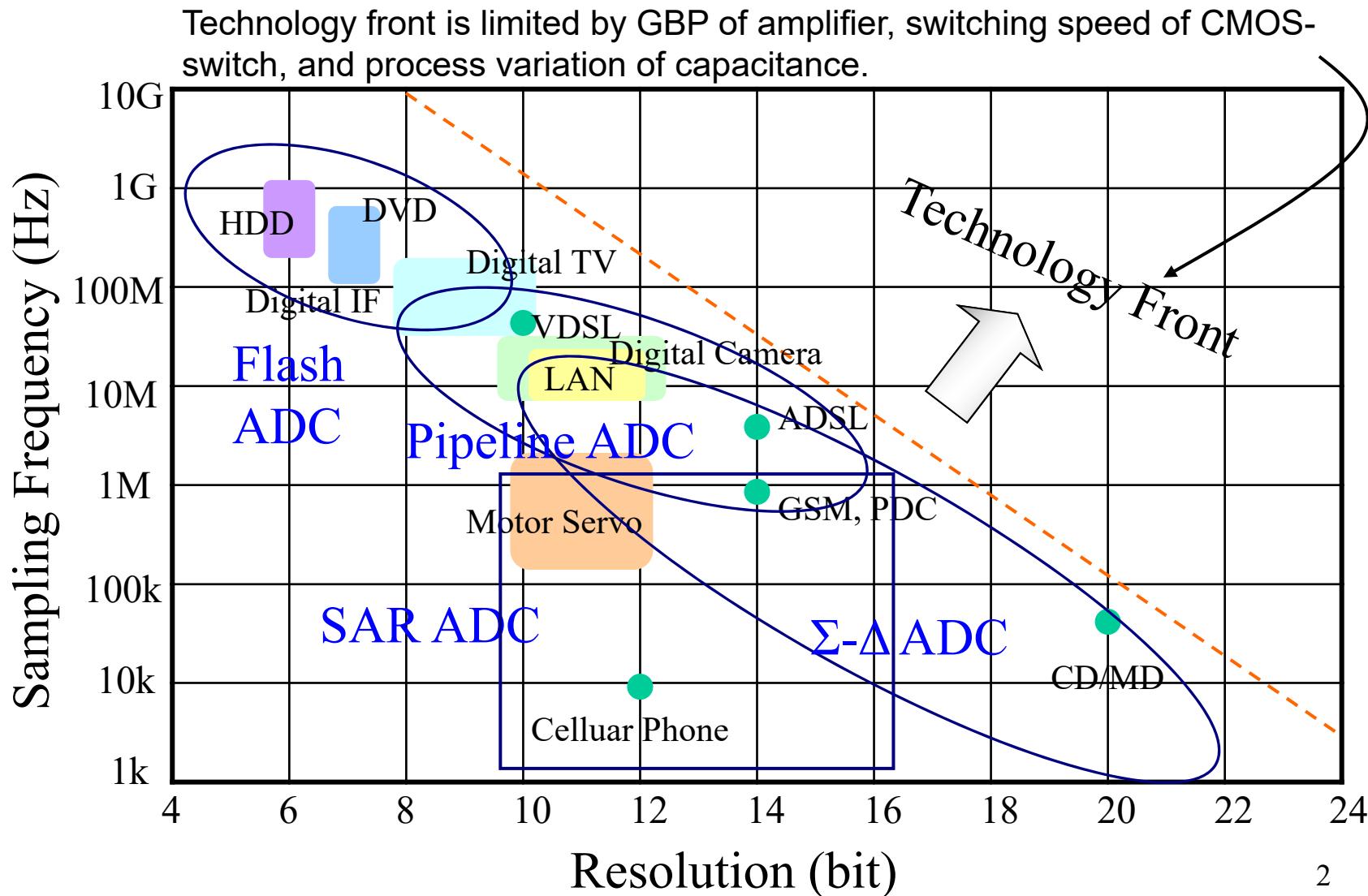


# 3. Data converters 1

(Oversampling converters)

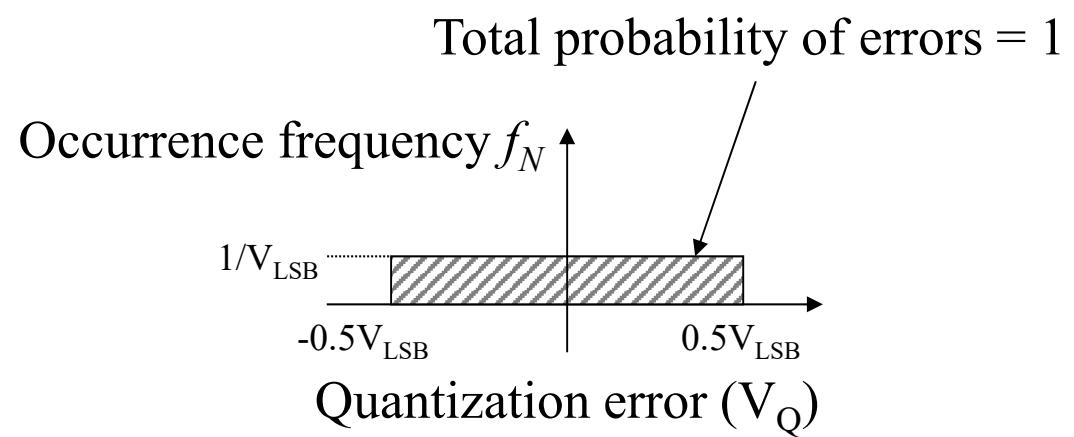
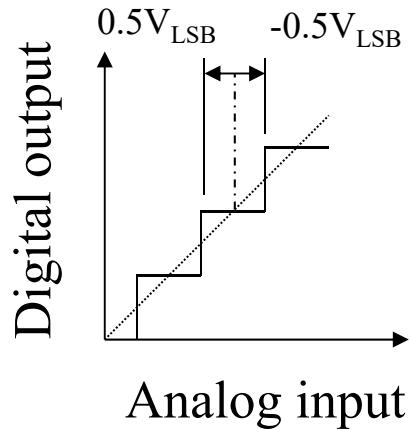
Kanazawa University  
Microelectronics Research Lab.  
Akio Kitagawa

# Performance of ADC architecture



# 3.1 Quantization noise and noise shaping

# Quantization noise



$$N\text{-bit quantization} \Leftrightarrow V_{\max} = (2^N - 1) \cdot V_{LSB}$$

$$\text{Noise level } V_{Q\_RMS} = \sqrt{\int_{-\infty}^{\infty} V_Q^2 f_N(V_Q) dV_Q} = \sqrt{\frac{1}{V_{LSB}} \int_{-0.5V_{LSB}}^{0.5V_{LSB}} V_Q^2 dV_Q} = \frac{V_{LSB}}{2\sqrt{3}}$$

$$\text{Maximum Signal level } V_{\max\_RMS} = \sqrt{\int_{-\infty}^{\infty} V_{in}^2 dV_Q} = \frac{V_{\max}}{2\sqrt{2}} = \frac{(2^N - 1)V_{LSB}}{2\sqrt{2}}$$

# SNR of quantized signal

$$SNR(\text{dB}) = 10 \log \frac{V_{\max\_RMS}^2}{V_{Q\_RMS}^2} = 10 \log \frac{\left( \frac{2^N - 1}{2\sqrt{2}} V_{LSB} \right)^2}{\left( \frac{V_{LSB}}{2\sqrt{3}} \right)^2} = 10 \log \frac{3}{2} (2^N - 1)^2$$
$$\cong 6.02N + 1.76 \quad (\text{dB})$$

# Oversampling

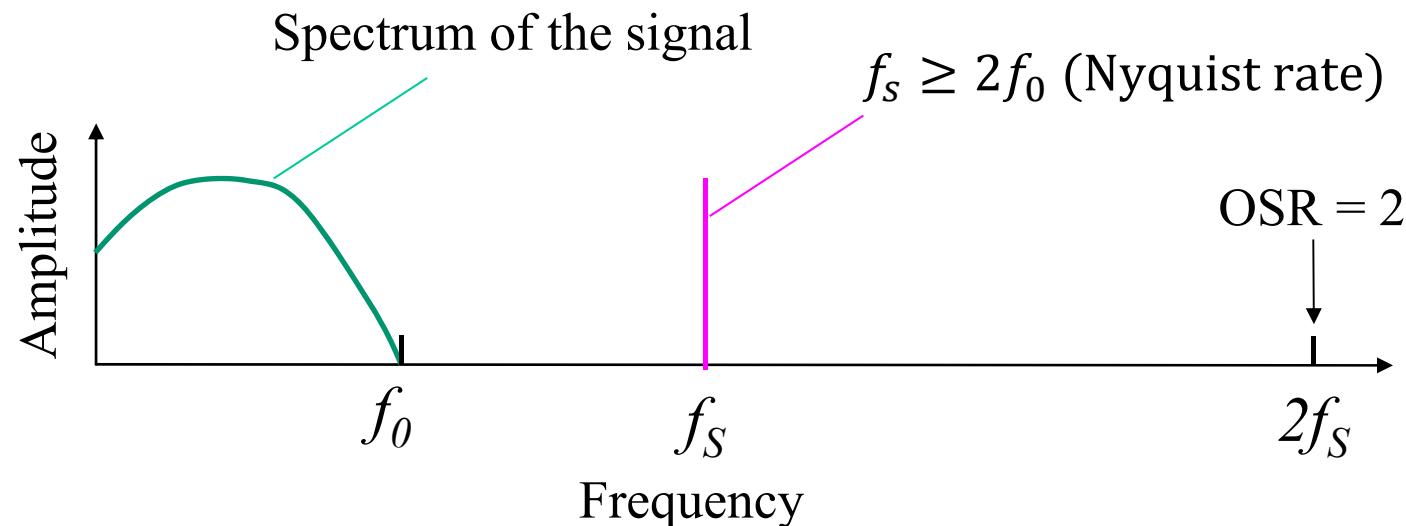
Oversampling ratio  $OSR \equiv \frac{f_s}{2 \cdot f_0}$

$f_s$  = Sampling rate

$f_0$  = Maximum signal frequency

Sampling theorem (Nyquist-Shannon's theorem):  $f_0 \leq \frac{f_s}{2}$

The minimum sampling frequency  $2f_0$  called Nyquist rate.



# Quantization noise by oversampling

Assuming that the quantization error generates the white noise,

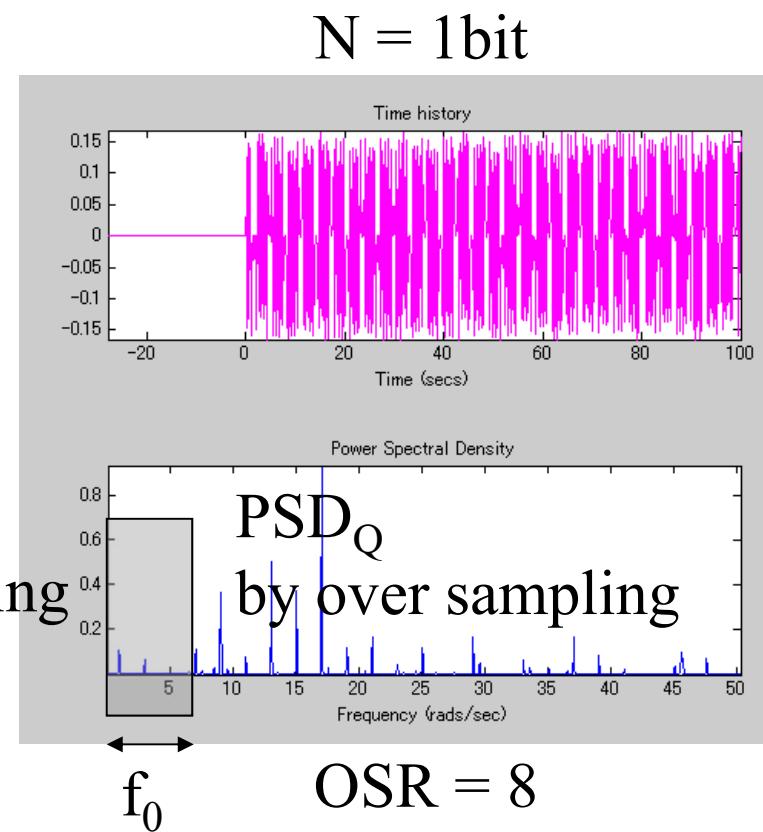
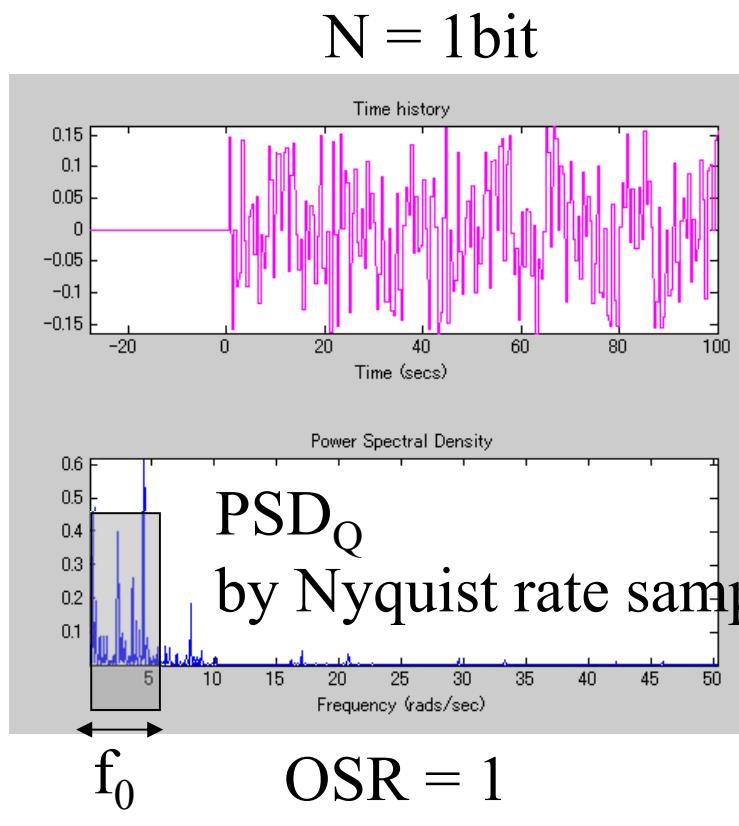
$$\text{Noise power spectrum density } PDS_Q = \frac{V_{Q\_RMS}^2}{f_s} = \frac{\left(\frac{V_{LSB}}{2\sqrt{3}}\right)^2}{f_s} = \frac{V_{LSB}^2}{12f_s}$$

$$\text{Noise power level in band } P_{Q\_RMS} = \int_{-f_0}^{f_0} PSD_Q df = \frac{V_{LSB}^2}{12f_s} 2f_0 = \frac{V_{LSB}^2}{12} \frac{1}{OSR}$$

