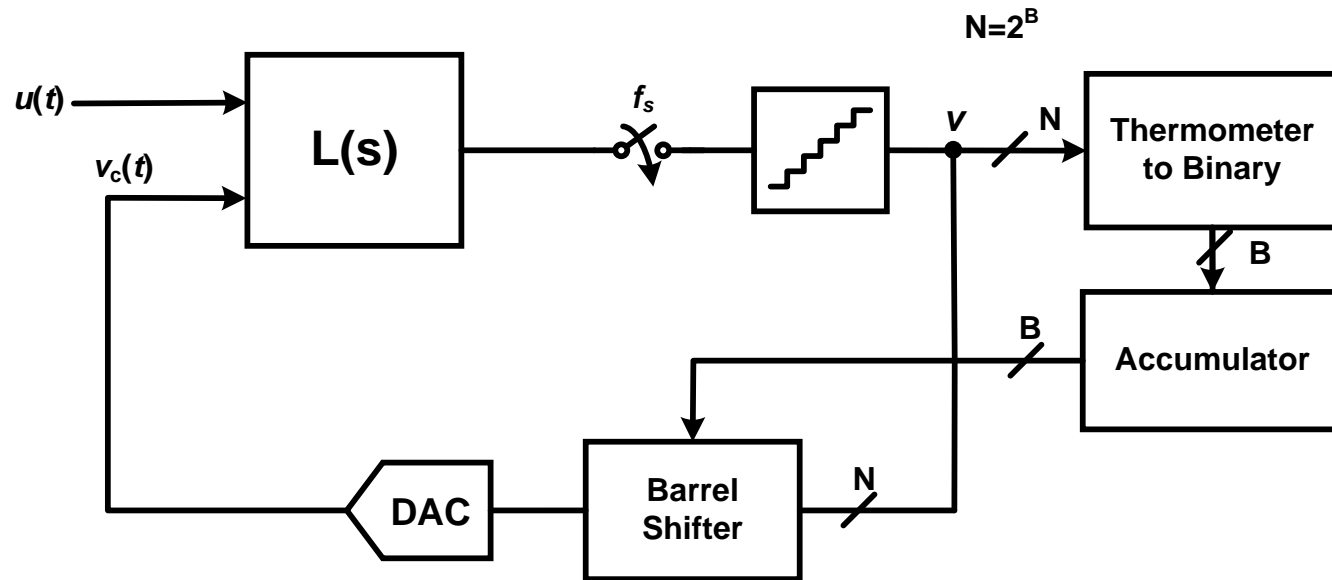


# Data Weighted Averaging (DWA)

- Cycle through all the current elements as fast as possible
- Accumulate the input code and move the pointer
- First-order mismatch noise shaping



# Data Weighted Averaging (DWA)

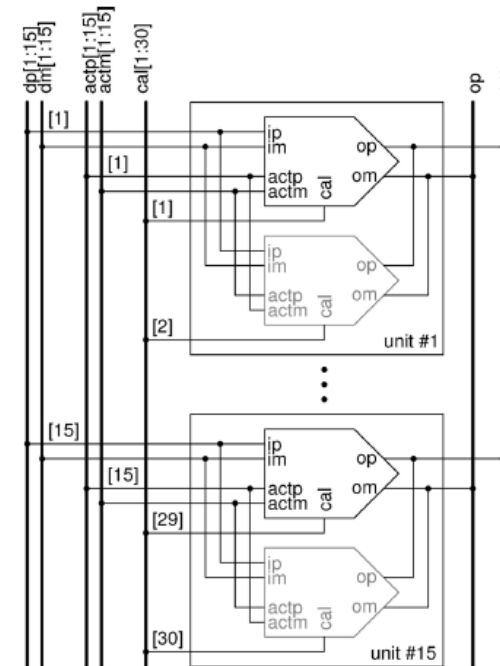
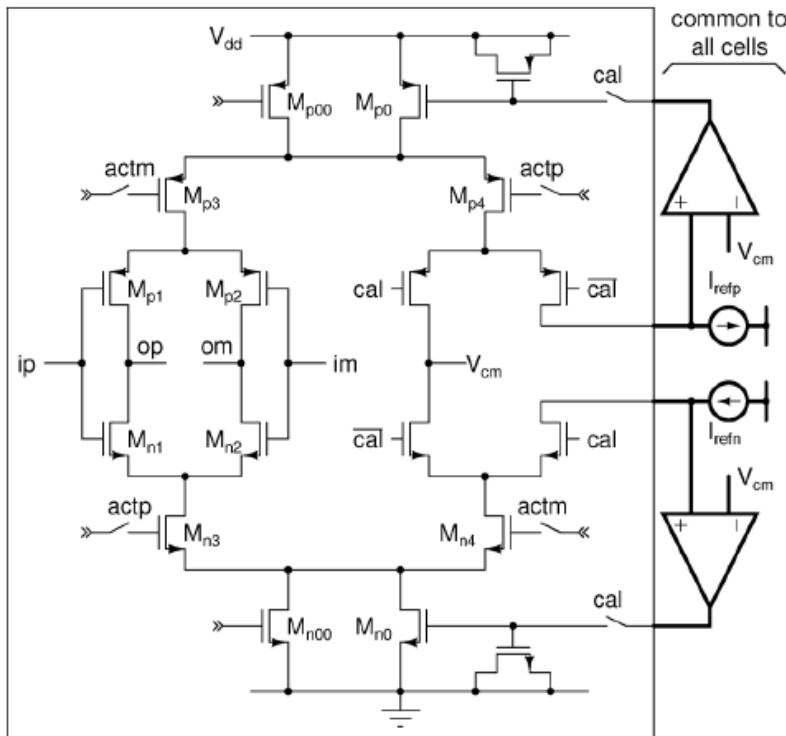


