

# Analog Integrated Systems Design

## Lecture 15 Oversampling Data Converters (2)

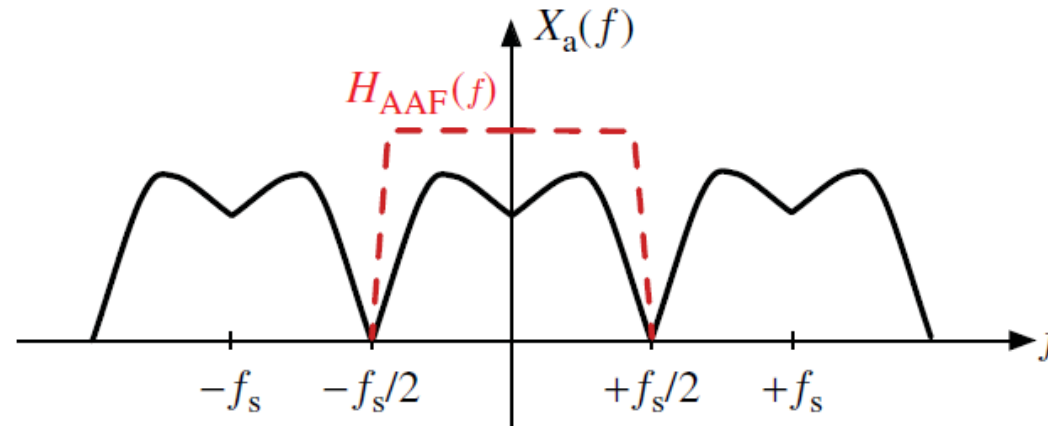
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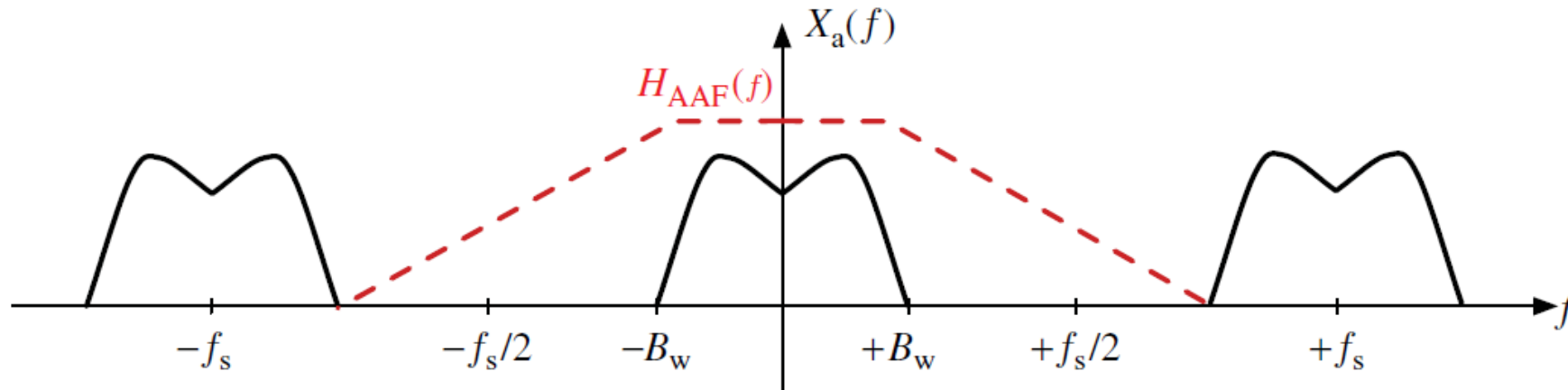
# Why Oversampling?

- ❑ Technology scaling enable very fast MOS transistors
  - GHz sampling and processing is possible
  - We can build faster ADCs for broadband signals
- ❑ But signals in many applications have limited bandwidth
  - Ex: sensors (baseband) and communication systems (passband)
- ❑ Oversampling:  $f_s \gg f_N = 2BW$ 
  - Make use of the high sample rate to improve the resolution
    - Oversampling Ratio (OSR)
$$OSR = \frac{f_s}{f_N} = \frac{f_s}{2BW}$$
  - Also simpler antialiasing filter
  - But higher digital power consumption

# Nyquist vs Oversampling ADC



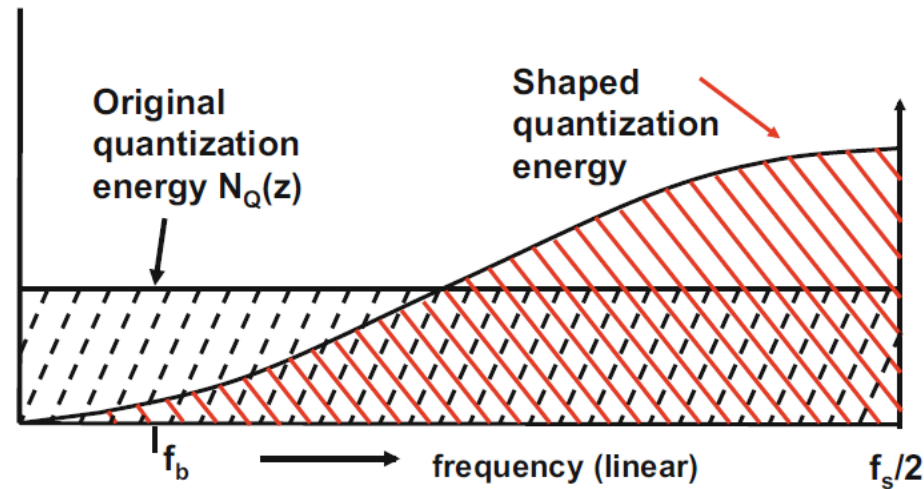
(a)



(b)

# Noise Shaping

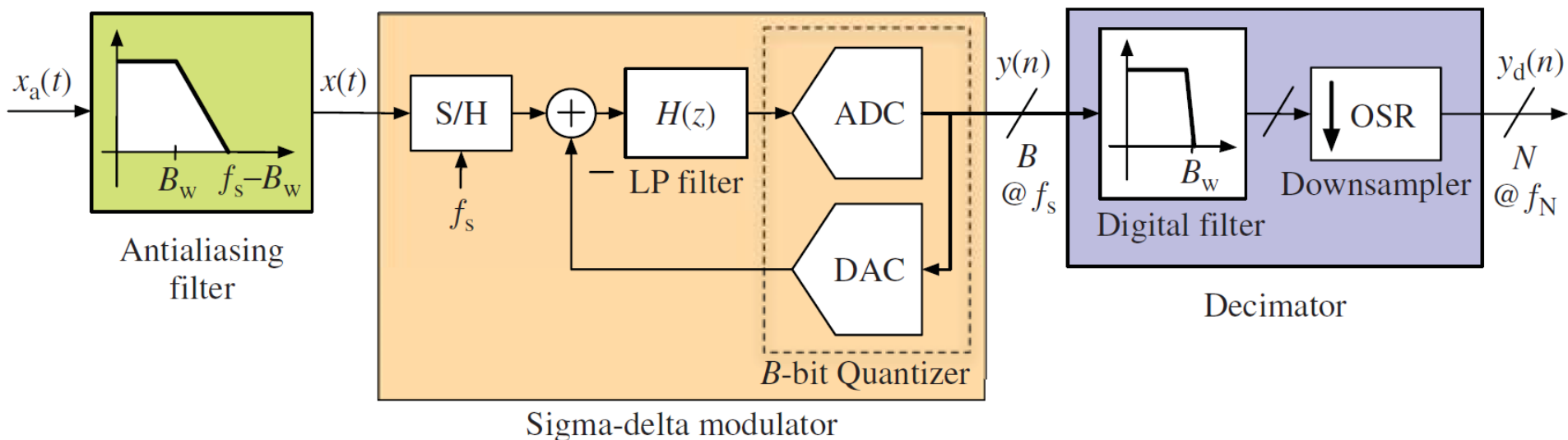
- ❑ Noise transfer function (NTF) is a HPF (differentiator)



- ❑ For 1<sup>st</sup> order NTF: The shaped noise has twice the noise power
  - But IBN is significantly reduced

# Sigma-Delta ( $\Sigma\Delta$ ) ADC

- ❑ Closed loop negative feedback system
- ❑  $H(z)$  is the loop filter
- ❑ The B-bit quantizer is typically 1 – 5 bit
  - Single bit: One bit DAC is inherently linear
    - We care more about DAC linearity (we will know why later)
  - Multibit: Each bit in the ADC/DAC adds 6dB to the SNR



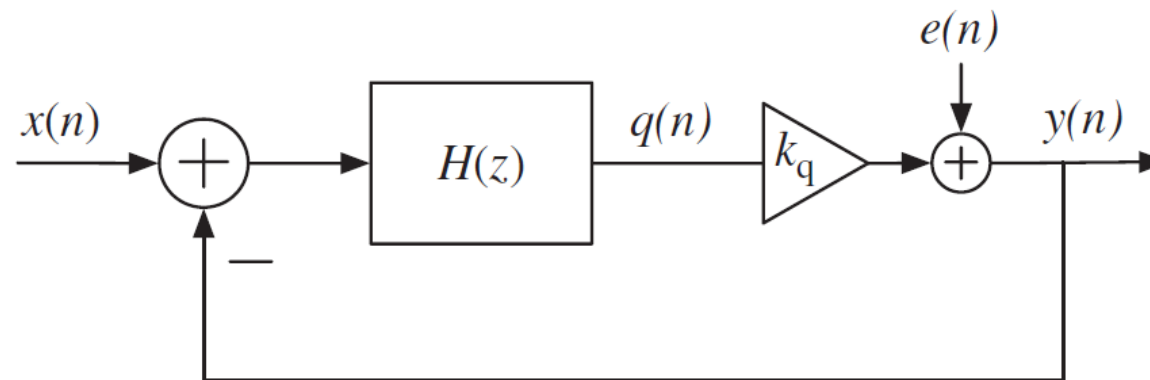
# First-Order $\Sigma\Delta\text{M}$

$$\text{Let } H(z) = \frac{z^{-1}}{1-z^{-1}} \text{ and } k_q = 1$$

$$STF(z) = \frac{H(z)}{1+H(z)} = z^{-1} \quad \rightarrow \quad \text{Delay}$$

$$NTF(z) = \frac{1}{1+H(z)} = 1 - z^{-1} \quad \rightarrow \quad \text{Noise shaping}$$

$$Y = STF \cdot X + NTF \cdot E = z^{-1} \cdot X + (1 - z^{-1}) \cdot E$$



# Higher-Order $\Sigma\Delta\mathbf{M}$

$$\text{Let } H(z) = \left(\frac{z^{-1}}{1-z^{-1}}\right)^L \text{ and } k_q = 1$$

$$STF(z) = \frac{H(z)}{1+H(z)} = z^{-L} \quad \rightarrow \quad \text{Delay}$$

$$NTF(z) = \frac{1}{1+H(z)} = (1 - z^{-1})^L \quad \rightarrow \quad \text{Noise shaping}$$

$$Y = STF \cdot X + NTF \cdot E = z^{-L} \cdot X + (1 - z^{-1})^L \cdot E$$

$$SQNR = 10 \log \left( \frac{P_{sig}}{IBN} \right)$$

$$\approx 1.76 + 6.02N + \mathbf{10 \log \left( \frac{2L + 1}{\pi^{2L}} \right)} + \mathbf{(2L + 1)10 \log(OSR)}$$