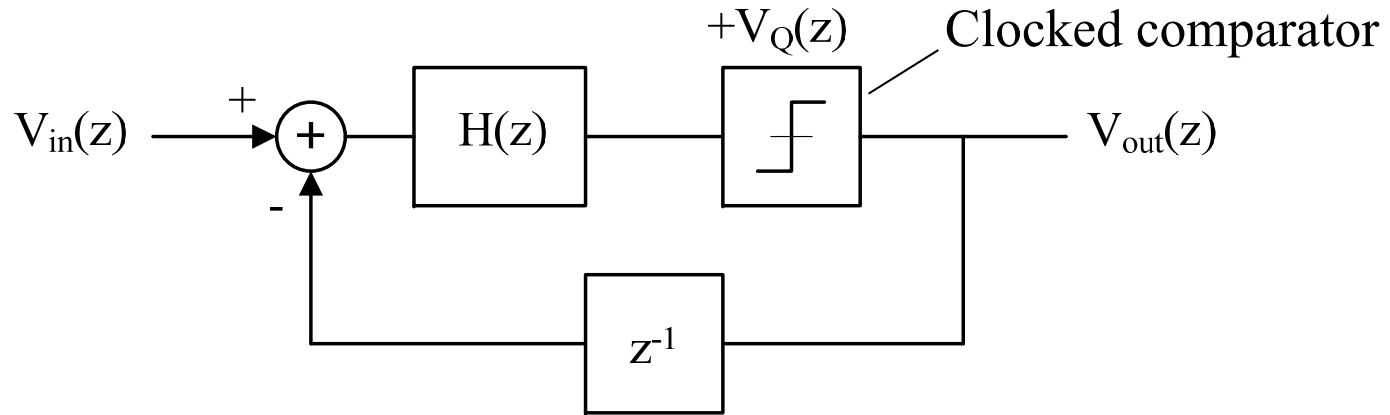
$$\begin{aligned} \text{SNR in band width } SNR(\text{dB}) &= 10 \log \frac{V_{\max\_RMS}^2}{P_{Q\_RMS}} = 10 \log \frac{\left(\frac{2^N - 1}{2\sqrt{2}} V_{LSB}\right)^2}{\frac{V_{LSB}^2}{12}} OSR \\ &= 10 \log \frac{3}{2} (2^N - 1)^2 + 10 \log OSR \\ &\approx 6.02N + 1.76 + 10 \log OSR \quad (\text{dB}) \end{aligned}$$

# Noise reduction by over sampling

Quantization error



# Delta-sigma modulator



$$V_{out}(z) = H(z)\{V_{in}(z) - z^{-1}V_{out}(z)\} + V_Q(z)$$

$$V_{out}(z) = \frac{H(z)}{1 + z^{-1}H(z)} V_{in}(z) + \frac{1}{1 + z^{-1}H(z)} V_Q(z)$$

For example,  $H(z) = \frac{1}{1 - z^{-1}}$  (BET Integral) Transfer function of noise

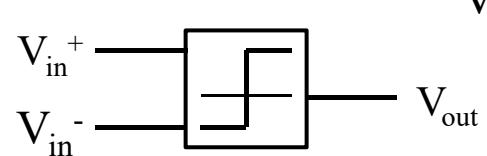
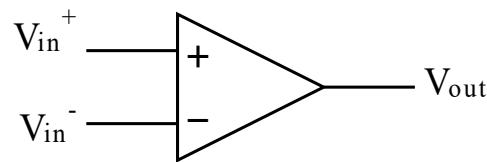
$$V_{out}(z) = V_{in}(z) + (1 - z^{-1})^M V_Q(z) \quad (\text{Order } M = 1)$$

Zero @  $f = 0$

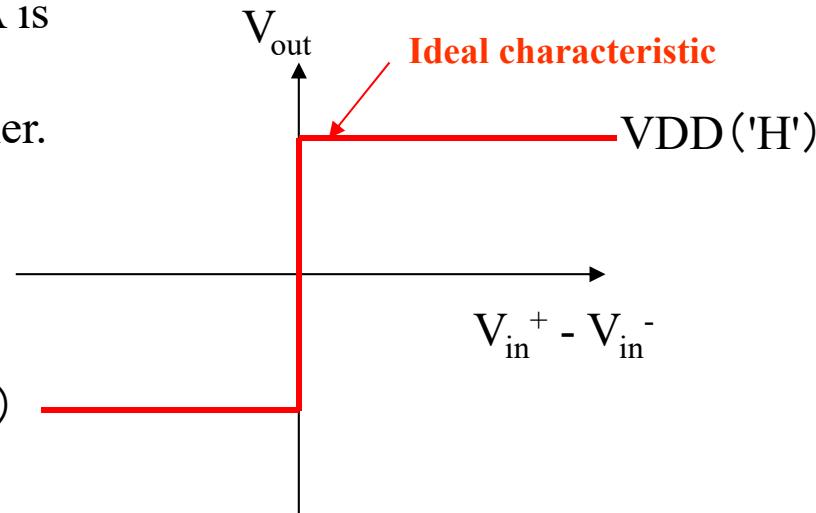
# Function of analog comparator

- If  $(V_{in}^+ > V_{in}^-)$ , then  $V_{out} = VDD$  (Logic level = 'H')
- If  $(V_{in}^+ < V_{in}^-)$ , then  $V_{out} = VSS$  (Logic level = 'L')

The same symbol as a single-end OPA is sometimes used for a comparator, but the circuitry is different from each other.



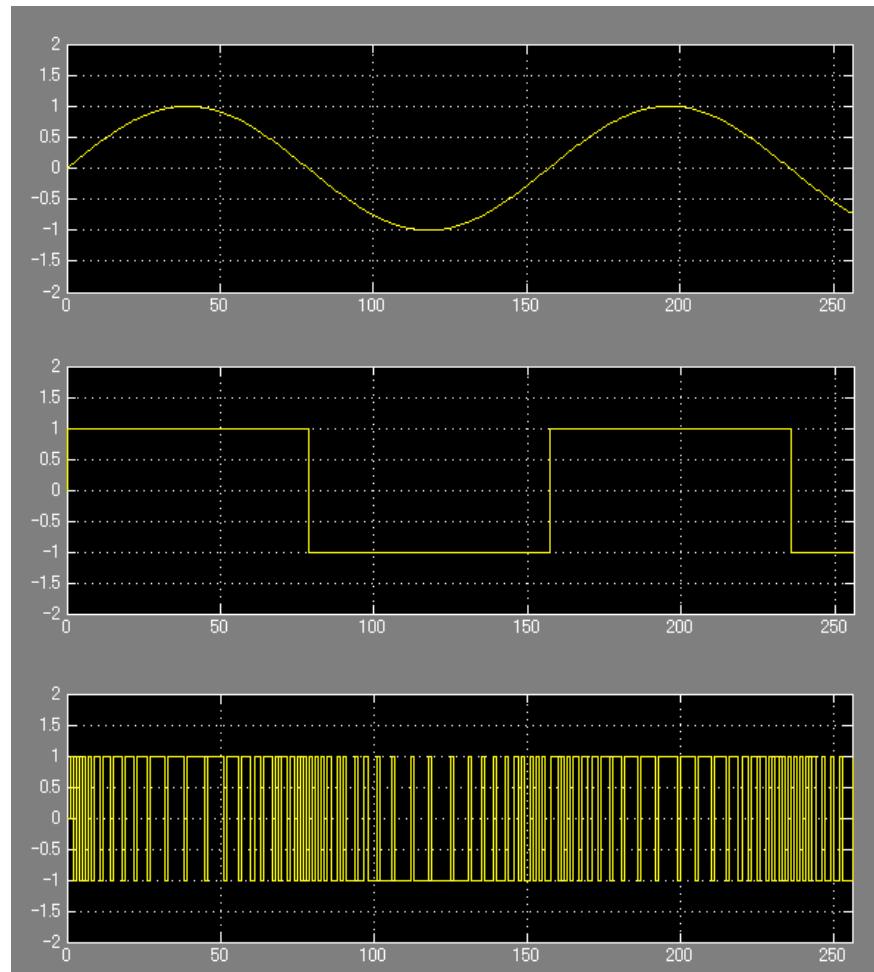
Symbol



# コンパレータの2つの機能

- 電圧差が0Vになるタイミングの検知(クロックなし)
- あるタイミングでの電圧の大小比較(クロックあり)
- ここでは、後者を用いる

# Time domain characteristic of delta-sigma modulator



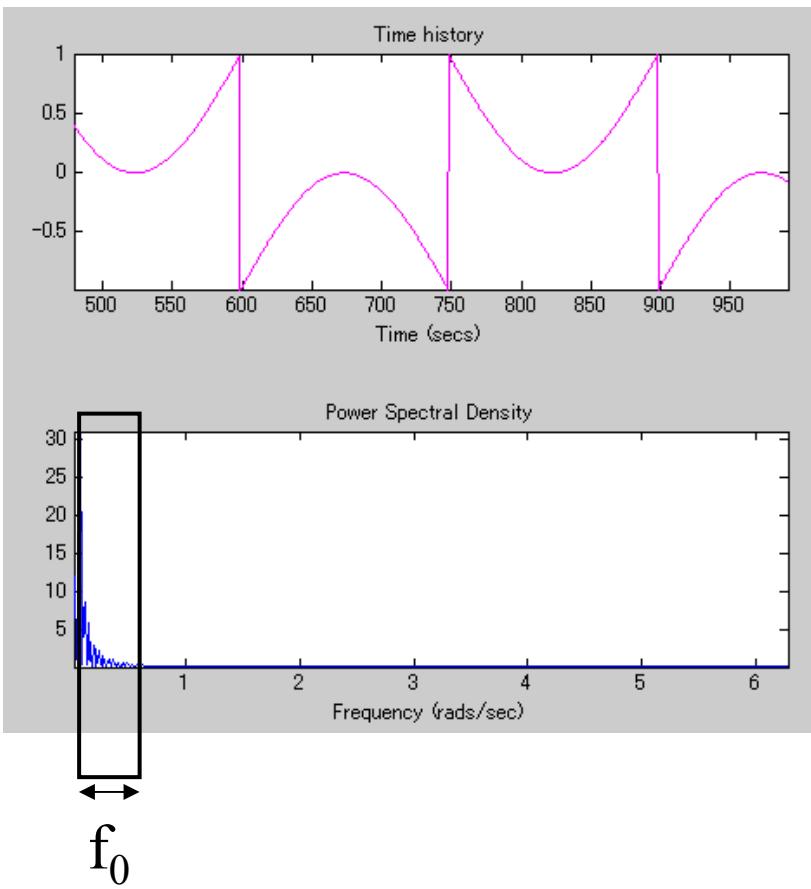
32 bit binary

Comparator  
(OSR = 24)

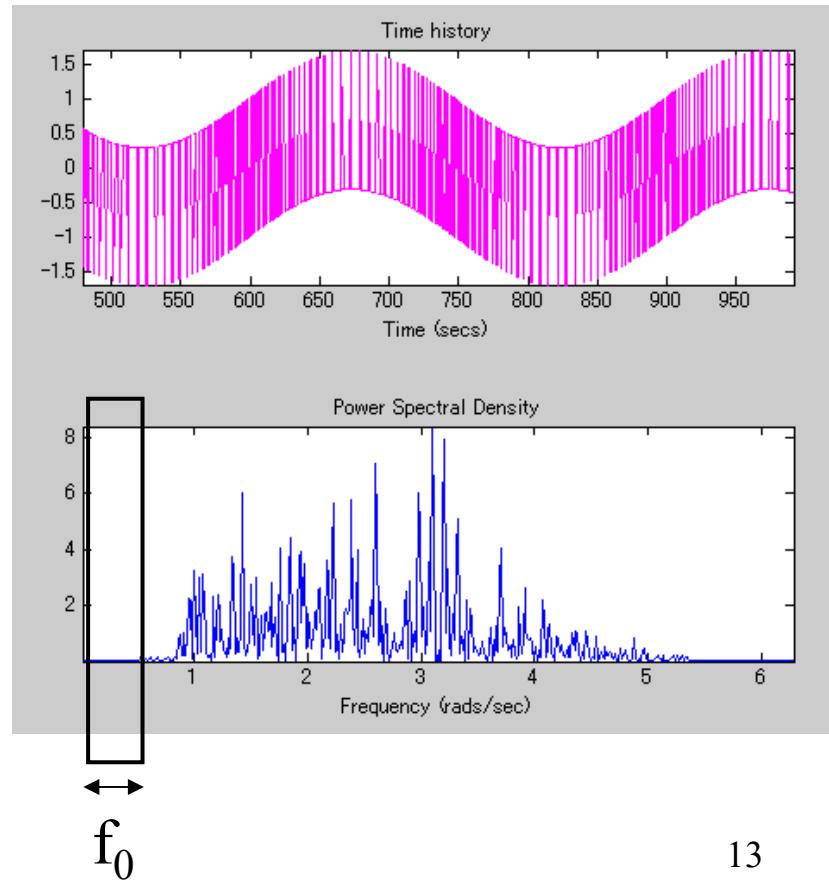
1-bit delta-sigma modulation  
= Pulse-density modulation  
(OSR = 24)