- Barrel shifter delay in the signal path increases loop delay
- Not viable at higher sampling rates (>400 MHz)

# DAC Calibration

## A 16 MHz BW 75 dB DR CT $\Delta\Sigma$ ADC Compensated for More Than One Cycle Excess Loop Delay

Vikas Singh, Nagendra Krishnapura, Shanthi Pavan, Baradwaj Vignanam, Debasish Behera, and Nimit Nigania



- Digital calibration – estimate element error and subtract from the output
  - Increased Decimation filter complexity
- Analog calibration – calibrate the elements with respect to a master
  - Need to calibrate at every cycle

# CT $\Delta\Sigma$ Advantages Summary

- ❑ Lower-power implementation
  - Relaxed bandwidth requirements for the integrators
- ❑ Inherent Anti-aliasing filtering (AAF)
  - Eliminates/relaxes input filtering
- ❑ Fixed resistive input impedance
- ❑ Higher sampling-rates extending to GHz-range
  - Suitable for RF integration
- ❑ Reduces supply and ground noise impact
- ❑ Less complicated clocking (compared to DT)

# CT $\Delta\Sigma$ Challenges Summary

- ❑ Complicated design due to hybrid CT-DT modeling
- ❑ Design doesn't scale with clock frequency
  - Loop-filter coefficient tuning for clock frequency migration
- ❑ High sensitivity to clock jitter (DAC reconstruction error)
- ❑ Excess loop-delay sensitivity
- ❑ Tuning required for RC time-constant variation
- ❑ Sensitive to DAC pulse shape, rise/fall time at high-speeds
- ❑ Comparator metastability at high speeds
- ❑ Higher-level simulation is challenging compared to DT
- ❑ Cascaded (MASH) designs are difficult due design complexity and mismatch in analog and digital transfer functions
  - Background calibration techniques

# Design Tools

- ❑ MATLAB and Simulink
  - Dr. Richard Schreier's  $\Delta\Sigma$  Toolbox
  - SIMSIDES Toolbox from Dr. Jose de la Rosa at University of Seville, Spain (Available under NDA)
- ❑ Verilog-A/AMS behavioral modeling in Cadence/Spectre
  - *Config view* in Virtuoso comes in handy for simulations
- ❑ Spectre Simulink Co-simulation Toolkit
- ❑ Berkeley Design Automation FastSpice for full-chip sims
- ❑ Synopsis tools for digital logic simulation and synthesis
- ❑ Mentor Graphics Calibre for DRC/LVS/Extraction

Use top-down design methodology with carefully thought mixed block-level simulations



# References

## Data Conversion Fundamentals

- A.1 M. Gustavsson, J. Wikner, N. Tan, *CMOS Data Converters for Communications*, Kluwer Academic Publishers, 2000.
- A.2 B. Razavi, *Principles of Data Conversion System Design*, Wiley-IEEE Press, 1994.
- A.3 ADC and DAC [Glossary](#) by Maxim.
- A.4 B. Murmann, "ADC Performance Survey 1997-2009," [[Online](#)].
- A.5 S. Pavan, N. Krishnapura, EE658: Data Conversion Circuits Course at IIT Madras [[Online](#)].
- A.6 The Fundamentals of FFT-Based Signal Analysis and Measurement, T.I. App Note [here](#).

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## Delta-Sigma Data Converters

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