# Noise Shaping Gain

$$SQNR \approx 1.76 + 6.02N + 10 \log \left( \frac{2L+1}{\pi^{2L}} \right) + (2L+1)10 \log(OSR)$$

$$ENOB \text{ Gain} = \frac{(2L+1)10 \log(OSR)}{6} \approx (2L+1) \times 0.5 \log_2(OSR)$$

- ❑ SNQR increases with OSR by  $3(2L+1)$  dB/octave
- ❑ ENOB increases with OSR by  $(L+0.5)$  bit/octave
- ❑ Need  $OSR > 4$  (more than two octaves) to reap  $\Sigma\Delta M$  benefits

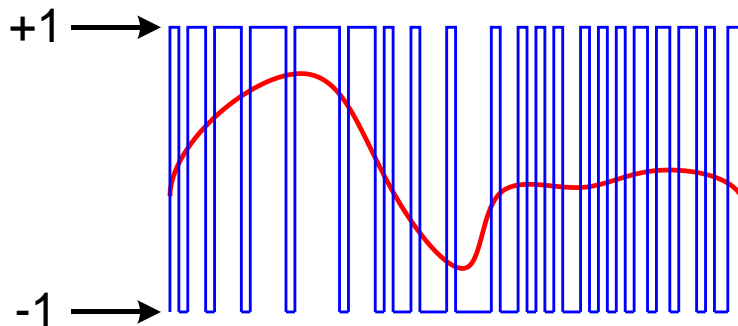
Order (L)	Static SNR loss	SNR gain	Static ENOB loss	ENOB gain
0	0	3 dB/octave	0	0.5 bit/octave
1	-5.2 dB	9 dB/octave	-0.86 bit	1.5 bit/octave
2	-12.9 dB	15 dB/octave	-2.14 bit	2.5 bit/octave
3	-21.4 dB	21 dB/octave	-3.55 bit	3.5 bit/octave
4	-30.2 dB	27 dB/octave	-5.02 bit	4.5 bit/octave



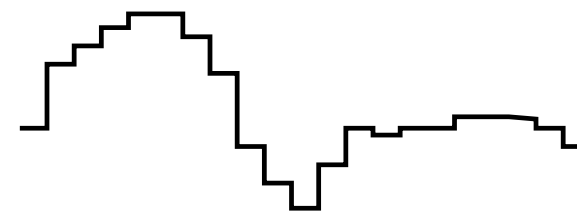
# $\Sigma\Delta$ vs Nyquist ADC

- ❑  $\Sigma\Delta$  ADC behaves quite differently from Nyquist converters
- ❑ Digital codes only display an “average” impression of the input
- ❑ INL, DNL, monotonicity, missing code, etc. do not directly apply in  $\Sigma\Delta$  converters
- ❑ Usually only dynamic specs are important (SNR, SNDR, SFDR, etc.)

$\Sigma\Delta$  ADC output (1-bit)



Nyquist ADC output



# $\Sigma\Delta$ Ms Classification

- ❑ Single-Bit vs Multibit  $\Sigma\Delta$ Ms
- ❑ First-order vs Higher-order  $\Sigma\Delta$ Ms
  - Order of the loop filter
- ❑ Single-Loop vs Cascade or MASH  $\Sigma\Delta$ Ms
  - Single-loop: uses only one quantizer
  - Cascade or MASH: uses several quantizers

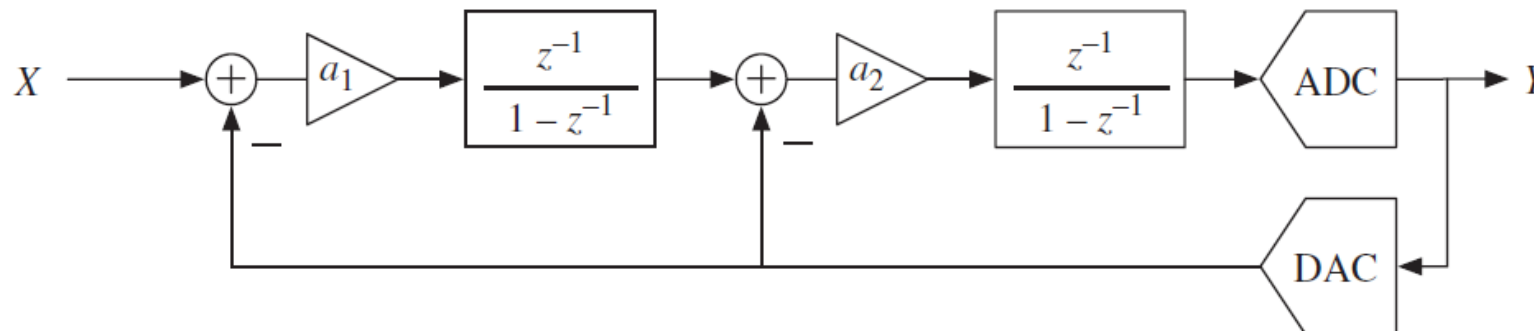
# Second-order $\Sigma\Delta\text{M}$

- Two DT integrators are cascaded
  - Each integrator receives a weighted feedback path

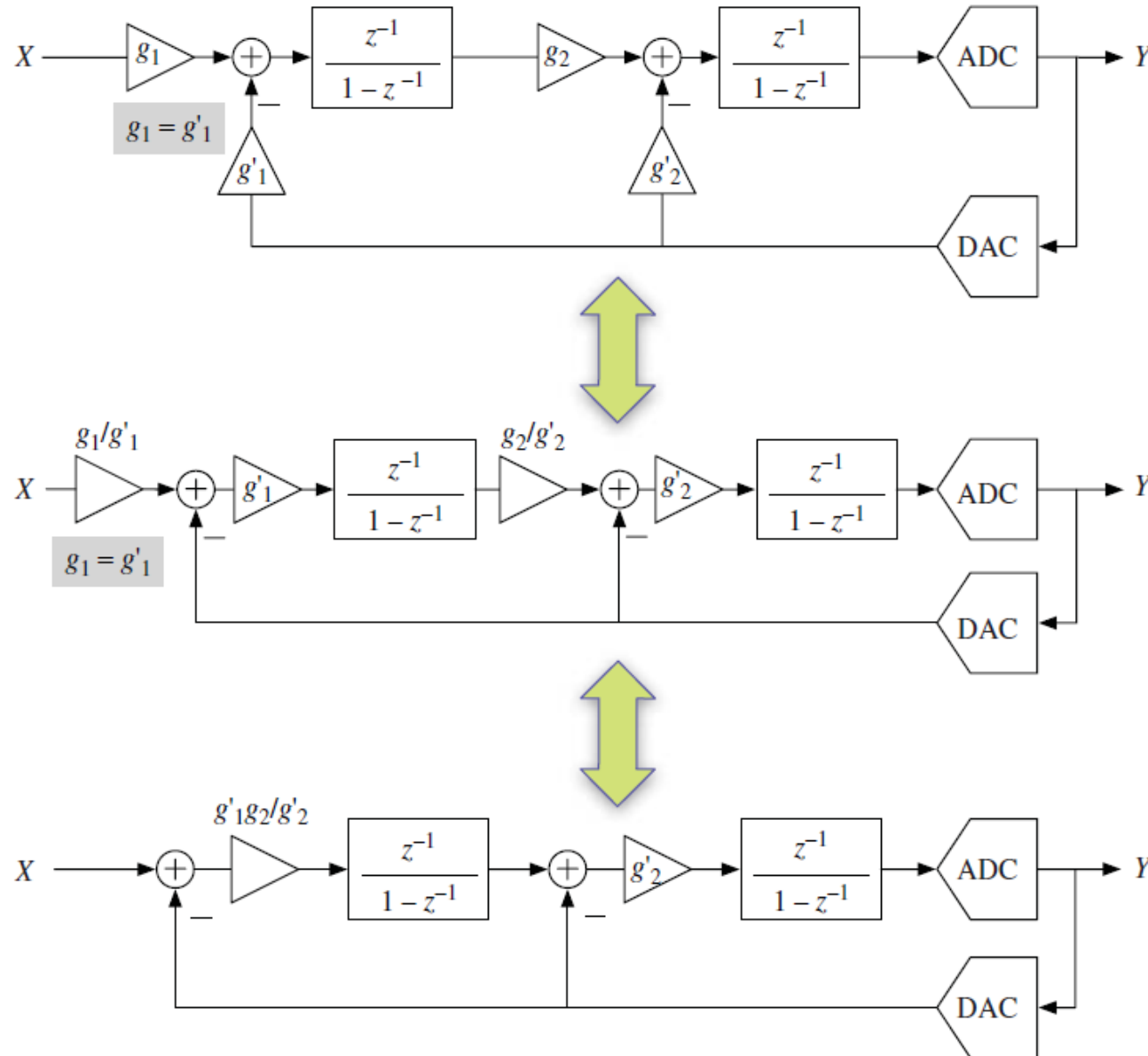
$$Y(z) = \frac{k_q a_1 a_2 \frac{z^{-2}}{(1-z^{-1})^2} X(z) + E(z)}{1 + k_q a_1 a_2 \frac{z^{-2}}{(1-z^{-1})^2} + k_q a_2 \frac{z^{-1}}{(1-z^{-1})}}$$