# Noise shaping by delta-sigma modulation

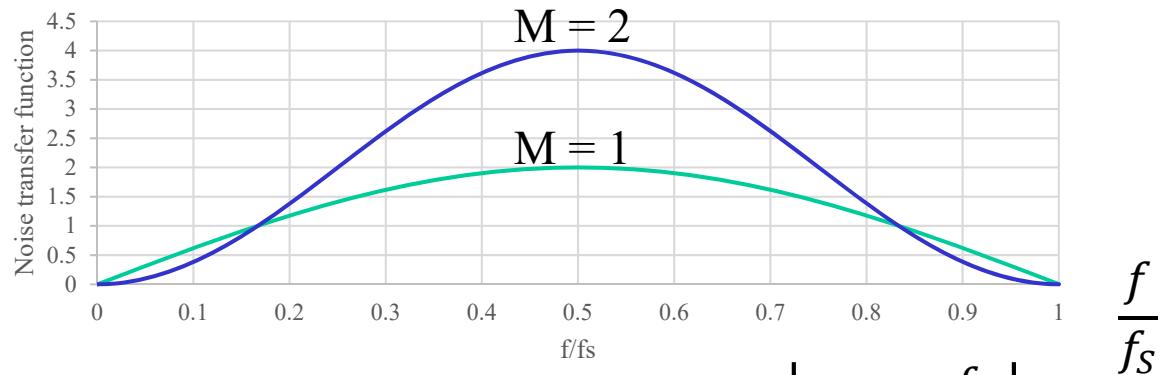
Error of comparator



Error of 1-bit delta-sigma modulation



# Noise reduction by noise shaping



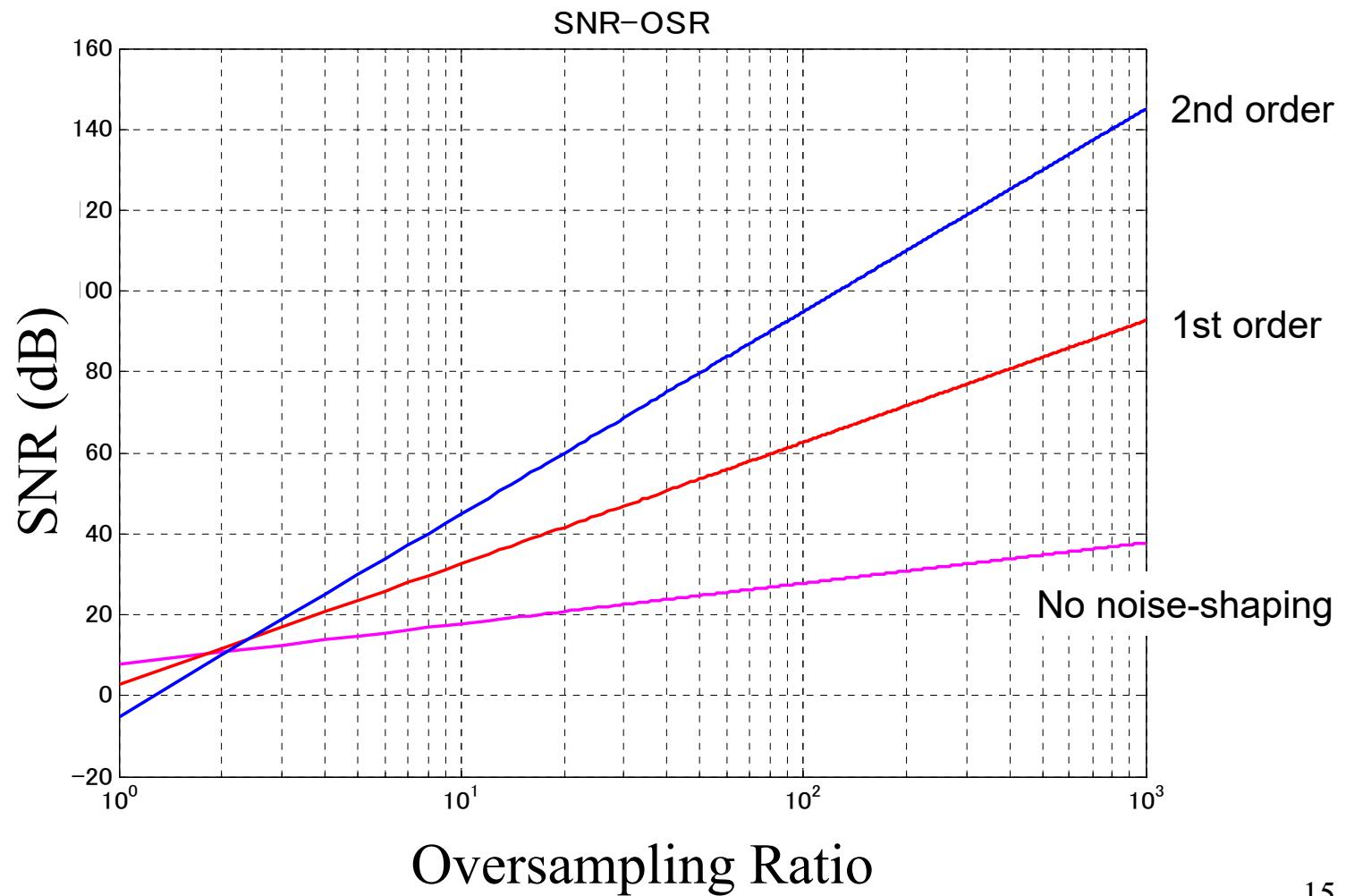
$$M=1 \quad |(1 - z^{-1})| = |1 - e^{-j\omega T_S}| = \left| 2\sin\left(\pi \frac{f}{f_S}\right) \right|$$

$$M=2 \quad |(1 - z^{-1})^2| = |1 - e^{-j\omega T_S}|^2 = \left| 2\sin\left(\pi \frac{f}{f_S}\right) \right|^2$$

$$SNR(dB) = \int_{-f_0}^{f_0} \frac{V_{\max\_RMS}^2}{P_{Q\_RMS} \cdot \left| 2\sin\left(\pi \frac{f}{f_S}\right) \right|^{2M}} df$$

$$\cong 6.02N + 1.76 - 20 \log \left( \frac{\pi^M}{\sqrt{2M+1}} \right) + (20M + 10) \log OSR$$

# SNR vs. OSR



# Effective number of bits (ENOB)

For 1-bit quantization (=Comparator),

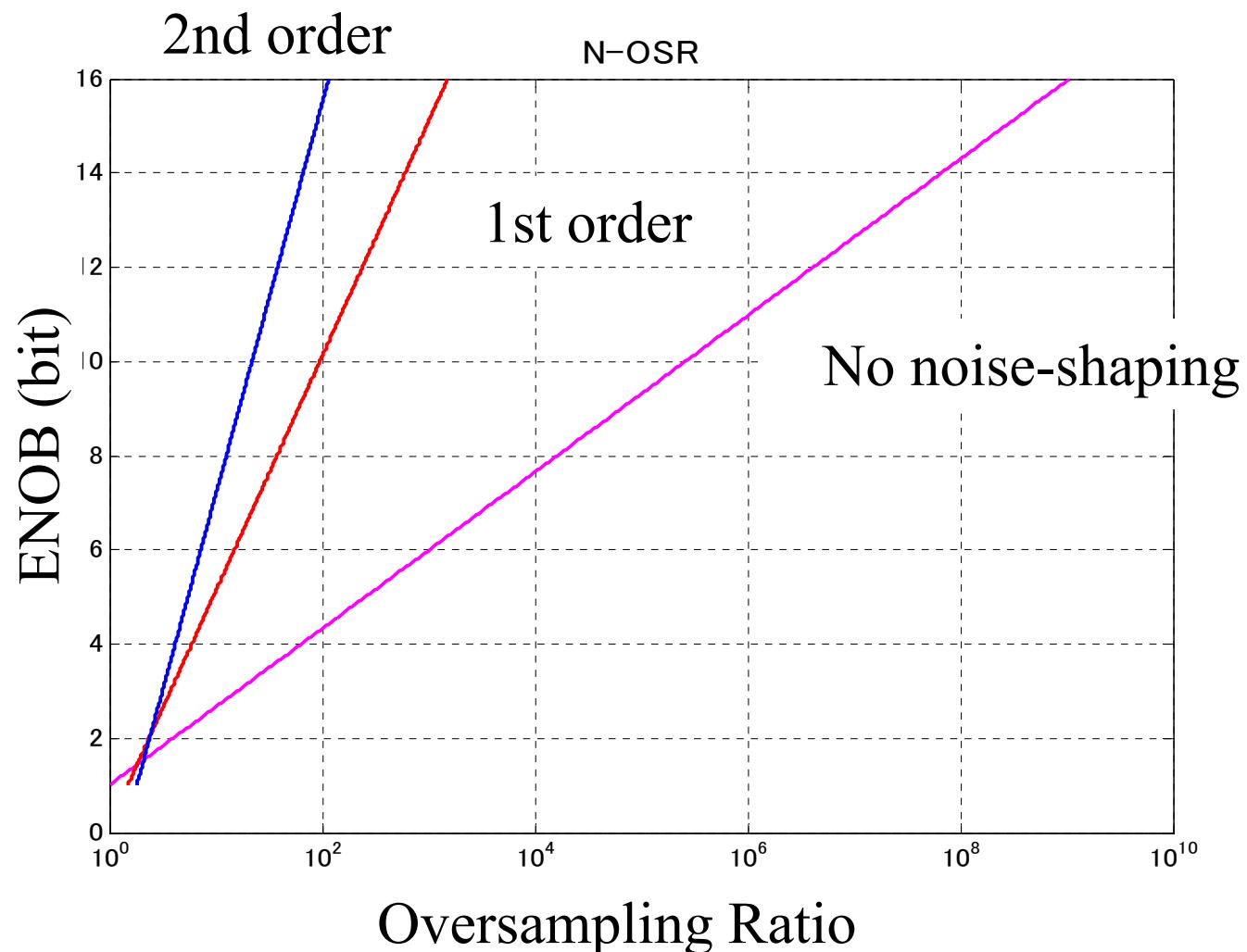
Simple quantization

Noise shaping ( $N = 1\text{bit}$ )

$$6.02ENOB + 1.76 = 6.02 + 1.76 - 20 \log\left(\frac{\pi^M}{\sqrt{2M+1}}\right) + (20M + 10)\log OSR$$

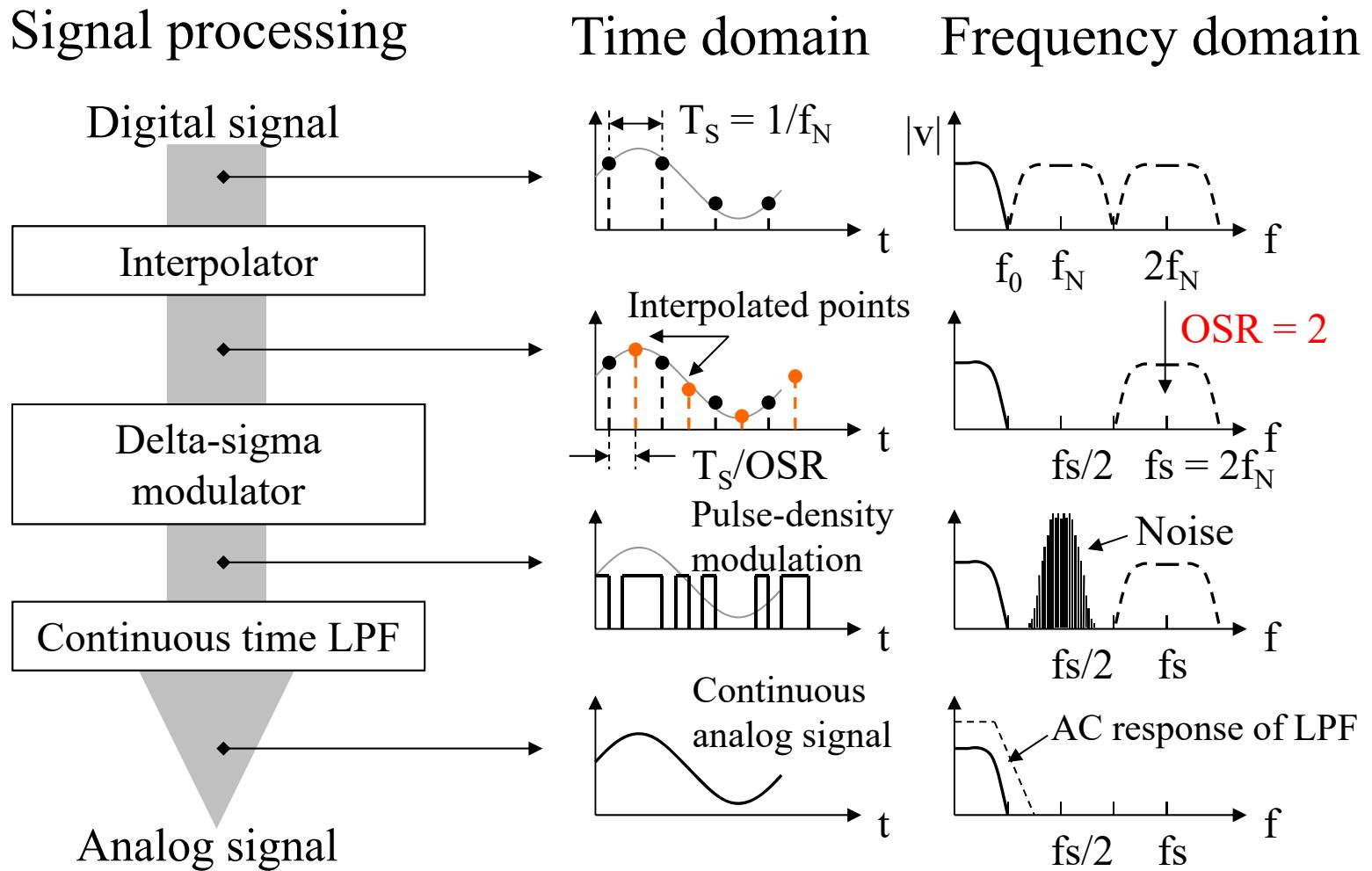
$$ENOB = 1 + \frac{1}{6.02} \left\{ (20M + 10)\log OSR - 20 \log\left(\frac{\pi^M}{\sqrt{2M+1}}\right) \right\}$$