$$Y(z) = z^{-2} X(z) + (1 - z^{-1})^2 E(z)$$

$$k_q a_1 a_2 = 1$$
$$k_q a_2 = 2$$



# Second-order $\Sigma\Delta\text{M}$



$$a_1 = \frac{g'_1 g_2}{g'_2}$$

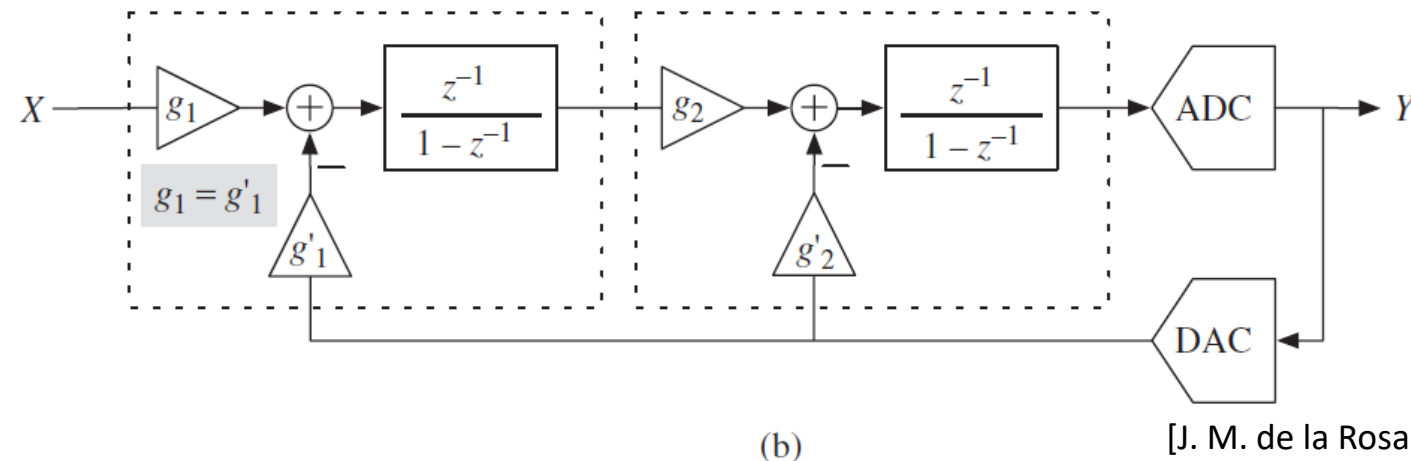
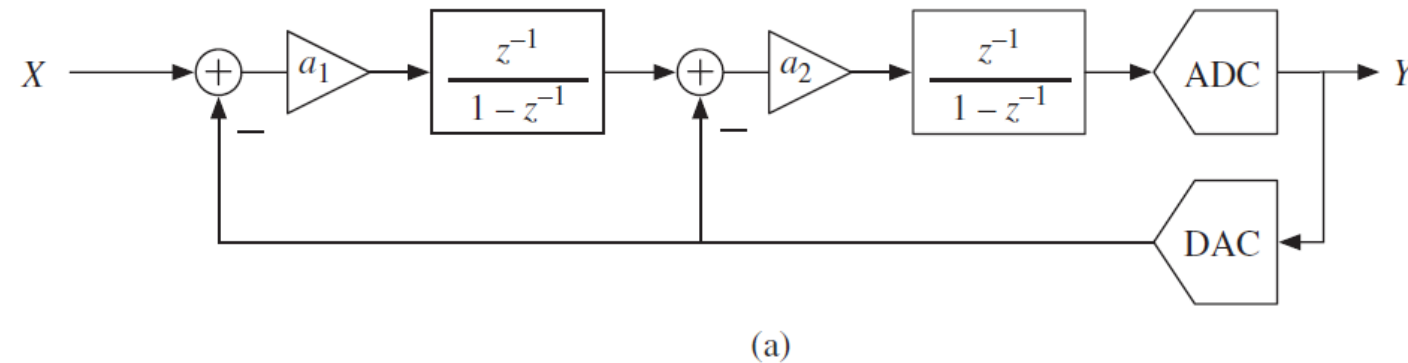
$$a_2 = g'_2$$

# Second-order $\Sigma\Delta\text{M}$

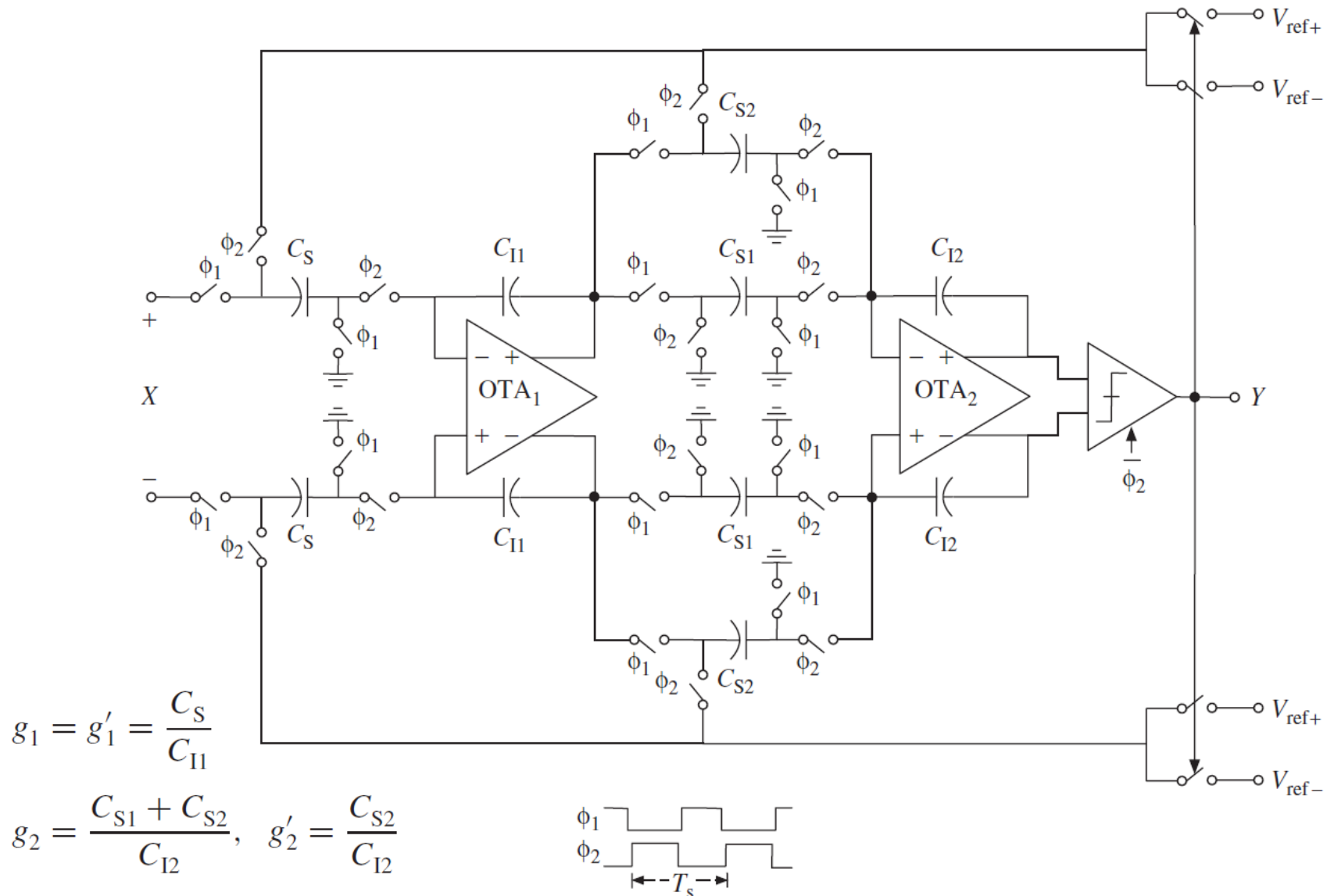
□ Two alternative representations are possible

- (a) suits system level
- (b) suits circuit level

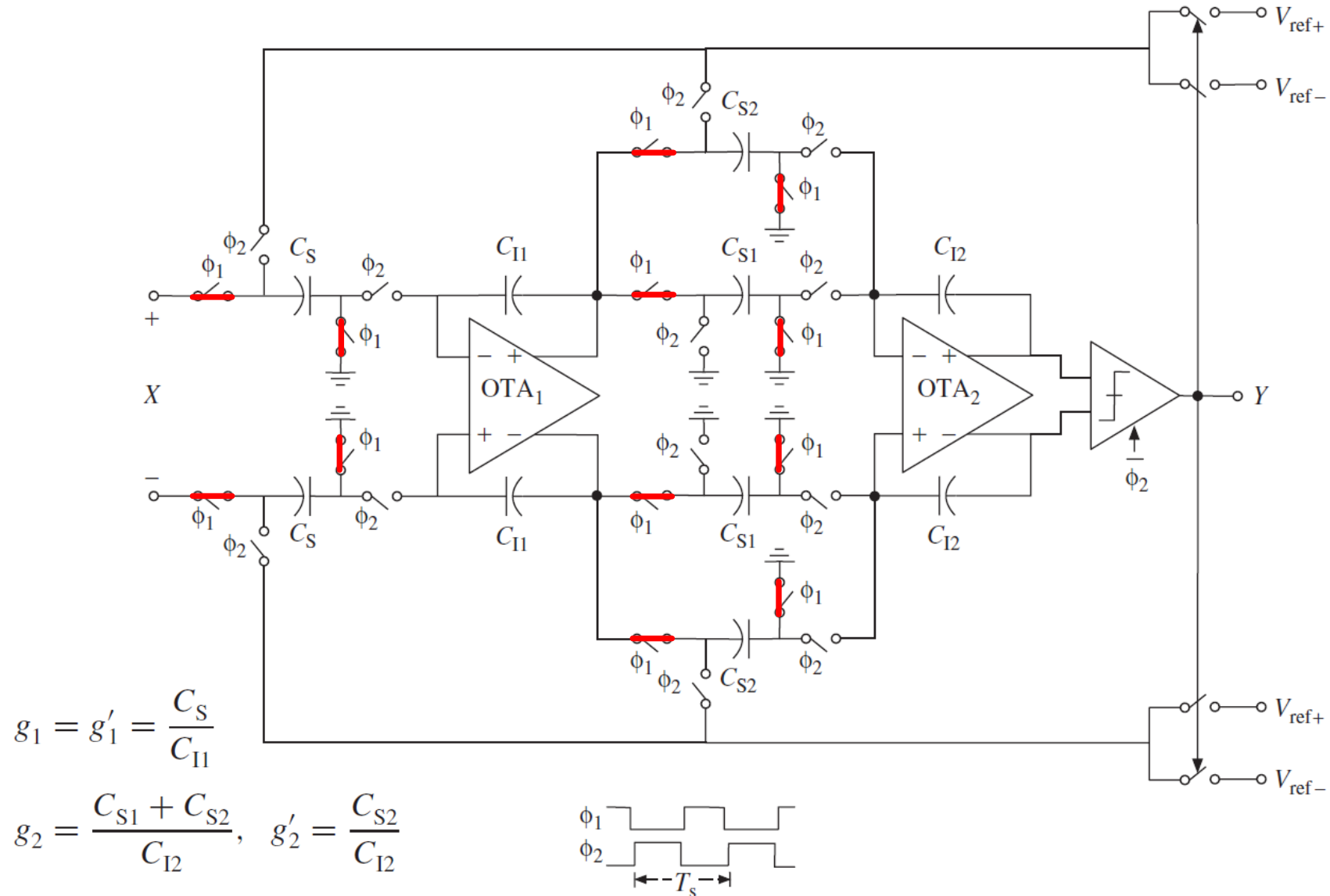
$$a_1 = \frac{g'_1 g_2}{g'_2}$$
$$a_2 = g'_2$$



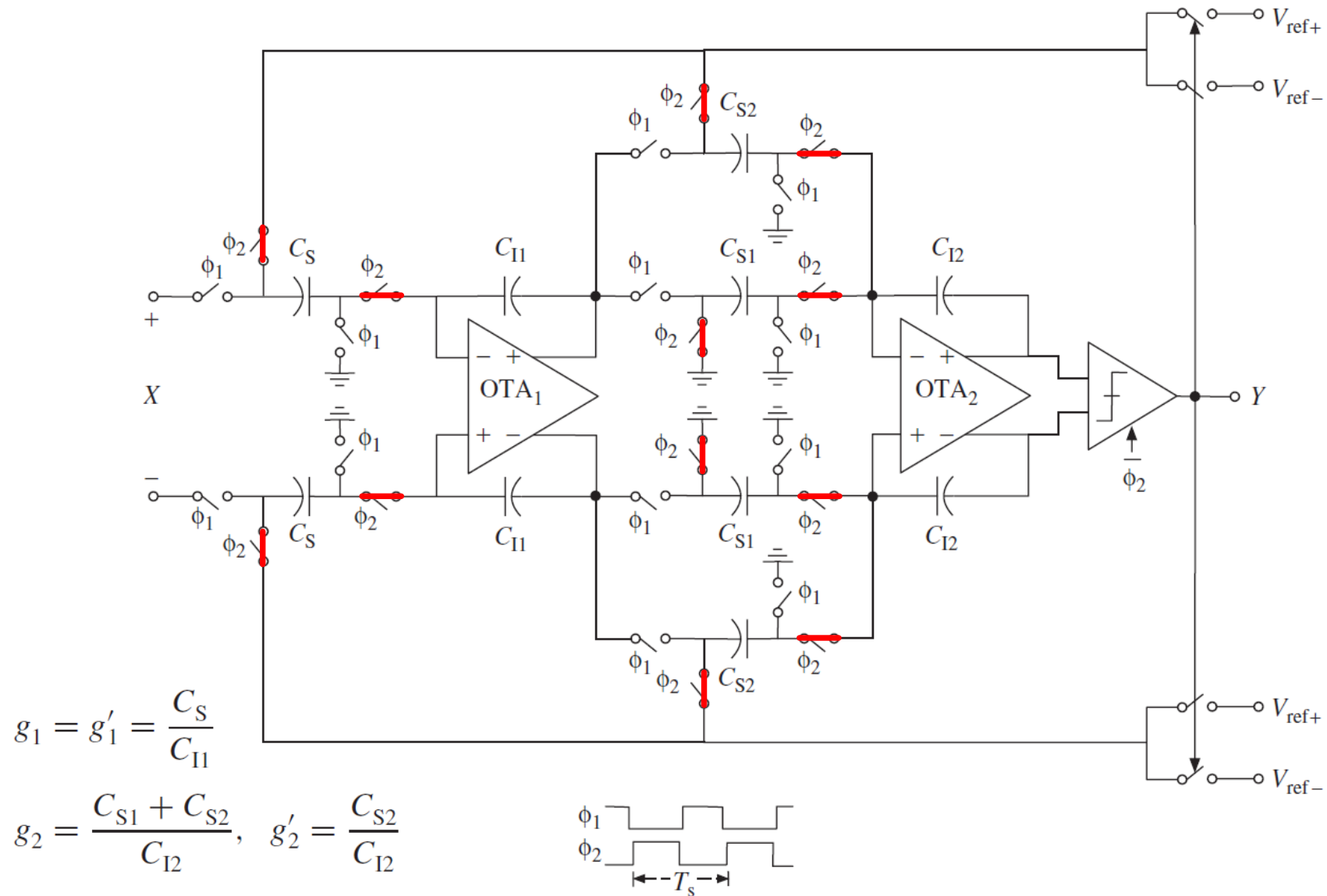
# Second-order $\Sigma\Delta\mathbf{M}$ : SC Implementation



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# Second-order $\Sigma\Delta\mathbf{M}$ : SC Implementation





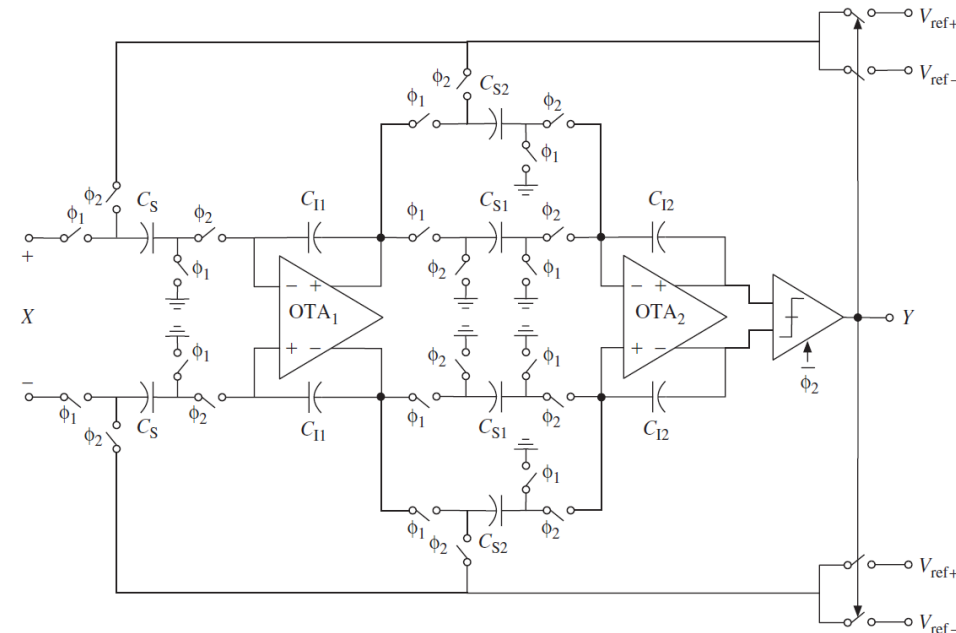
# Second-order $\Sigma\Delta\text{M}$ : Design Examples

$$k_q a_1 a_2 = 1$$

$$k_q a_2 = 2$$

$$g_1 = g'_1 = \frac{C_S}{C_{I1}}$$

$$g_2 = \frac{C_{S1} + C_{S2}}{C_{I2}}, \quad g'_2 = \frac{C_{S2}}{C_{I2}}$$



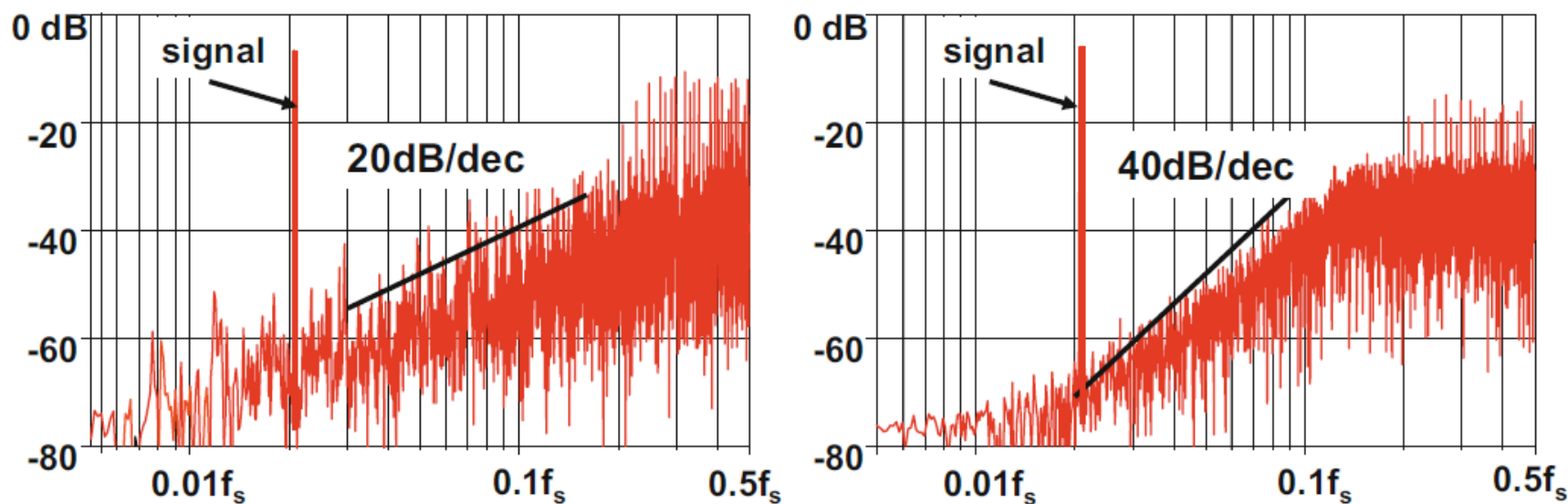
$$a_1 = \frac{g'_1 g_2}{g'_2}$$

$$a_2 = g'_2$$

$g_1, g'_1$	1/2, 1/2	1/4, 1/4	1/2, 1/2	1/3, 1/3
$g_2, g'_2$	1/2, 1/2	1/2, 1/4	1, 1/2	3/5, 2/5
$a_1, a_2$	0.5, 0.5	0.5, 0.25	0.5, 0.5	0.5, 0.4
Overload level	-4 dBFS	-4 dBFS	-4 dBFS	-4 dBFS
Integrator output swing	$\pm 1.5 V_{\text{ref}}$	$\pm 0.75 V_{\text{ref}}$	$\pm 1.25 V_{\text{ref}}$	$\pm 1.0 V_{\text{ref}}$
Unit capacitors (2 $\times$ in fully diff)	6(= 3 + 3)	11(= 5 + 6)	9(= 5 + 4)	12(= 4 + 8)

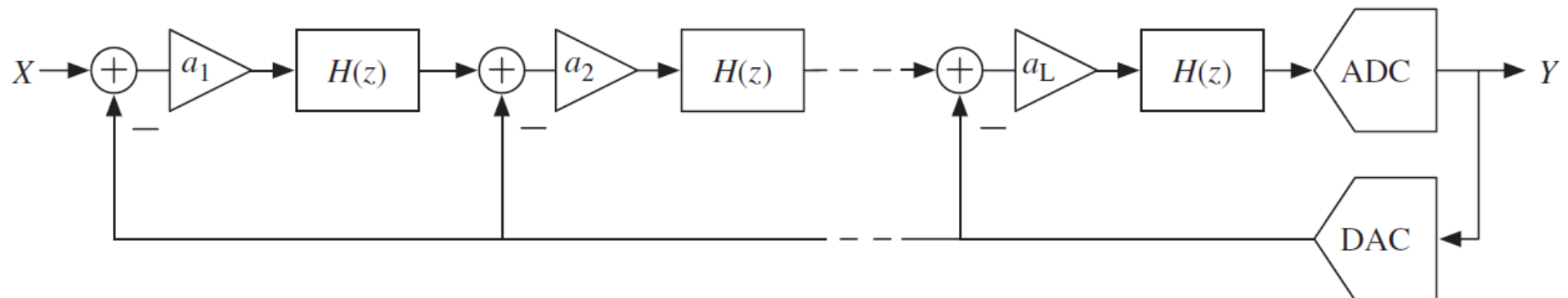
# First- vs Second-order SDM

- ❑ The dual integration of the signal and the two feedback paths from the quantizer create a much more complex pattern
  - Less correlated products → Less idle tones
- ❑ Higher-order converters scramble the pattern even more due to the extra integration stages in the filters
  - Third and fourth order show hardly any idle tones