# ENOB vs. OSR



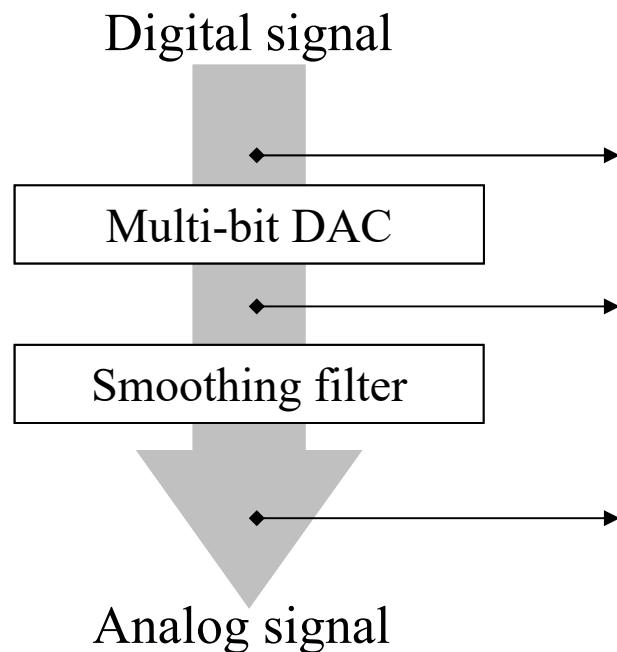
## 3.2 Oversampling DAC

# Principle of oversampling DAC

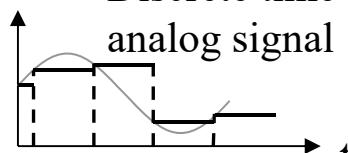
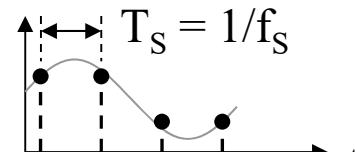


# Advantage of oversampling DAC

Signal processing of  
Nyquist rate DAC

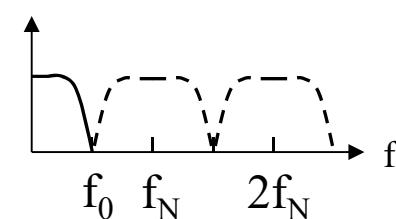


Time domain

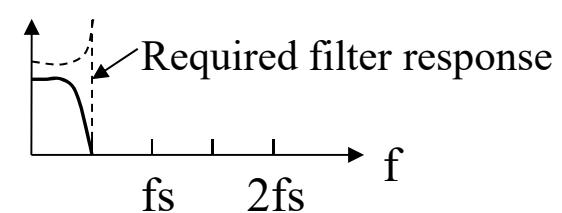
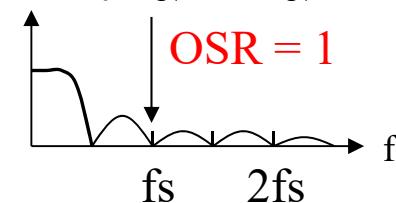


Continuous time  
analog signal

Frequency domain



$\text{OSR} = 1$



Nyquist rate DAC requires the ideal LPF for smoothing.

# Appendix: Frequency response of sample and hold circuit

Sample and hold of the signal  $x(t)$

$$x_{du}(t) = \sum_n x(nT_S) \cdot \underbrace{\{u(t - nT_S) - u(t - (n+1)T_S)\}}_{\text{Step sampler}}$$

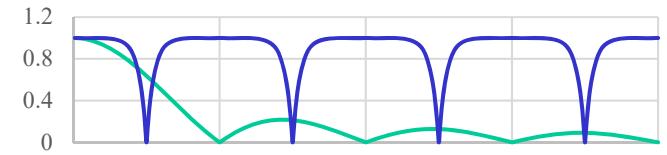
$\mathcal{L}$  

$$X_{du}(s) = \int_0^\infty \sum_n x(nT_S) \cdot \{u(t - nT_S) - u(t - (n+1)T_S)\} \cdot e^{-st} dt$$

$$= \sum_n x(nT_S) \cdot \int_0^\infty \{u(t - nT_S) - u(t - (n+1)T_S)\} \cdot e^{-st} dt$$

$$= \sum_n x(nT_S) \cdot \frac{e^{-snT_S} (1 - e^{-sT_S})}{s}$$

$$= \frac{1 - e^{-sT_S}}{s} \sum_n x(nT_S) \cdot e^{-snT_S} \xrightarrow{z} \underbrace{\frac{1 - e^{-sT_S}}{s}}_{\text{Transfer function of step sampler}} \underbrace{\sum_n x(nT_S) \cdot z^{-n}}_{\text{Impulse train}}$$

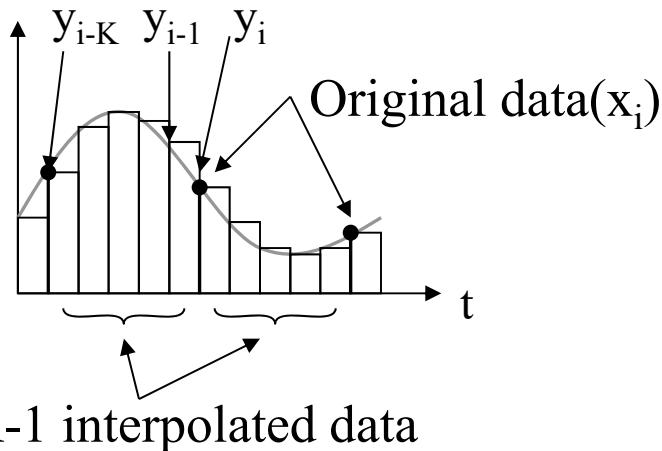


Transfer function of step sampler

Impulse train

# Interpolator

Example: Linear interpolation



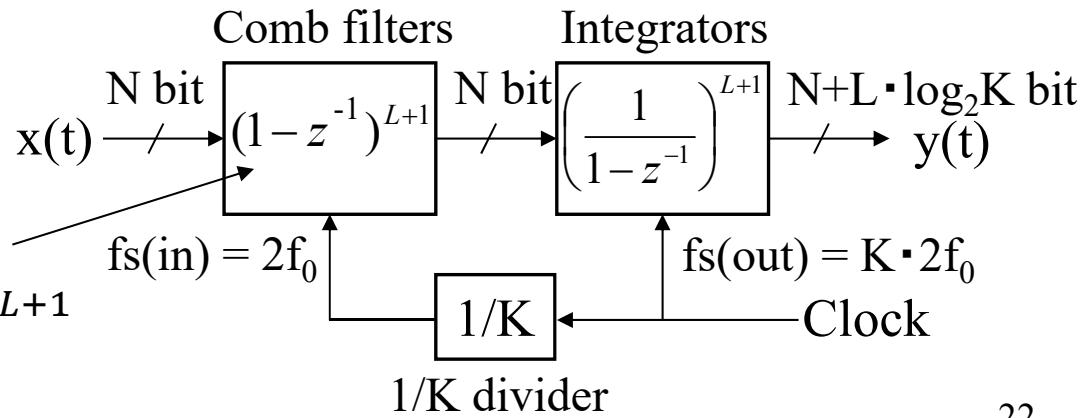
Sampling rate conversion  
(up conversion)

$$\frac{f_s}{K} = e^{-sKT_s} = z^{-K}$$

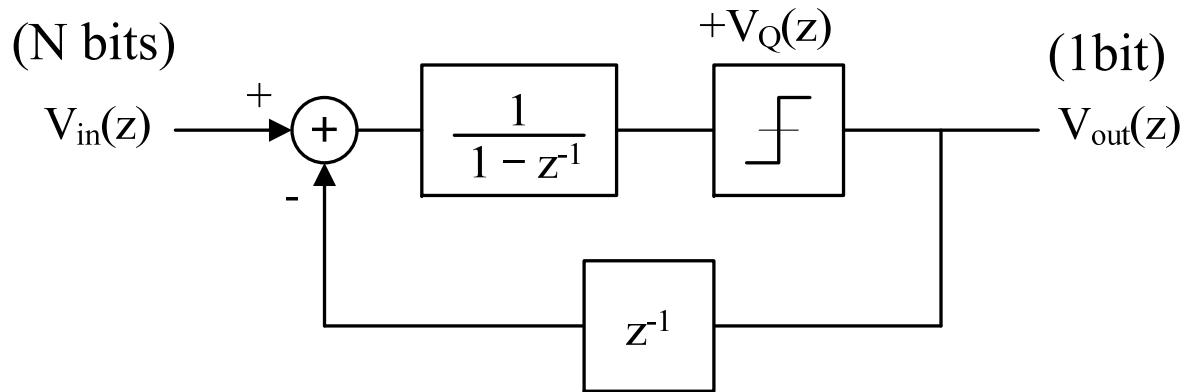
$$(1 - z^{-1})^{L+1} \xrightarrow{f_s/K} (1 - z^{-K})^{L+1}$$

$$\begin{aligned} \frac{y_i - y_{i-1}}{T_s} &= \frac{x_i - x_{i-K}}{K \cdot T_s} \\ Y(z)(1 - z^{-1}) &= X(z) \frac{1 - z^{-K}}{K} \\ H(z) &= \frac{Y(z)}{X(z)} = \frac{1}{K} \frac{1 - z^{-K}}{1 - z^{-1}} \end{aligned}$$

↓ Circuits implementation



# Pulse-density modulation by delta-sigma modulator (DSM)

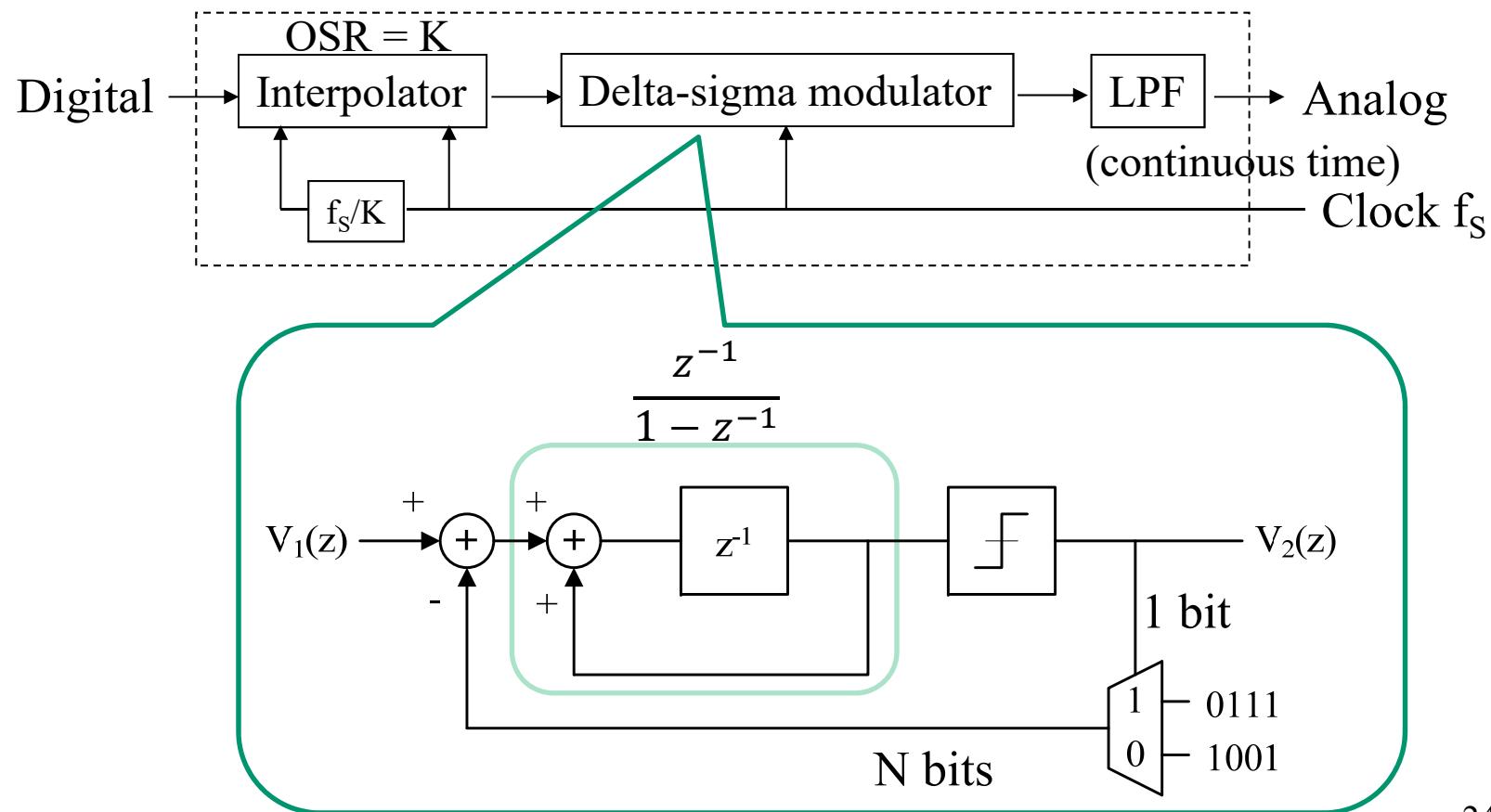


$$V_{out}(z) = V_{in}(z) + (1 - z^{-1})^M V_Q(z)$$

↑                   ↑                   ↑  
1 bit signal      N bits signal      Quantization noise

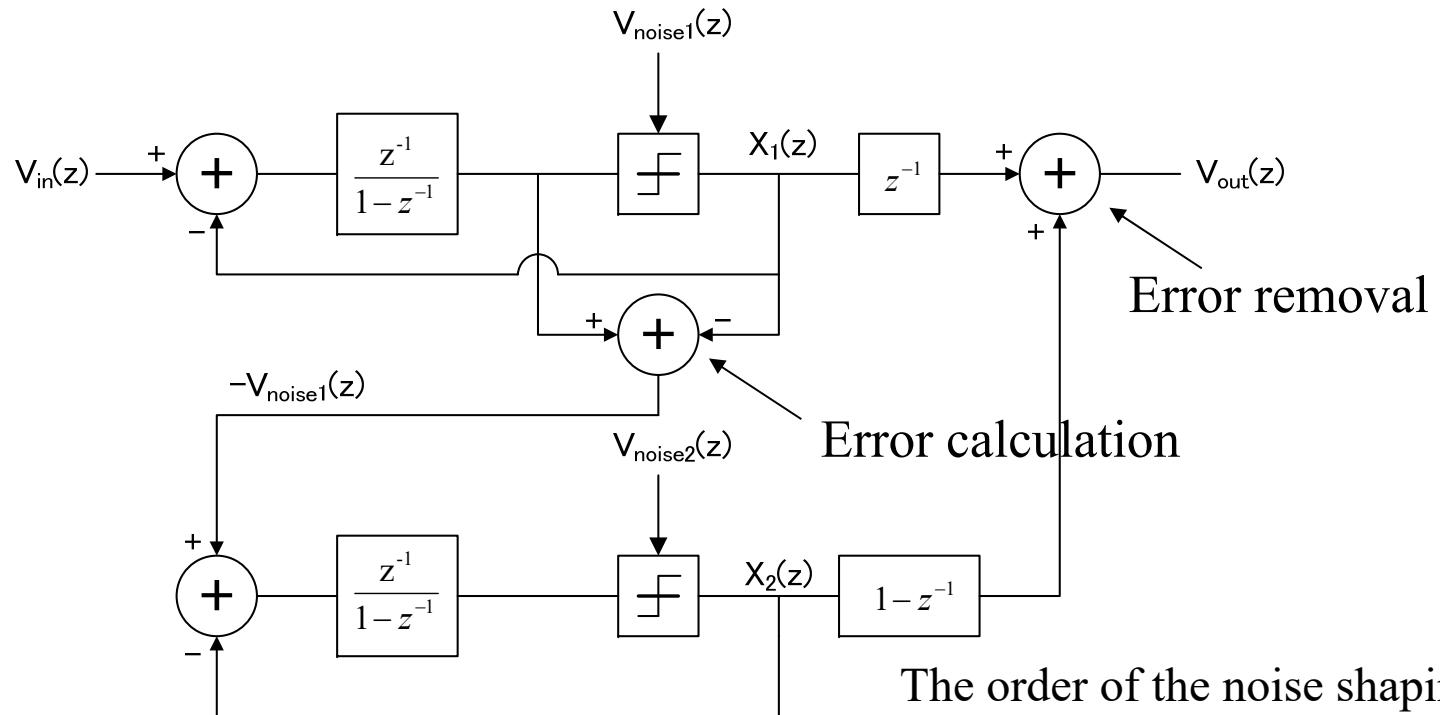
The figure shows the block diagram for the case of "M = 1".