# Higher-order $\Sigma\Delta\mathbf{M}$ with Distributed FB

- Simply include  $L$  integrators before the quantizer
- Derive a set of relations between the integrator scaling coefficients to fulfill pure differentiator noise shaping
$$Y = z^{-L} \cdot X + (1 - z^{-1})^L \cdot E$$
- But pure-differentiator NTFs are prone to instability if  $L > 2$ 
  - Instability appears at the modulator output as a large-amplitude and low-frequency oscillation
    - Long bitstreams of alternating +1s and -1s

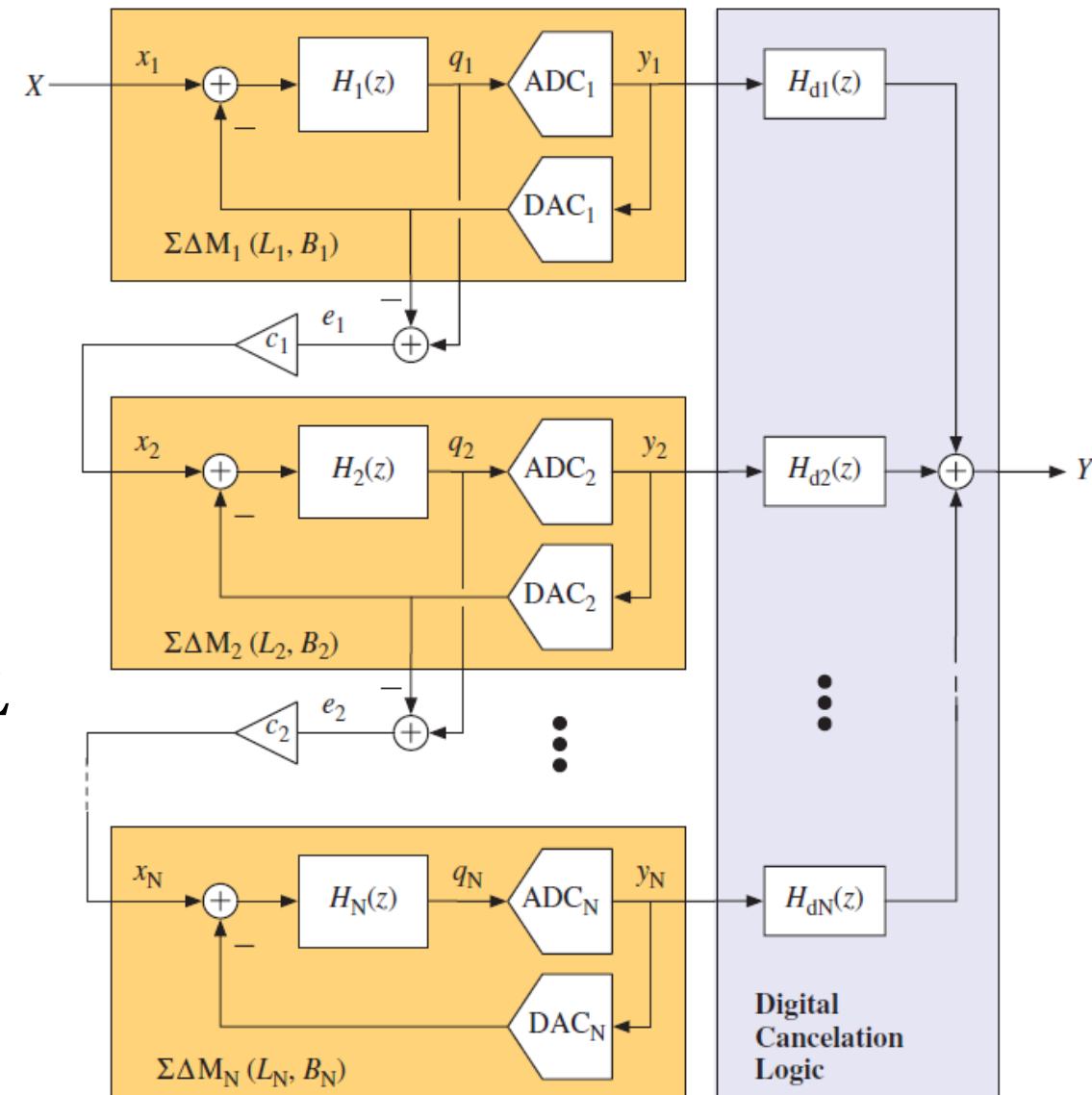


# $\Sigma\Delta$ Ms Classification

- ❑ Single-Bit vs Multibit  $\Sigma\Delta$ Ms
- ❑ First-order vs Higher-order  $\Sigma\Delta$ Ms
  - Order of the loop filter
- ❑ Single-Loop vs Cascade or MASH  $\Sigma\Delta$ Ms
  - Single-loop: uses only one quantizer
  - Cascade or MASH: uses several quantizers

# Cascade $\Sigma\Delta$ Ms

- ❑ A.k.a. multiloop  $\Sigma\Delta$ M or **m**ultistage noise **s**haping (**MASH**)  $\Sigma\Delta$ M
- ❑ An alternative approach to obtain a high-order noise shaping while avoiding instabilities
- ❑  $L = \sum L_i$
- ❑ Unconditionally stable if  $L_i \leq 2$
- ❑ No inter-stage feedback
- ❑ Digital cancelation logic (DCL) combines outputs such that only  $e_3$  appears at output (shaped by  $L = \sum L_i$ )
- ❑  $L$  limited by circuit non-idealities (noise leakage)
  - Practically  $e_1$  (shaped by  $L_1$ ) and  $e_2$  (shaped by  $L_1 + L_2$ ) will leak to output



# MASH $\Sigma\Delta$ M Topologies

□ The first stage is usually 2<sup>nd</sup> order SDM

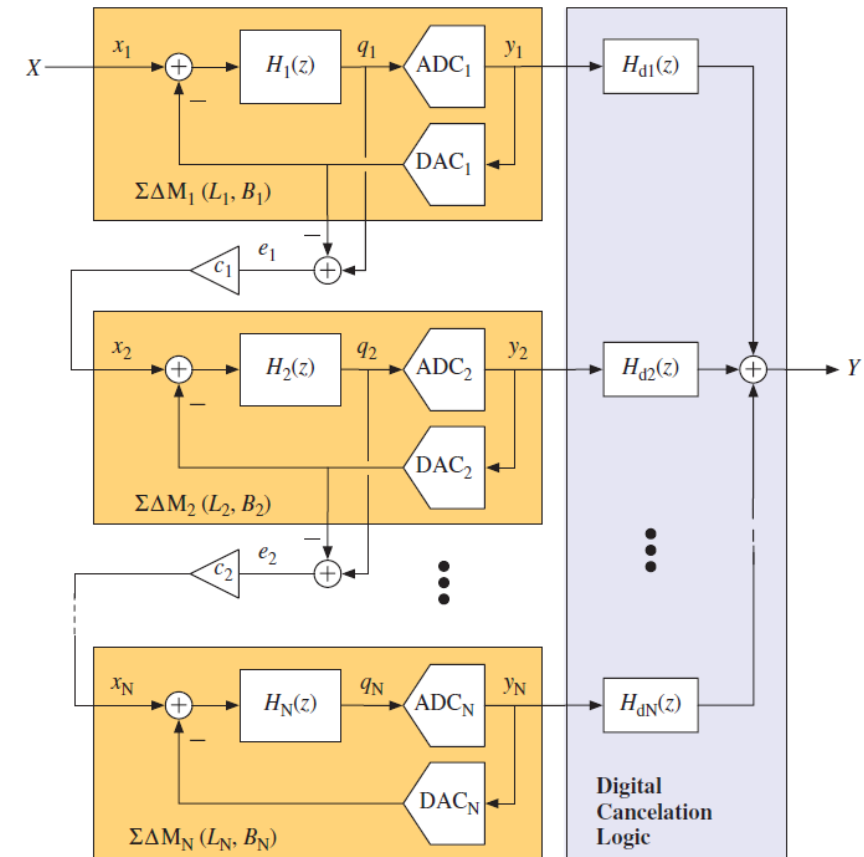
- Reduce noise leakage
- Avoid idle tones

□ Example MASH topologies

- 2-1                    → 3<sup>rd</sup> order
- 2-2                    → 4<sup>th</sup> order
- 2-1-1                → ...
- 2-2-1
- 2-1-1-1
- 2-2-2
- etc.

# Two-Stage Cascade $\Sigma\Delta$ Ms Example

- ❑  $X_1(z) = X(z)$
- ❑  $X_2(z) = -c_1 E_1(z)$
- ❑ The digital transfer function should track the analog transfer function
  - Imperfect tracking means  $E_1$  will leak to output (not completely canceled)