# Architecture of oversampling DAC



# Higher order DSM

Multi-stage noise shaping (MASH)



$$X_1(z) = z^{-1}V_{in}(z) + (1-z^{-1})V_{noise1}(z)$$

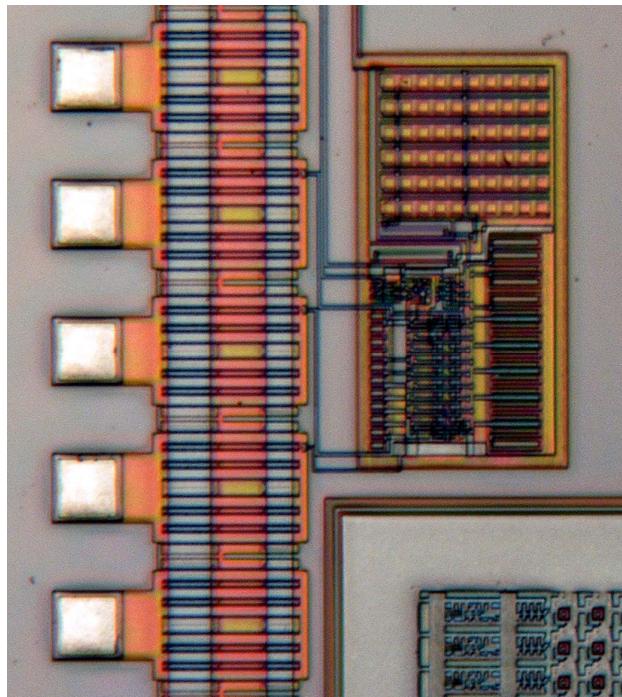
$$X_2(z) = -z^{-1}V_{noise1}(z) + (1-z^{-1})V_{noise2}(z)$$

$$V_{out}(z) = z^{-1}X_1(z) + (1-z^{-1})X_2(z) = z^{-2}V_{in}(z) + (1-z^{-1})^2V_{noise2}(z)$$

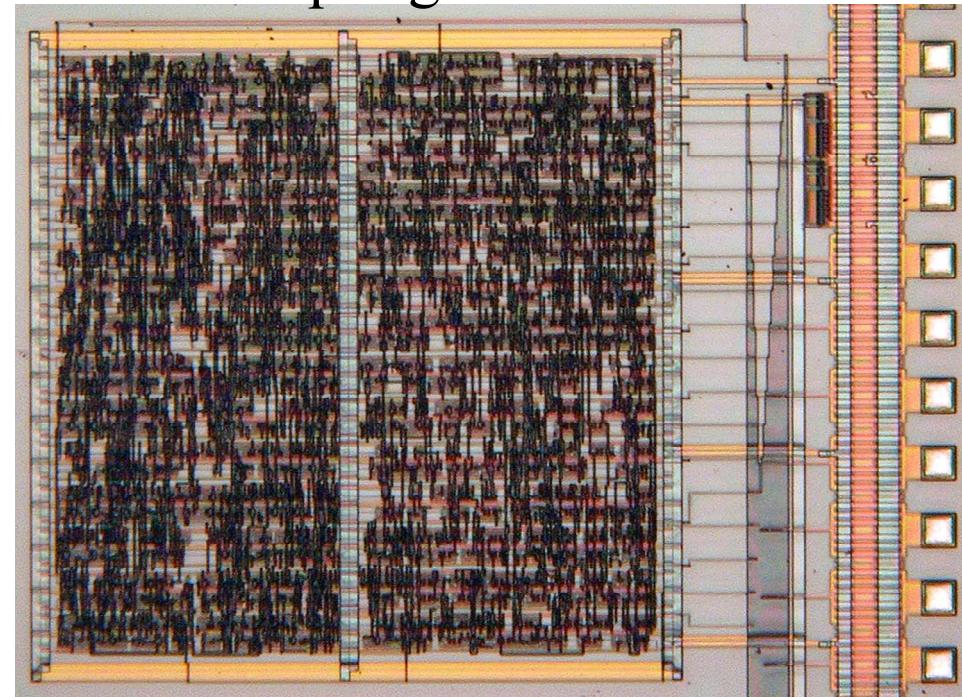
The order of the noise shaping  
is increased by repeating the  
error calculation.

# Example of oversampling DAC

Continuous time LPF



Oversampling DAC

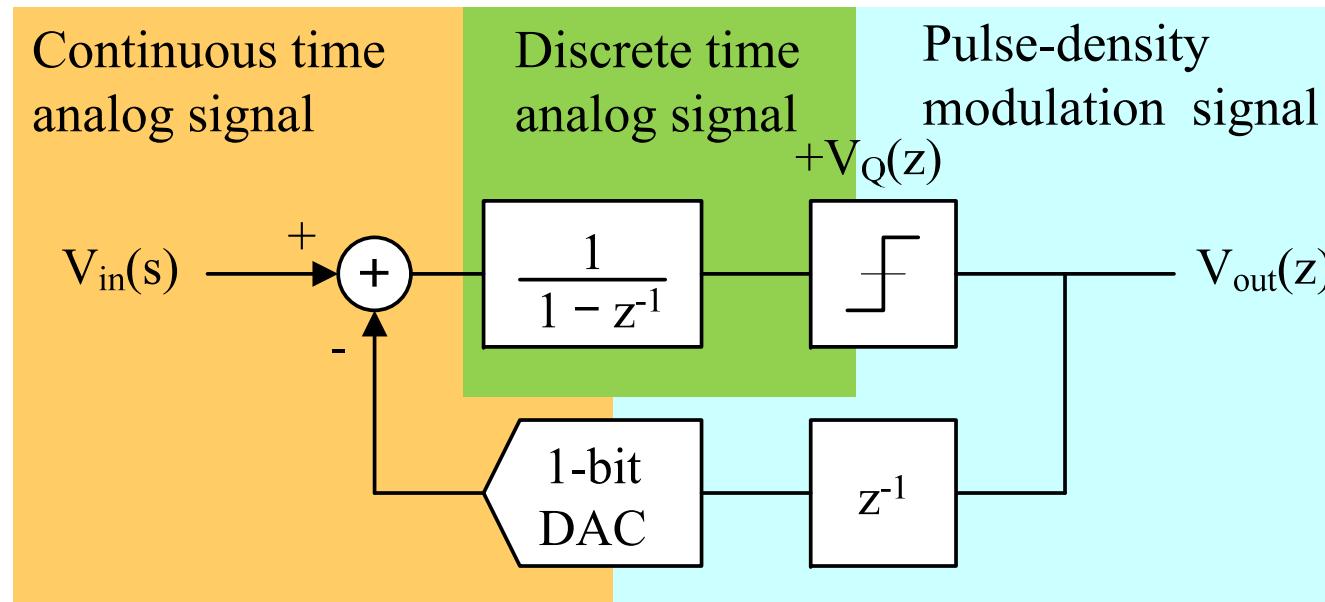


(c) Kanazawa University

SOI-CMOS 0.6um, 16bit, 2nd order DSM, OSR=128

## 3.3 Delta-Sigma ADC

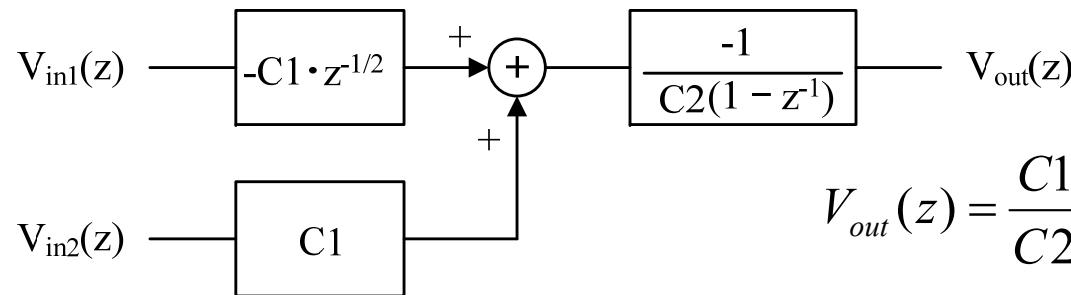
# Principle of analog DSM



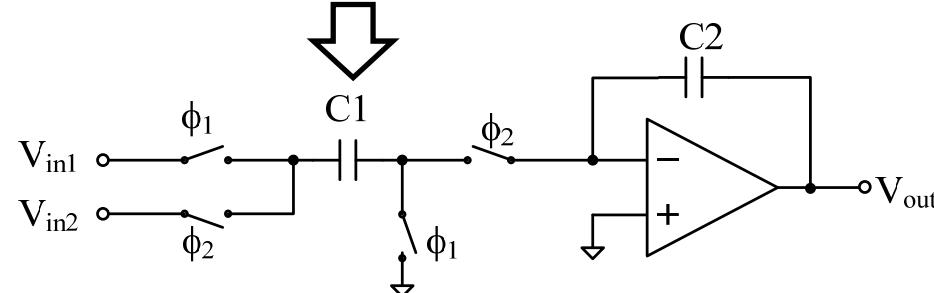
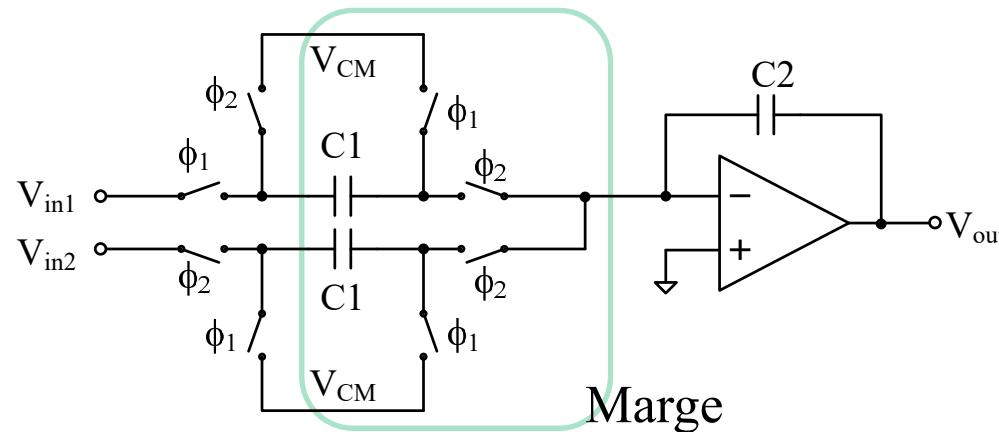
$$V_{out}(z) = V_{in}(z) + (1 - z^{-1})^M V_Q(z)$$

↑                      ↑                      ↑  
 Pulse-density      Continuous time      Quantization noise  
 modulated signal    analog signal

# DAI with reference input

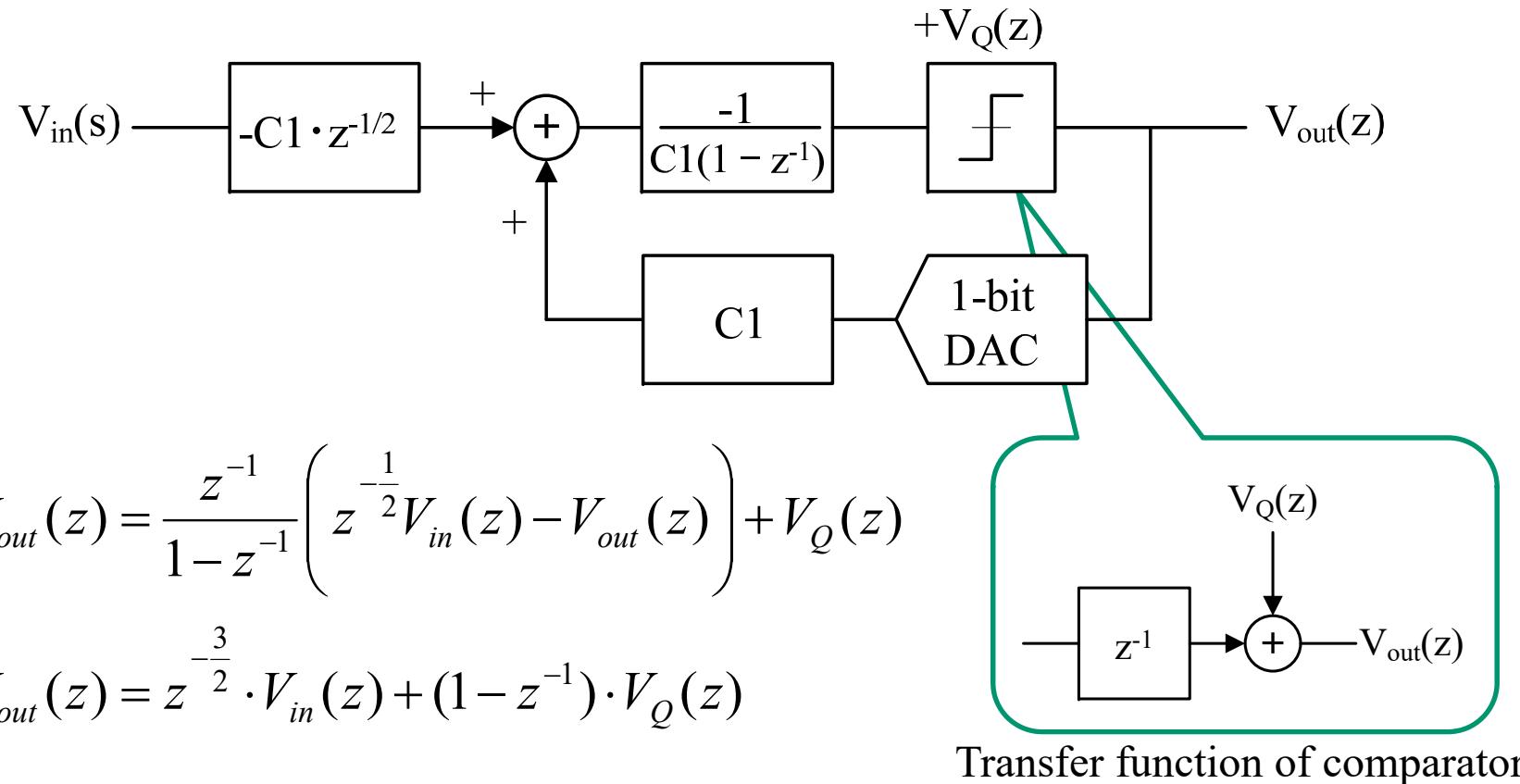


$$V_{out}(z) = \frac{C1}{C2} \frac{1}{1 - z^{-1}} \left\{ z^{-\frac{1}{2}} V_{in1}(z) - V_{in2}(z) \right\}$$

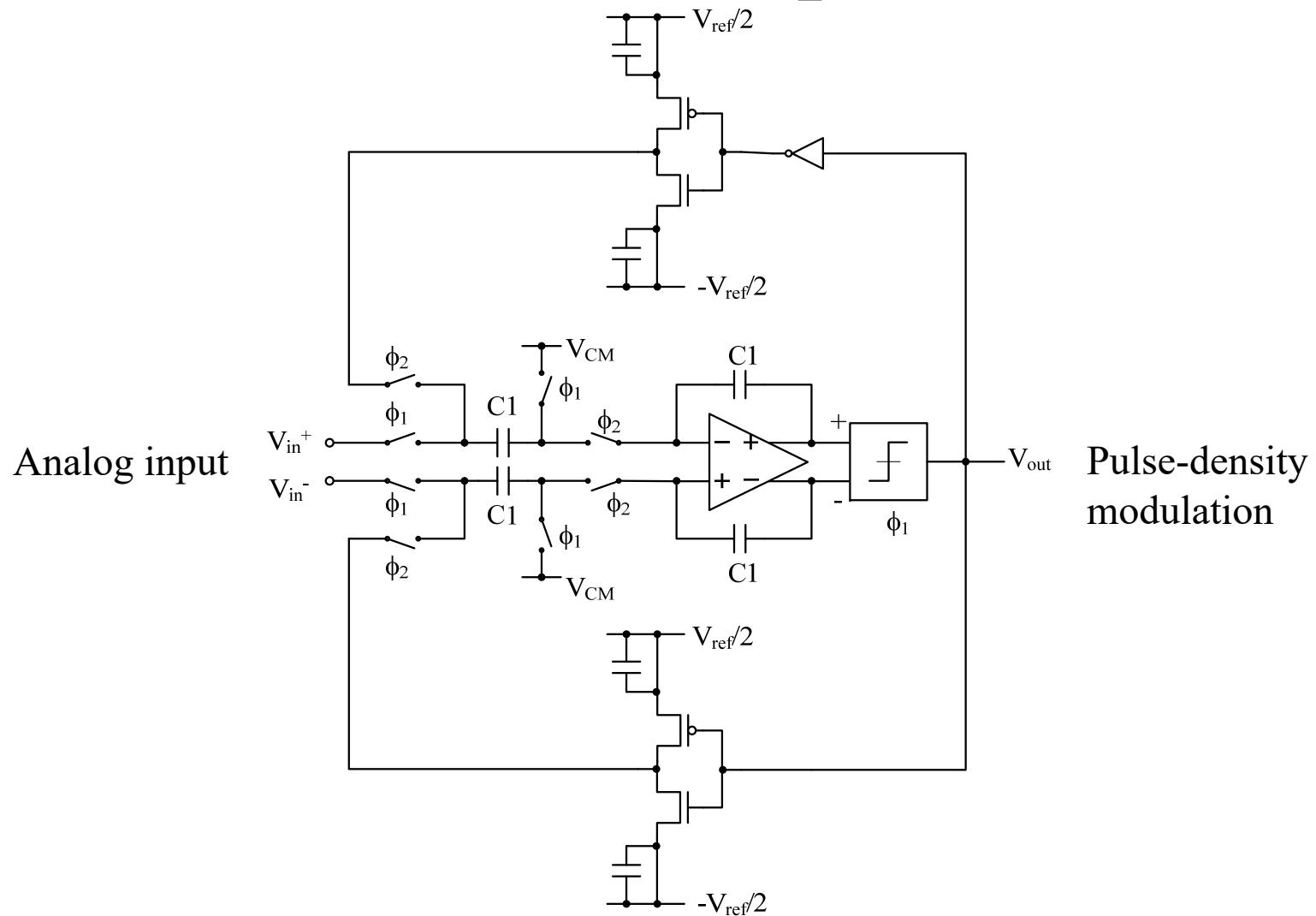


2-input DAI

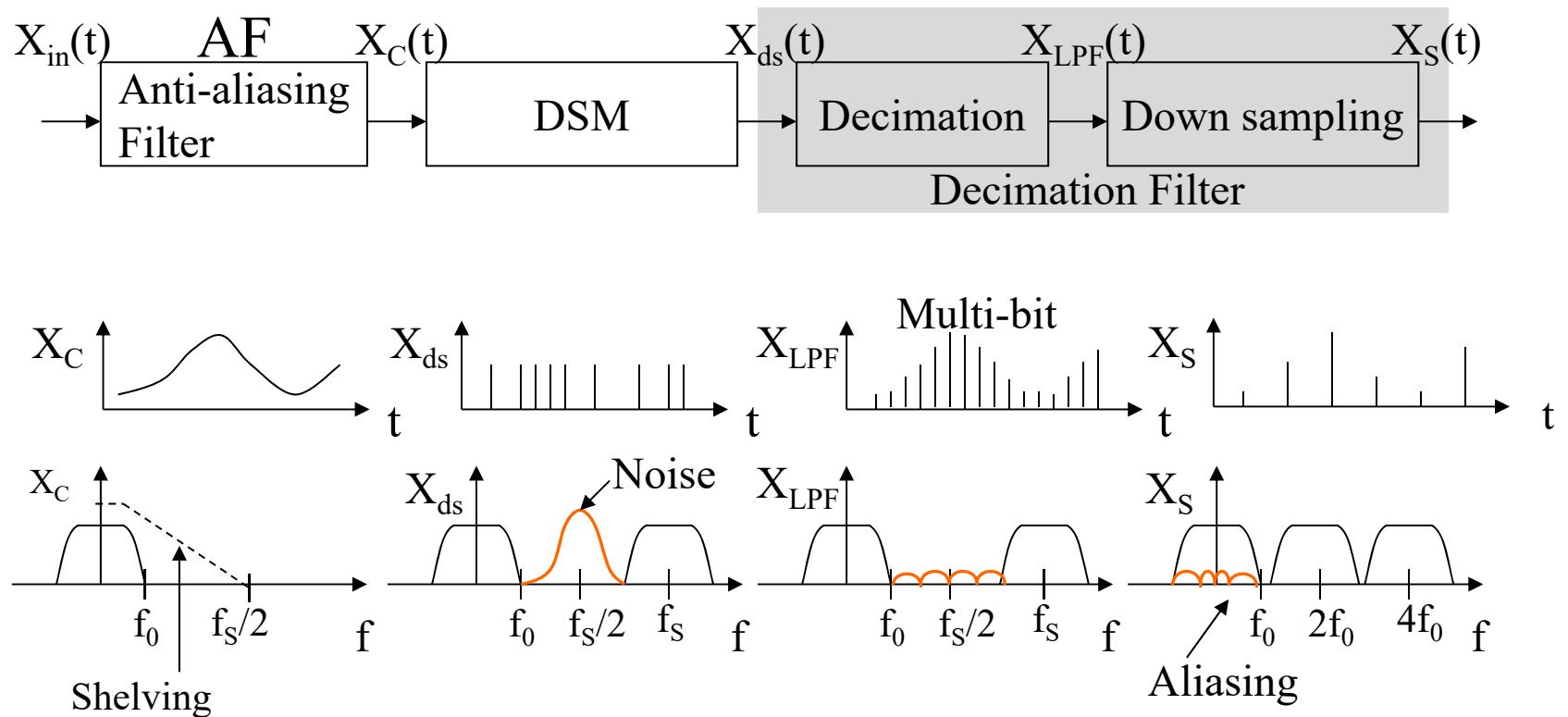
# Architecture of delta-sigma ADC



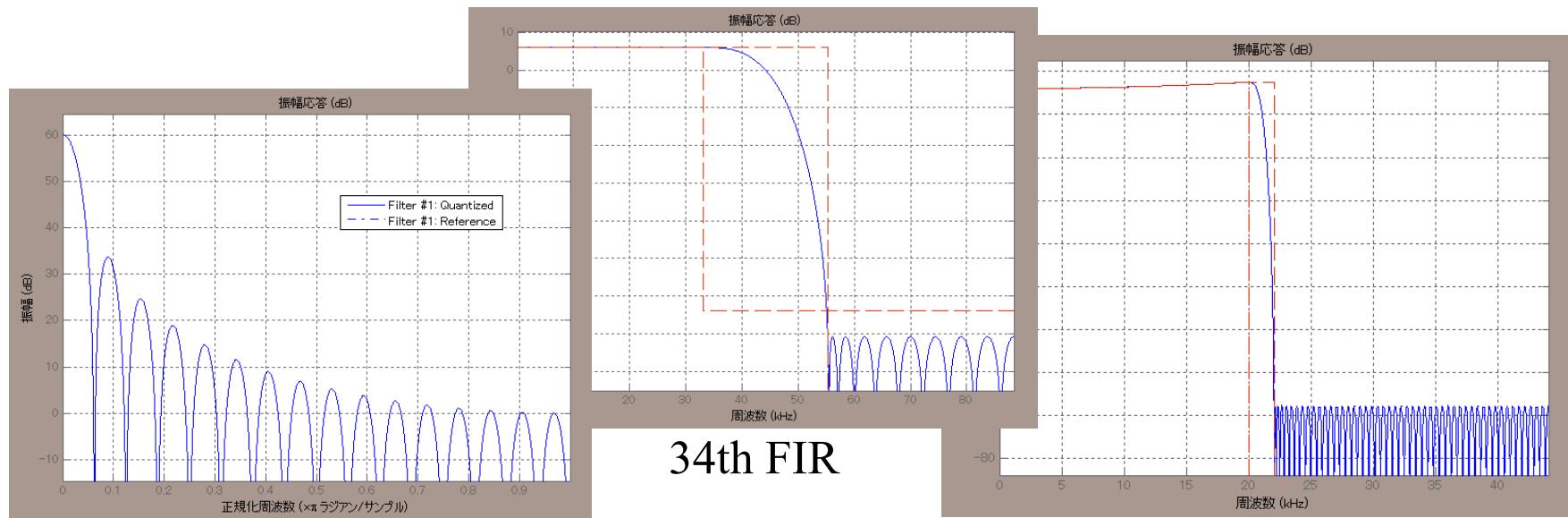
# Full differential implementation



# Conversion to binary signal



# Structure of decimation filter



SinC IIR

186th FIR

# Design of SinC<sup>L+1</sup> filter

Averaging of M data     $T_A = \frac{1}{M} \sum_{i=0}^{M-1} z^{-i} \equiv \frac{X_{out}(z)}{X_{in}(z)}$

$$MX_{out}(z) = \left\{ \sum_{i=0}^{M-1} z^{-i} \right\} X_{in}(z) = (z^{-1} + z^{-2} + \dots + z^{-M}) X_{in}(z) + (1 - z^{-M}) X_{in}(z)$$