# Two-Stage Cascade $\Sigma\Delta$ Ms Example

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- ❑  $X_2(z) = -c_1 E_1(z)$
- ❑ The digital transfer function should track the analog transfer function
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$$Y_1(z) = \text{STF}_1(z)X_1(z) + \text{NTF}_1(z)E_1(z)$$

$$Y_2(z) = \text{STF}_2(z)X_2(z) + \text{NTF}_2(z)E_2(z)$$

$$Y(z) = H_{d1}(z)Y_1(z) + H_{d2}(z)Y_2(z)$$

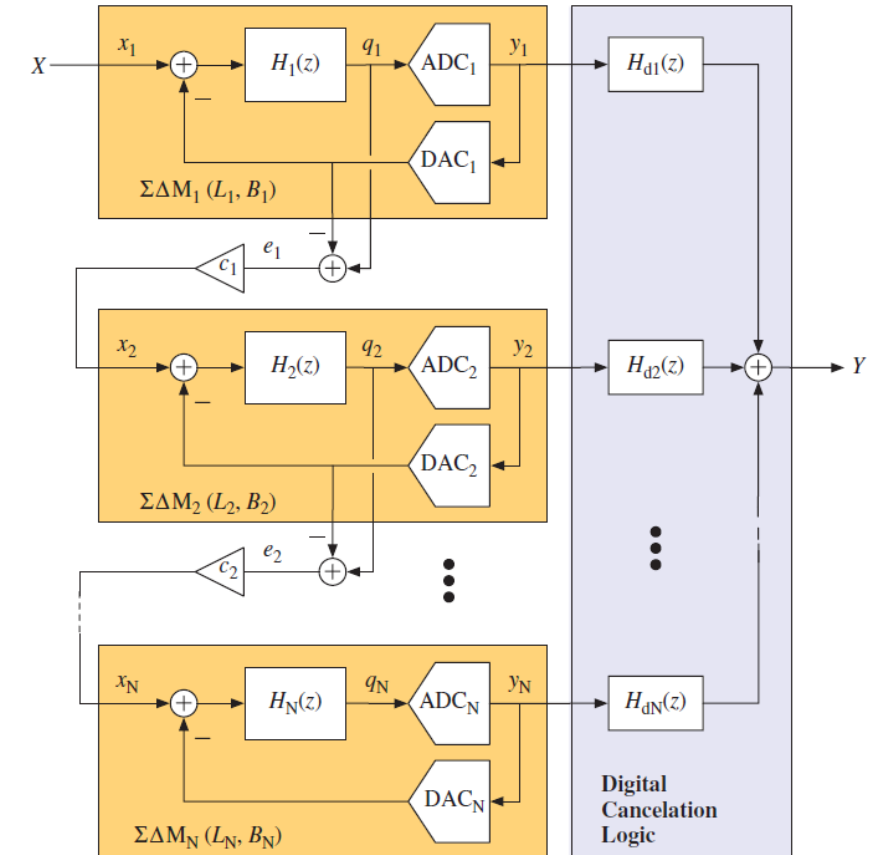
$$= \text{STF}_{\text{casc}}(z)X(z) + \text{NTF}_{1,\text{casc}}(z)E_1(z) + \text{NTF}_{2,\text{casc}}(z)E_2(z)$$

$$\text{STF}_{\text{casc}}(z) = H_{d1}(z)\text{STF}_1(z)$$

$$\text{NTF}_{1,\text{casc}}(z) = H_{d1}(z)\text{NTF}_1(z) - c_1 H_{d2}(z)\text{STF}_2(z)$$

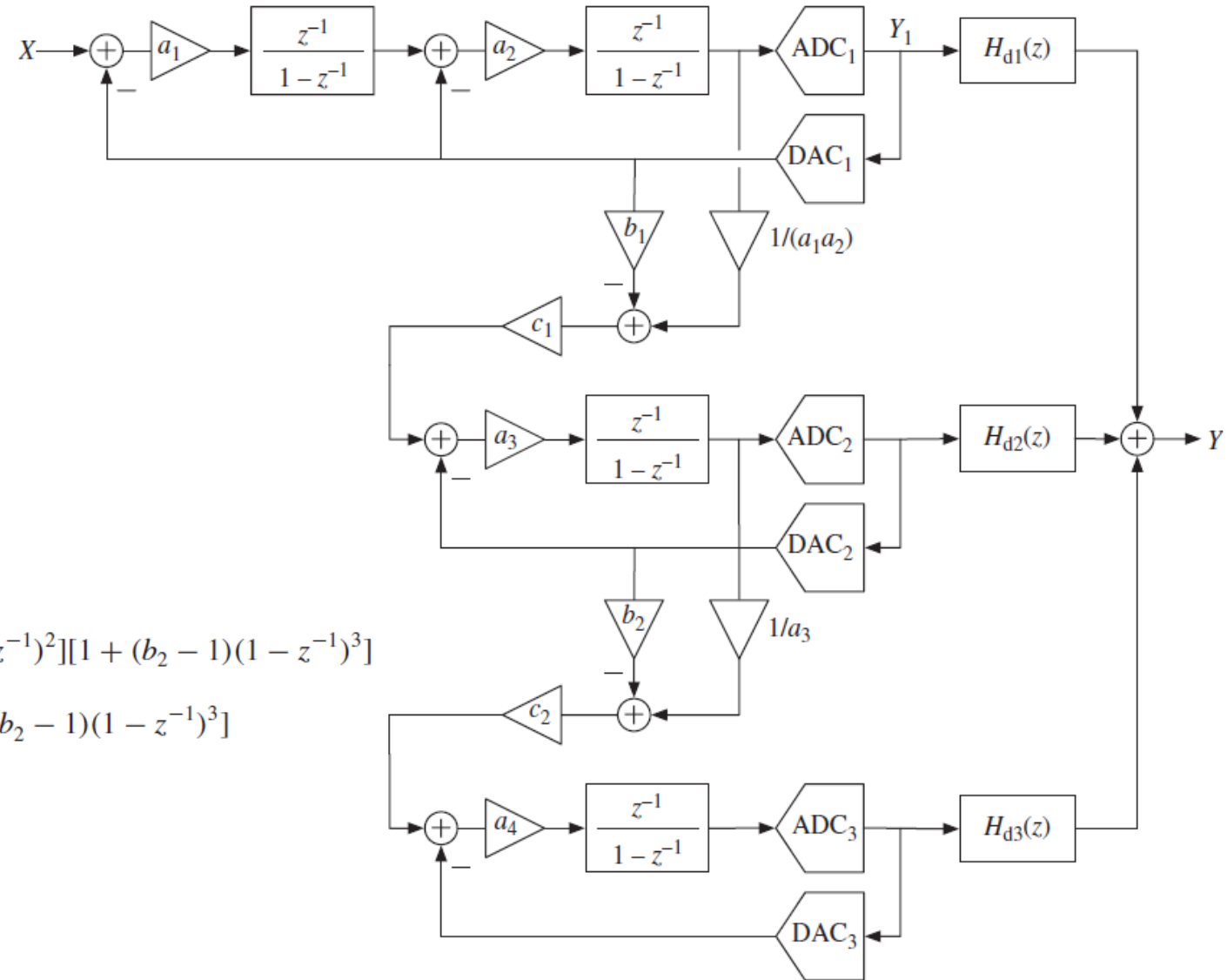
$$\text{NTF}_{2,\text{casc}}(z) = H_{d2}(z)\text{NTF}_2(z)$$

$$\left. \begin{aligned} H_{d1}(z) &= \text{STF}_2(z) \\ H_{d2}(z) &= \frac{1}{c_1} \text{NTF}_1(z) \end{aligned} \right\} \Rightarrow \left\{ \begin{aligned} \text{STF}_{\text{casc}}(z) &= \text{STF}_1(z)\text{STF}_2(z) \\ \text{NTF}_{1,\text{casc}}(z) &= 0 \\ \text{NTF}_{2,\text{casc}}(z) &= \frac{1}{c_1} \text{NTF}_1(z)\text{NTF}_2(z) \end{aligned} \right.$$





# Example: MASH 2-1-1 SDM



$$k_{q1}a_1a_2 = 1, \quad k_{q1}a_2 = 2$$

$$k_{q2}a_3 = 1$$

$$k_{q3}a_4 = 1$$

$$H_{d1}(z) = z^{-2}[1 + (b_1 - 1)(1 - z^{-1})^2][1 + (b_2 - 1)(1 - z^{-1})^3]$$

$$H_{d2}(z) = \frac{1}{c_1} z^{-1} (1 - z^{-1})^2 [1 + (b_2 - 1)(1 - z^{-1})^3]$$

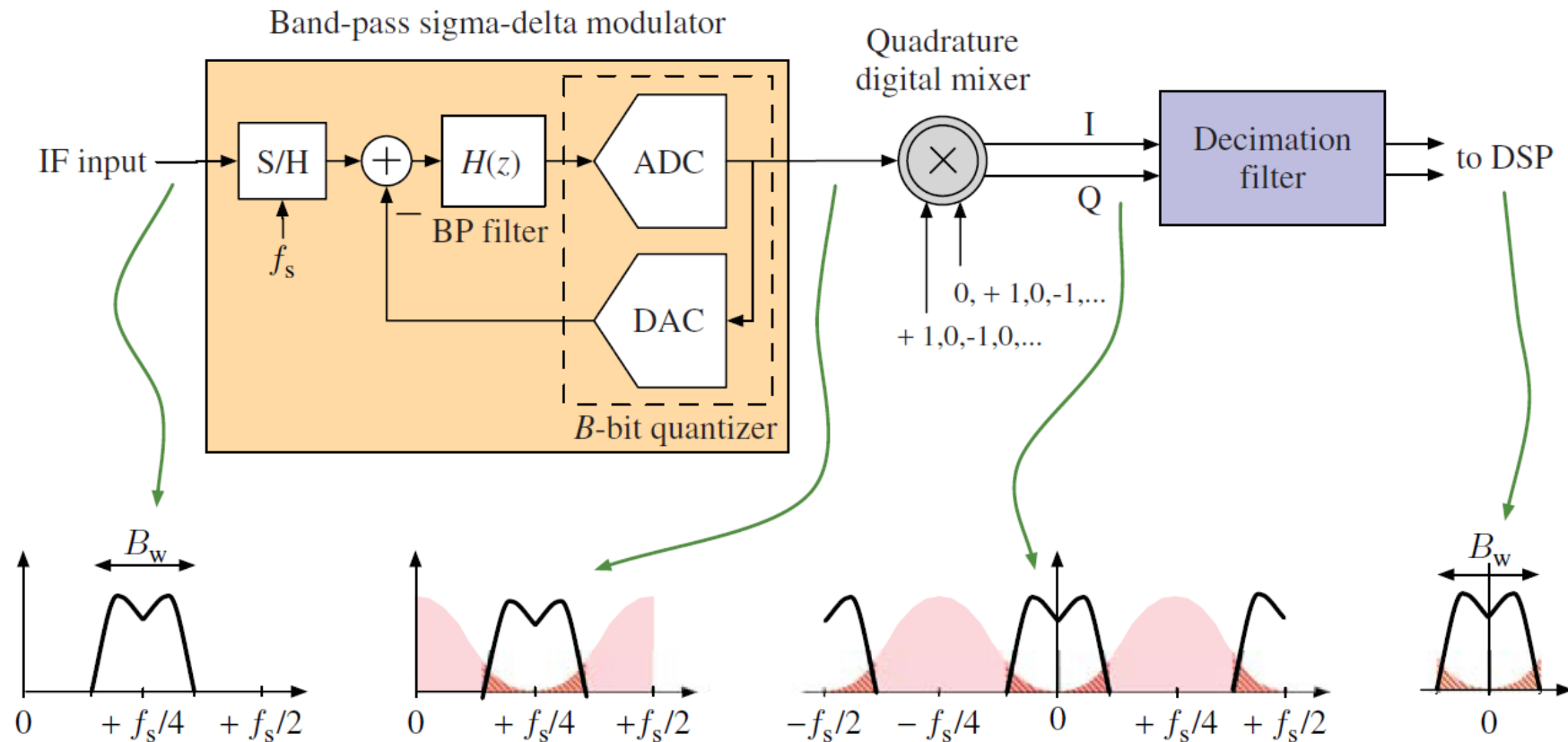
$$H_{\text{d3}}(z) = \frac{1}{c_1 c_2} (1 - z^{-1})^3$$