$$= Mz^{-1} X_{out}(z) + (1 - z^{-M}) X_{in}(z)$$

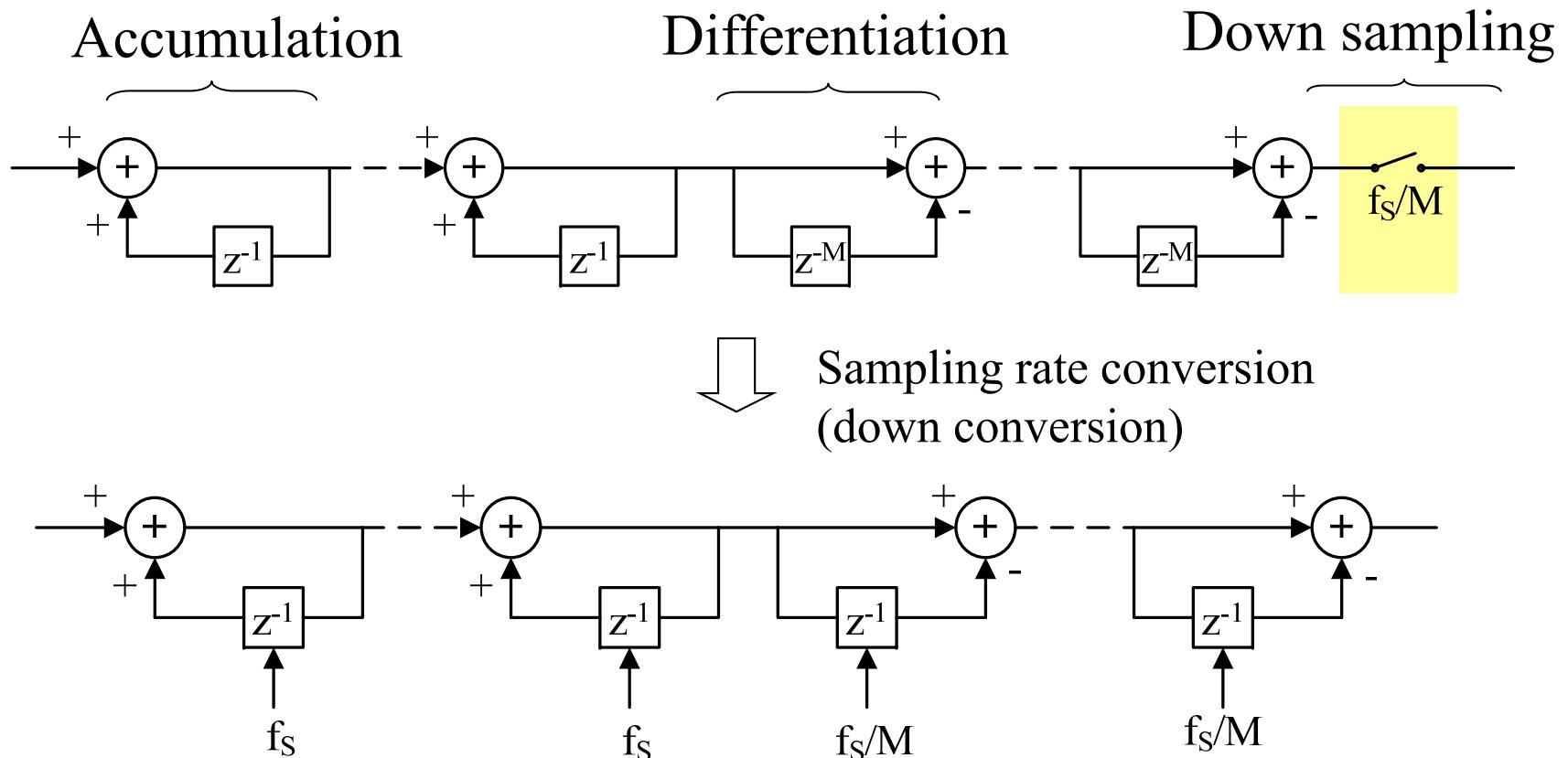
$$T_A = \frac{1}{M} \frac{1 - z^{-M}}{1 - z^{-1}}$$

SinC<sup>L+1</sup> Filter     $H_{SinC} = \frac{1}{M^{L+1}} \left[ \frac{1 - z^{-M}}{1 - z^{-1}} \right]^{L+1}$

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

Accumulation   Differentiation   Constant

# Implementation of SinC<sup>L+1</sup> Filter



SinC filter = CIC (Cascaded Integration-Comb) filter

# Order of SinCL<sup>+1</sup> Filter

Without noise shaping

$$SNR = 6.02 \cdot ENOB + 1.76$$

With 1st-order noise shaping

$$SNR = 6.02 + 1.76 - 5.17 + 30\log OSR$$

With 2nd-order noise shaping

$$SNR = 6.02 + 1.76 - 12.9 + 50\log OSR$$



With 1st-order noise shaping

$$ENOB = 1 + (-5.17 + 30\log OSR)/6.02$$

With 2nd-order noise shaping

$$ENOB = 1 + (-12.9 + 50\log OSR)/6.02$$

Number of bits of decimation filter  $N_{out} = N_{stage} \log_2(OSR)$

$N_{out}$  must be larger than ENOB.

With 1st-order noise shaping

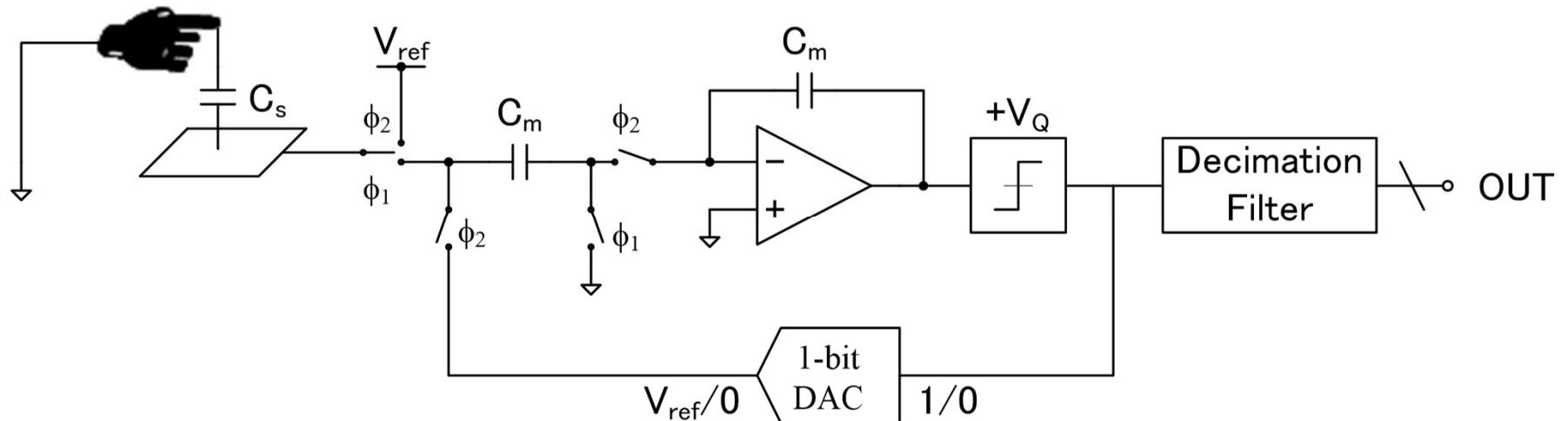
$$N_{stage} > \frac{ENOB}{\log_2(OSR)} \approx 1.5 > L$$

With 2nd-order noise shaping

$$N_{stage} > \frac{ENOB}{\log_2(OSR)} \approx 2.5 > L$$

where L is a order of the noise shaping.

# Application example of cap sense (touch panel)

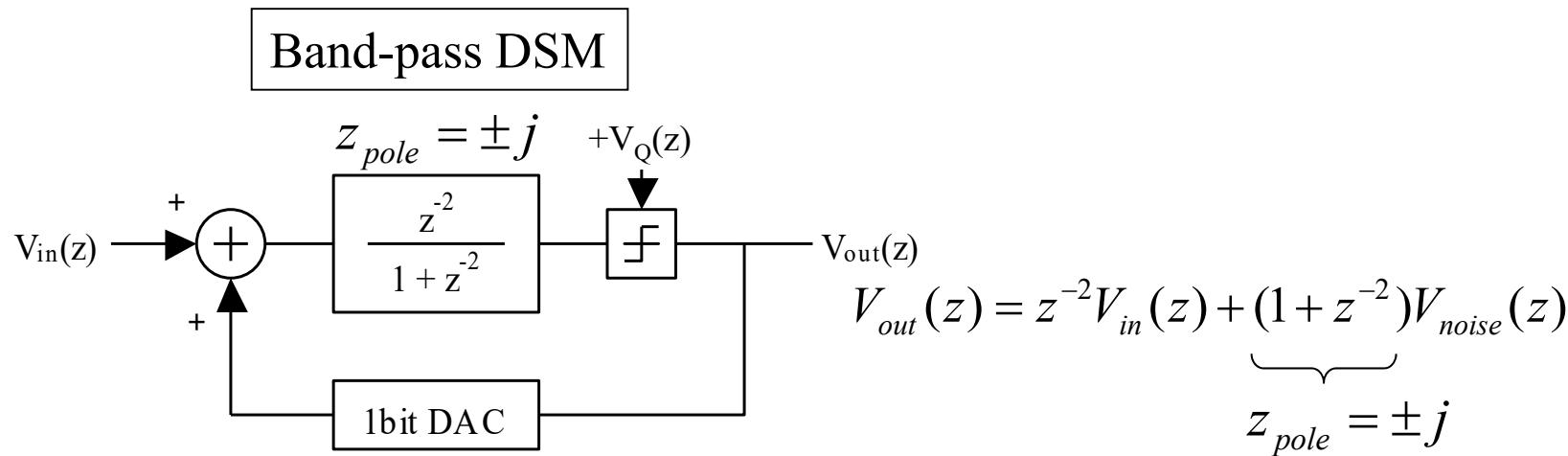
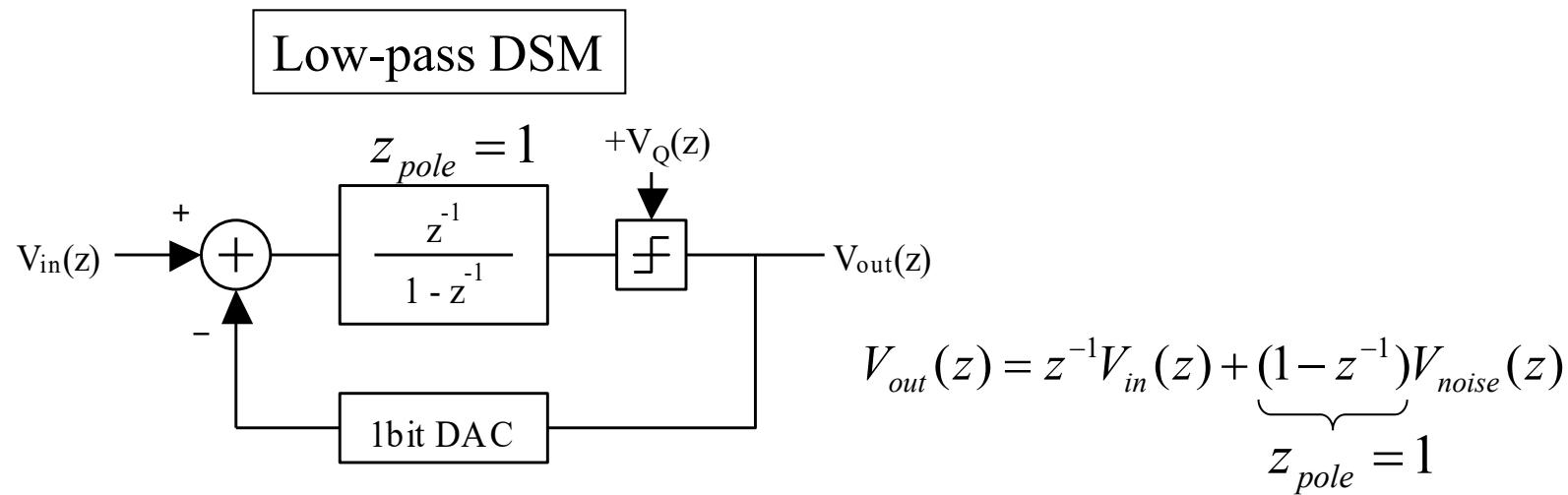


$$V_{out}(z) = z^{-\frac{3}{2}} \cdot V_{ref}(z) \frac{C_s}{C_s + C_m} + (1 - z^{-1}) \cdot V_Q(z)$$

When  $f = 0\text{Hz}$  and  $z = 1$  (DC),  $C_s = \frac{V_{out}}{V_{ref} - V_{out}} C_m$