# More $\Sigma\Delta$ Ms Classification

- ❑ Low-Pass vs Band-Pass  $\Sigma\Delta$ Ms
- ❑ Discrete-Time vs Continuous-Time  $\Sigma\Delta$ Ms
  - Discrete-time: uses DT filter (switched capacitor filter)
  - Continuous-time: uses CT filter (e.g., Gm-C filter)

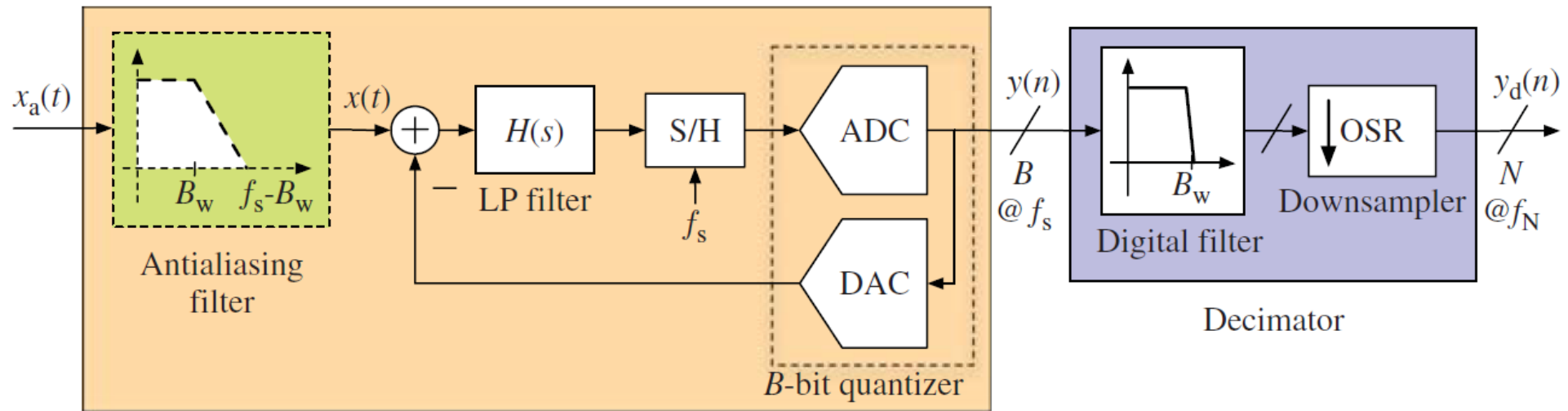
# Band-pass SDM

- ❑ In low-pass SDM, the NTF is HPF
- ❑ In band-pass SDM, the NTF is band-stop filter
- ❑ BP-SDM is used for digitizing IF signals in wireless receivers



# Continuous-Time (CT) SDMs

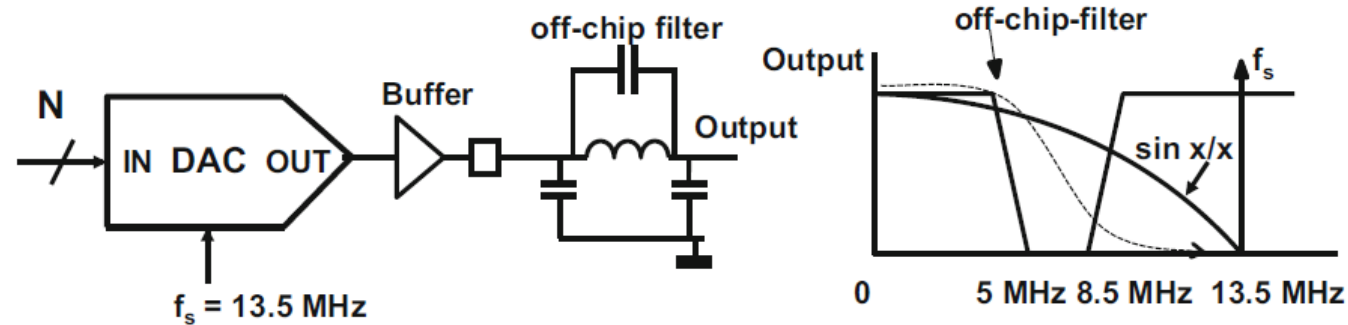
- ❑ The majority of SDMs are implemented using SC DT circuits
- ❑ CT SDMs can operate at higher sampling rates with lower power consumption
- ❑ The sampling operation is moved just before the quantizer



Continuous-Time Sigma-Delta modulator

# Nyquist vs Oversampling DAC

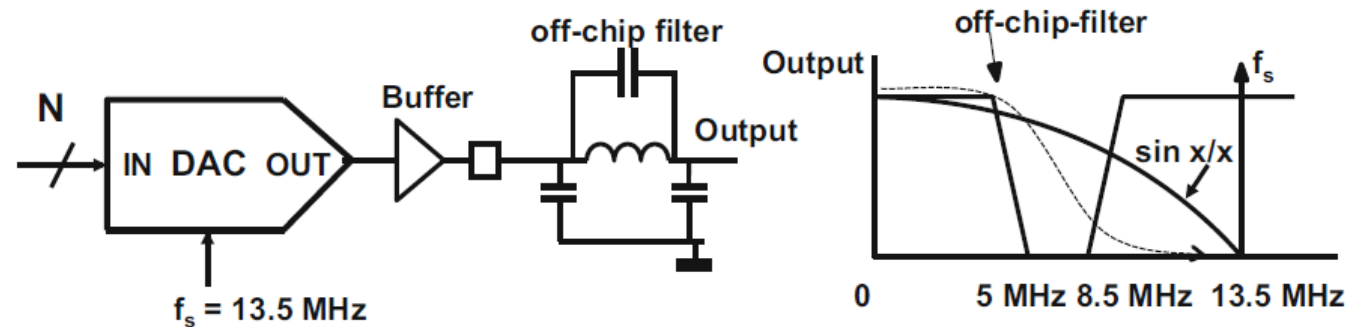
- ❑ DAC with BW close to Nyquist limit
  - Requirements on the filter (sharpness) and the buffer (SR) become hard to meet



# Nyquist vs Oversampling DAC

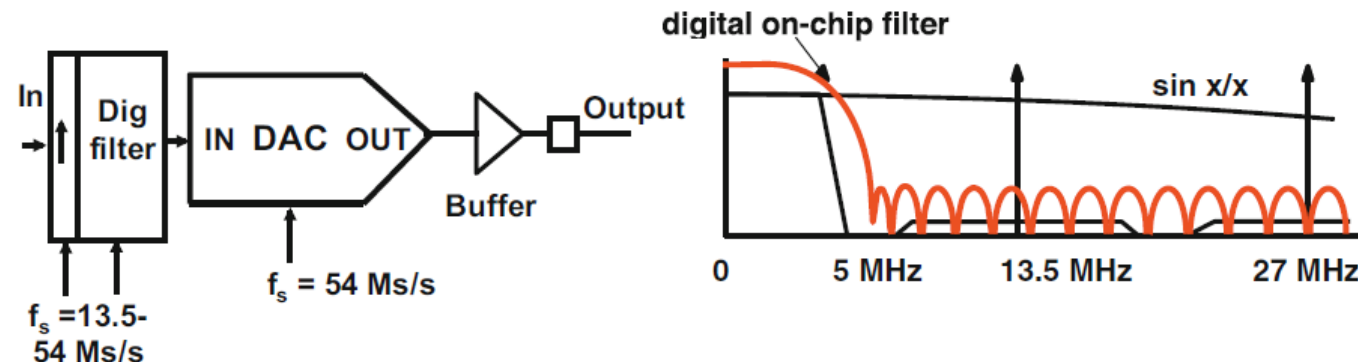
## ❑ DAC with BW close to Nyquist limit

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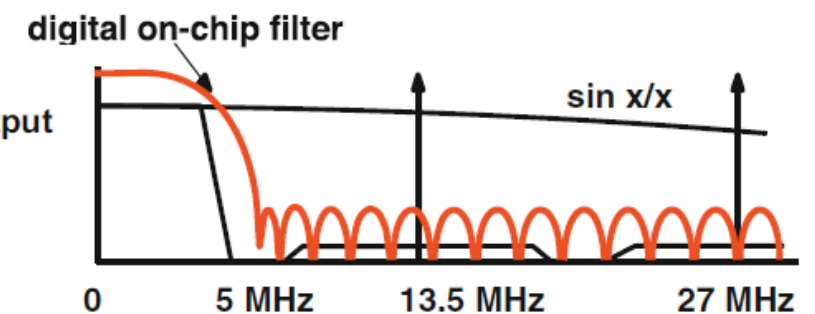
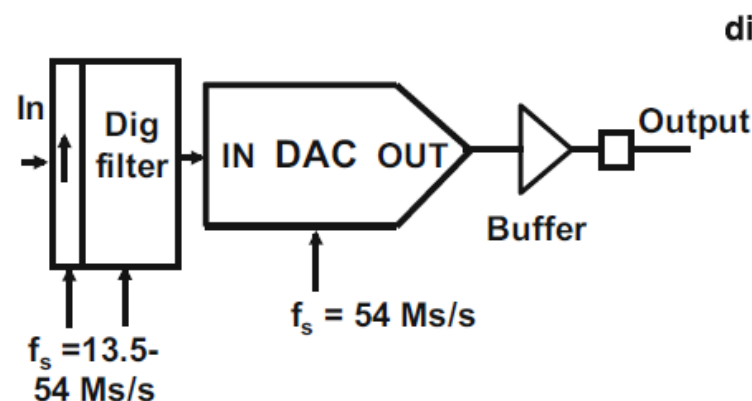
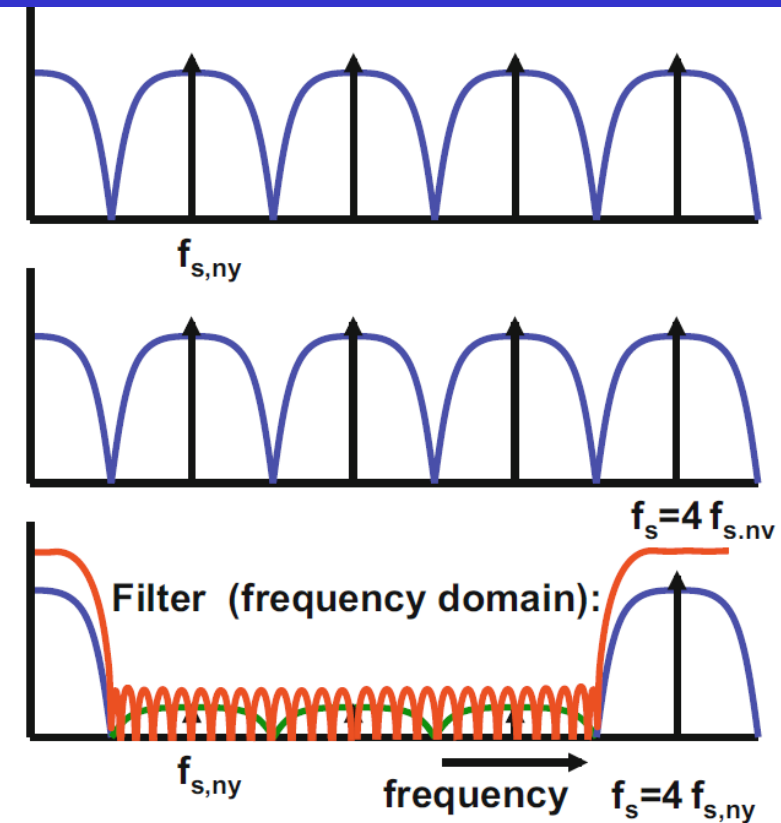
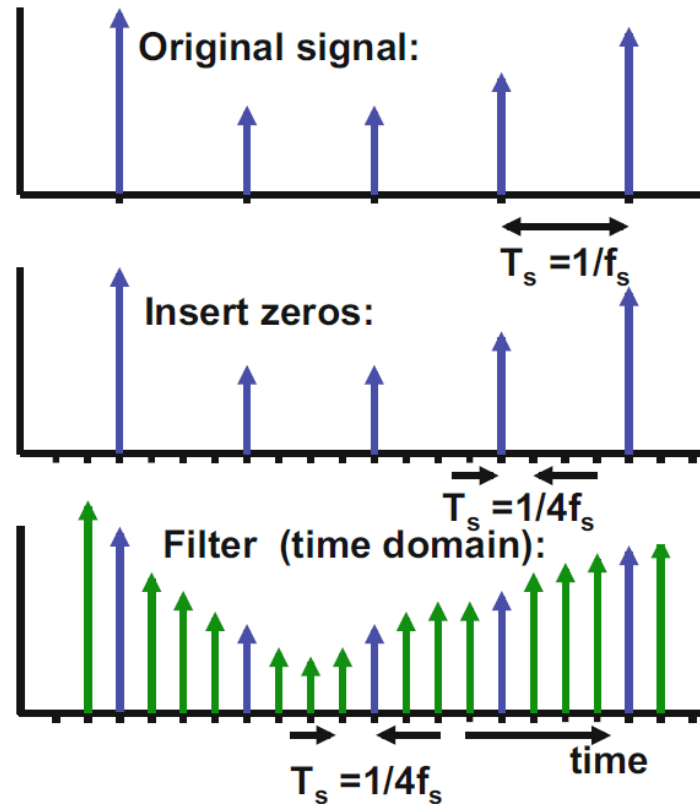
## ❑ Oversampling DAC

- Simpler filter and buffer  $\rightarrow$  but digital filtering required
- $\text{sinc}(x)$  distortion reduced  $\rightarrow$  no  $\text{sinc}(x)$  compensation required



# Oversampling DAC

- ❑ Up-sampling by zero stuffing
  - ❑ Digital filter is required after up-sampling to suppress alias bands
    - Performs interpolation in digital domain
- ❑ Smaller transient steps at output
  - Buffer SR and distortion specs are relaxed



# Nyquist vs Oversampling DAC

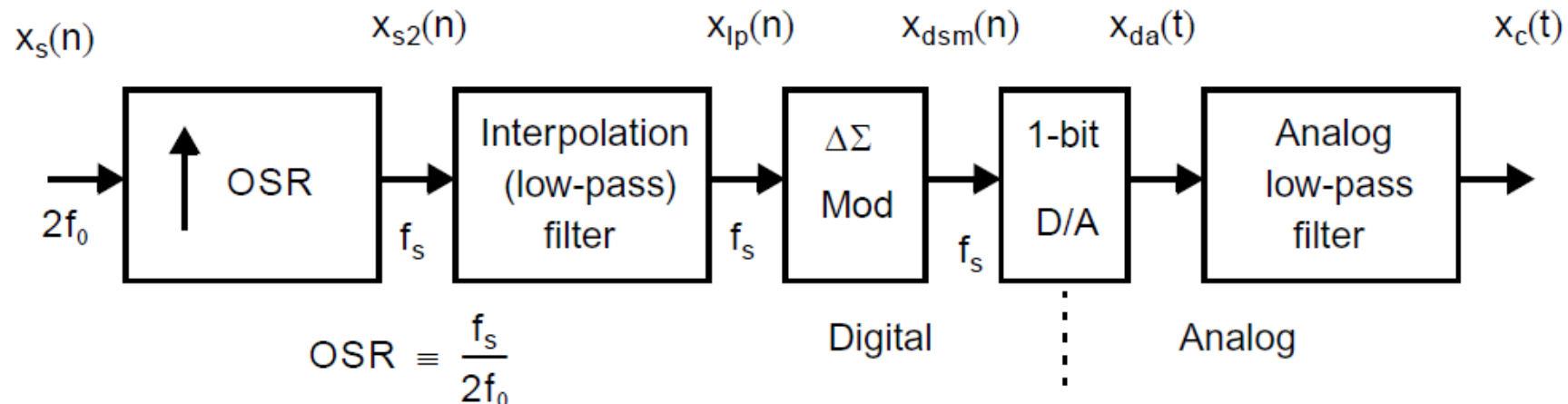
- ❑ Trade-off between analog buffer and filter power/complexity and digital filter power/area
  - Modern CMOS technologies favors the oversampling solution
  - Area and power of the digital filter shrinks and the switching speed of the short channel transistors allows high oversampling frequencies
- ❑ Comparison for OSR = 4:

Nyquist rate solution	Oversampling solution
External multi-pole filter needed	Internal digital CMOS filter
High slew current in driver	Medium current in driver power for digital filter
$\sin(x)/x$ loss of 2 dB	$\sin(x)/x$ loss = 0.3 dB
Standard sample rate	4× sample rate needed

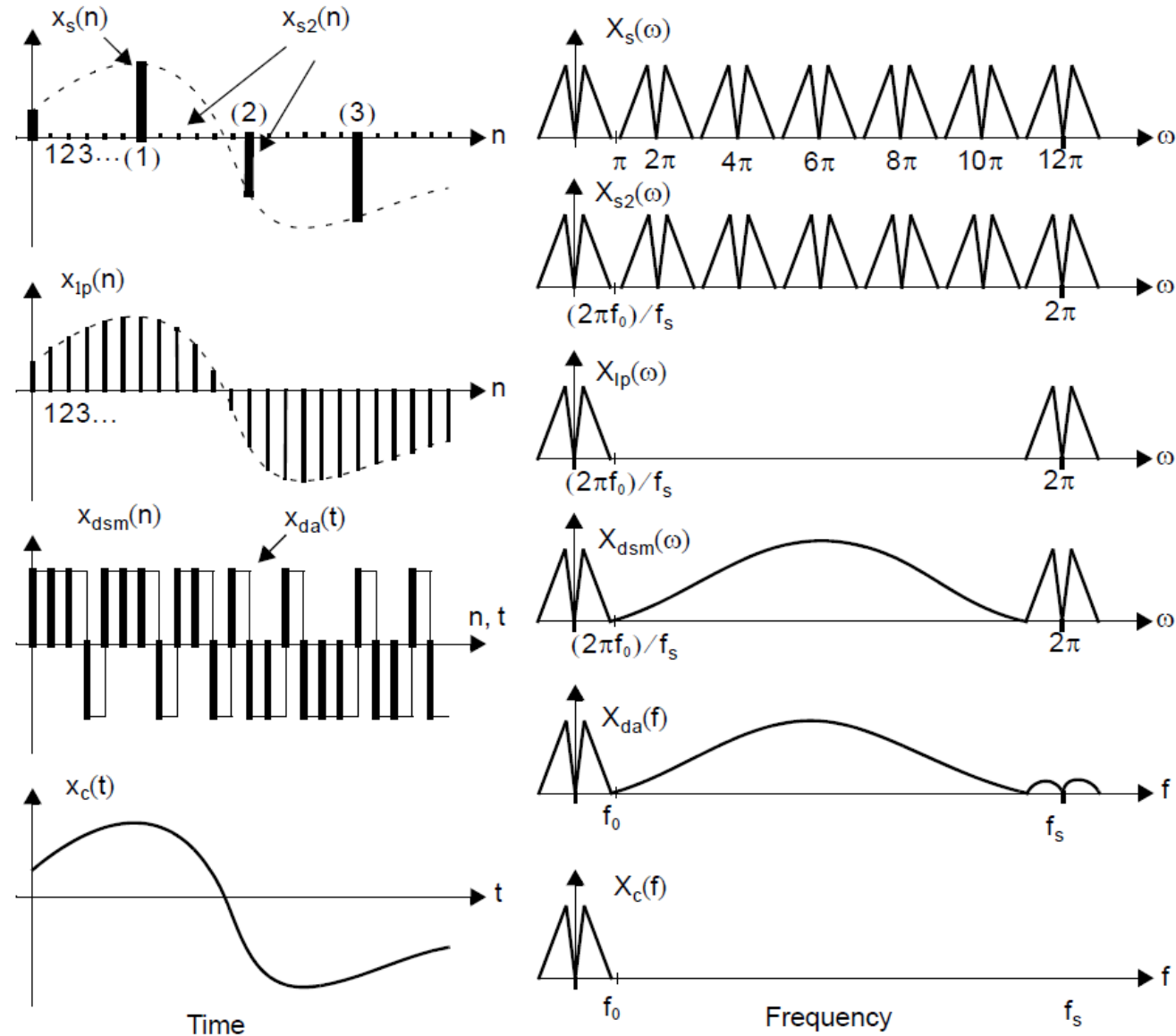


# Sigma-Delta DAC

- ❑ The order of the analog low pass filter should be at least one order higher than that of the modulator.
  - If the analog filter's order is equal to that of the modulator, the slope of the rising quantization noise will match the filter's falling attenuation
- ❑ Oversampling is often used with multi-bit D/A converters to reduce this analog-smoothing filter's complexity



# Sigma-Delta DAC



# References

- ❑ M. Pelgrom, Analog-to-Digital Conversion, Springer, 3<sup>rd</sup> ed., 2017.
- ❑ J. M. de la Rosa and R. del Rio, CMOS Sigma-Delta Converters: Practical Design Guide, Wiley, 2013.
- ❑ T. C. Carusone, D. Johns, and K. W. Martin, “Analog Integrated Circuit Design,” 2<sup>nd</sup> ed., Wiley, 2012.
- ❑ Y. Chiu, EECT 7327, UTD.

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**Thank you!**