## 3.4 Band-pass Delta-Sigma ADC

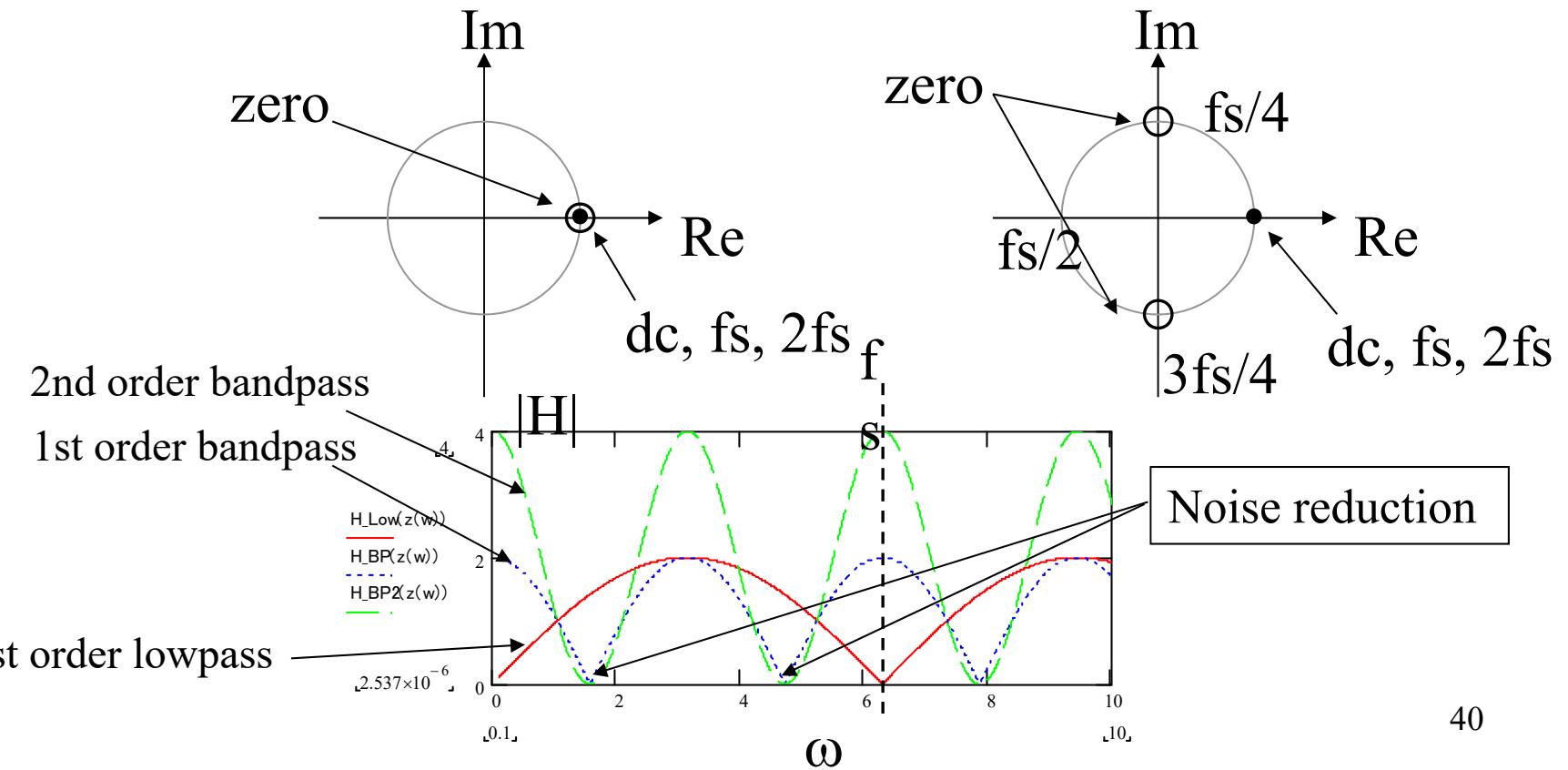
# Transfer function of band-pass DSM



# Noise shaping of band-pass DSM

Low-pass

$$V_{out}(z) = z^{-1}V_{in}(z) + (1-z^{-1})V_{noise}(z)$$

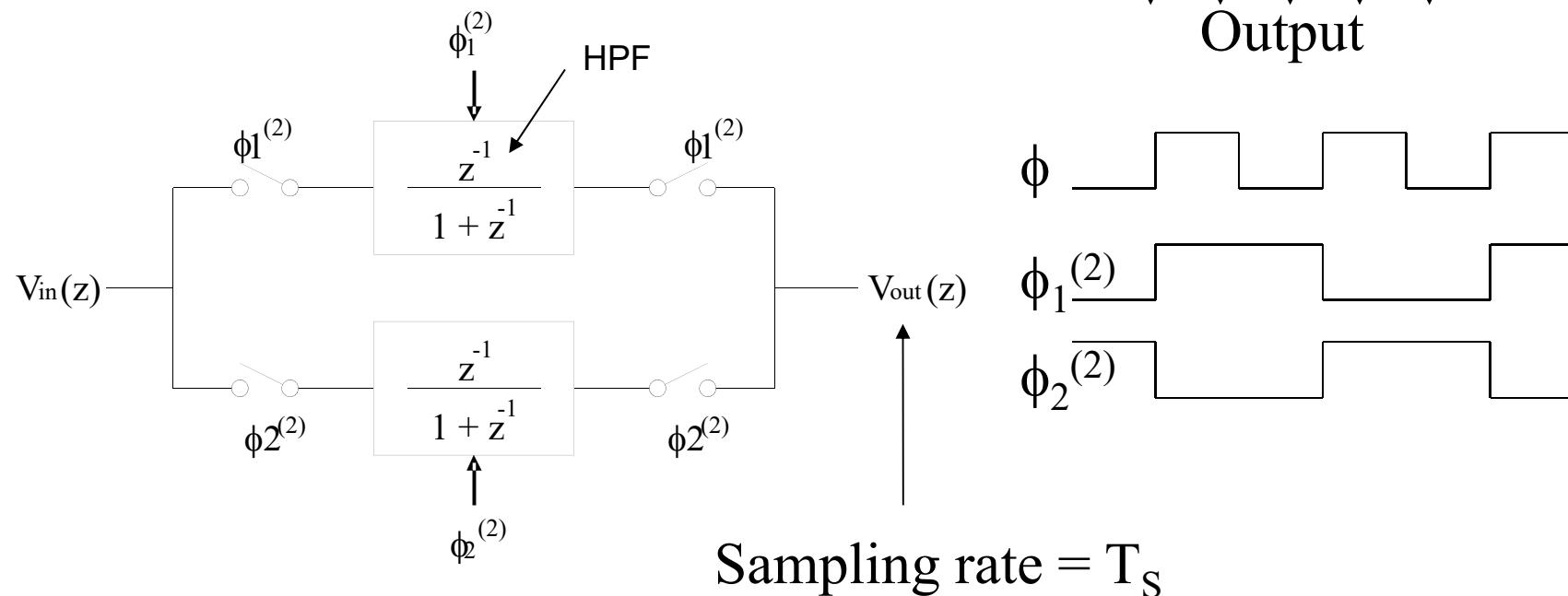
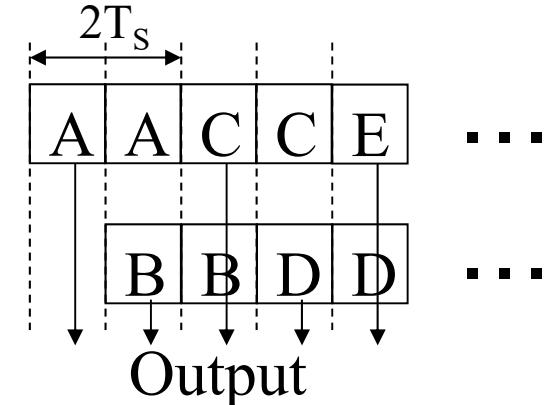


# $z^{-2}$ operation with N-Path technique

$$T_S \rightarrow 2T_S$$

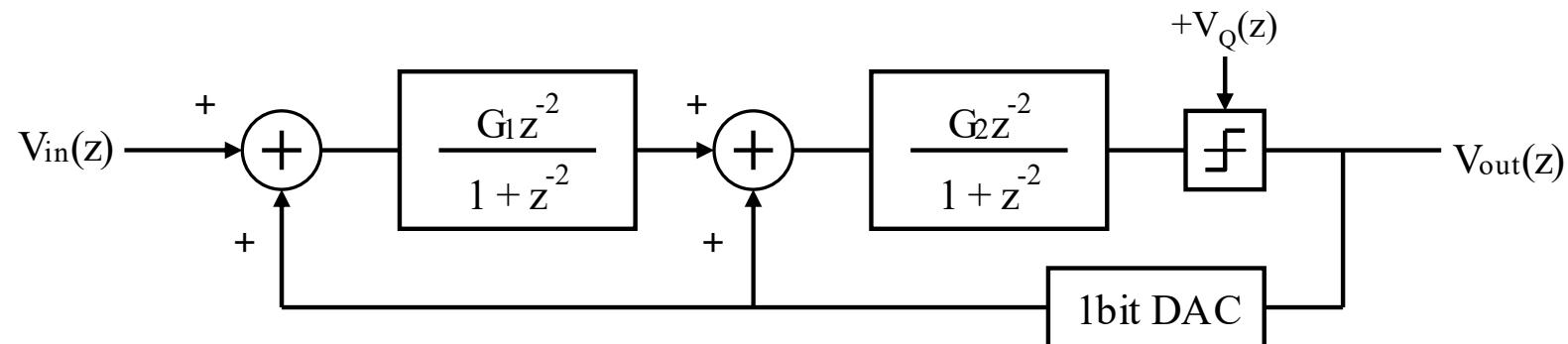
$$z^{-1} = e^{-sT_S} \quad \square \rightarrow \quad z^{-2} = e^{-s \cdot 2T_S} \quad \square \rightarrow$$

2 path technique



# 4th order band-pass DSM

High-order band-pass ADC is often used for the wireless communication systems.



$$\begin{cases} G_1 = -0.5 \\ G_2 = 2 \end{cases} \Rightarrow V_{out}(z) = -z^{-4}V_{in}(z) + \underbrace{(1 + z^{-2})^2}_{z = \pm j} V_{noise}(z)$$

# Exercise

Question:

Derive the transfer function of the block diagram shown in the previous slide.

Example solution:

$$V_{out}(z) = \frac{G_2 z^{-2}}{1 + z^{-2}} \left\{ \frac{G_1 z^{-2}}{1 + z^{-2}} (V_{in}(z) + V_{out}(z)) + V_{out}(z) \right\} + V_{noise}(z)$$

$$\left( 1 - \frac{G_1 G_2 z^{-4}}{(1 + z^{-2})^2} - \frac{G_2 z^{-2}}{1 + z^{-2}} \right) \cdot V_{out}(z) = \frac{G_1 G_2 z^{-4}}{(1 + z^{-2})^2} V_{in}(z) + V_{noise}(z)$$

$$\frac{(1 + z^{-2})^2 - G_1 G_2 z^{-4} - G_2 z^{-2} (1 + z^{-2})}{(1 + z^{-2})^2} V_{out}(z) = \frac{G_1 G_2 z^{-4}}{(1 + z^{-2})^2} V_{in}(z) + V_{noise}(z)$$

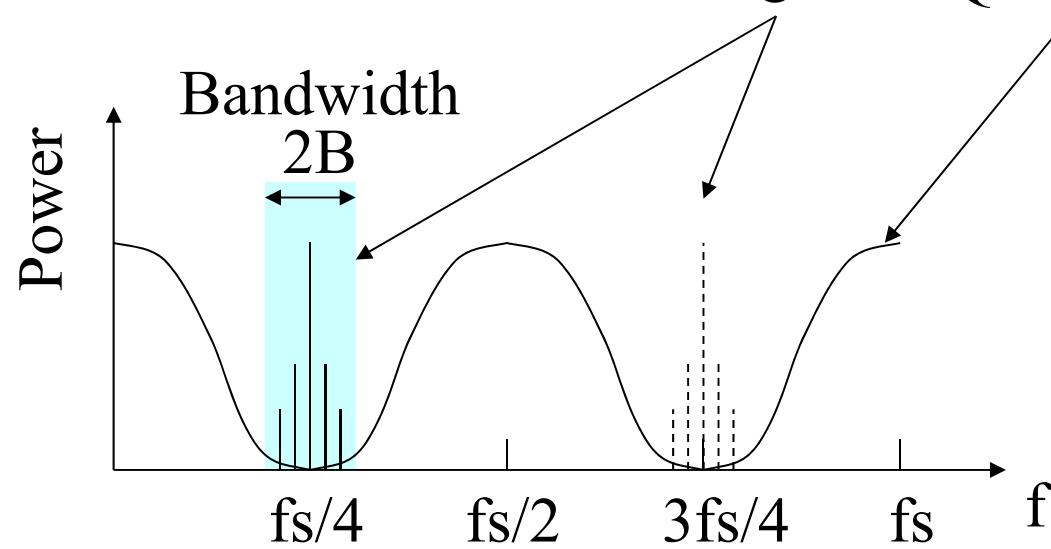
$$V_{out}(z) = \frac{G_1 G_2 z^{-4} \cdot V_{in}(z) + (1 + z^{-2})^2 \cdot V_{noise}(z)}{1 + 2z^{-2} + z^{-4} - G_1 G_2 z^{-4} - G_2 z^{-2} - G_2 z^{-4}}$$

# Oversampling rate of band-pass ADC

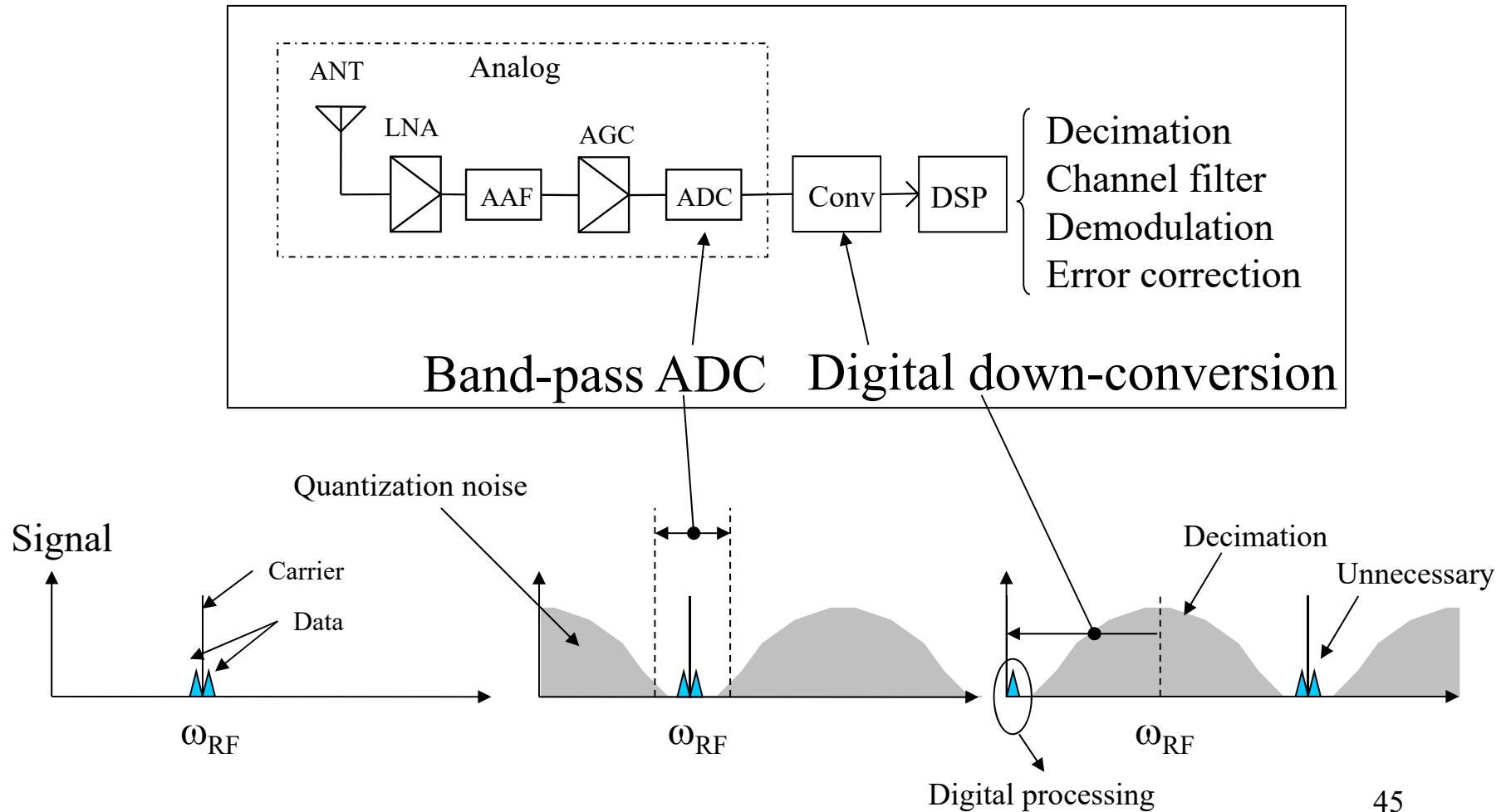
The OSR of the band-pass DSM cannot be determined by the carrier frequency, but should be calculated by the bandwidth of the converted signal.

$$OSR = \frac{f_s}{2 \cdot (2B)}$$

$$V_{out}(z) = \frac{z^{-4}V_{in}(z)}{\text{Signal}} + \frac{(1+z^{-2})^2 V_Q(z)}{\text{Quantization noise}}$$

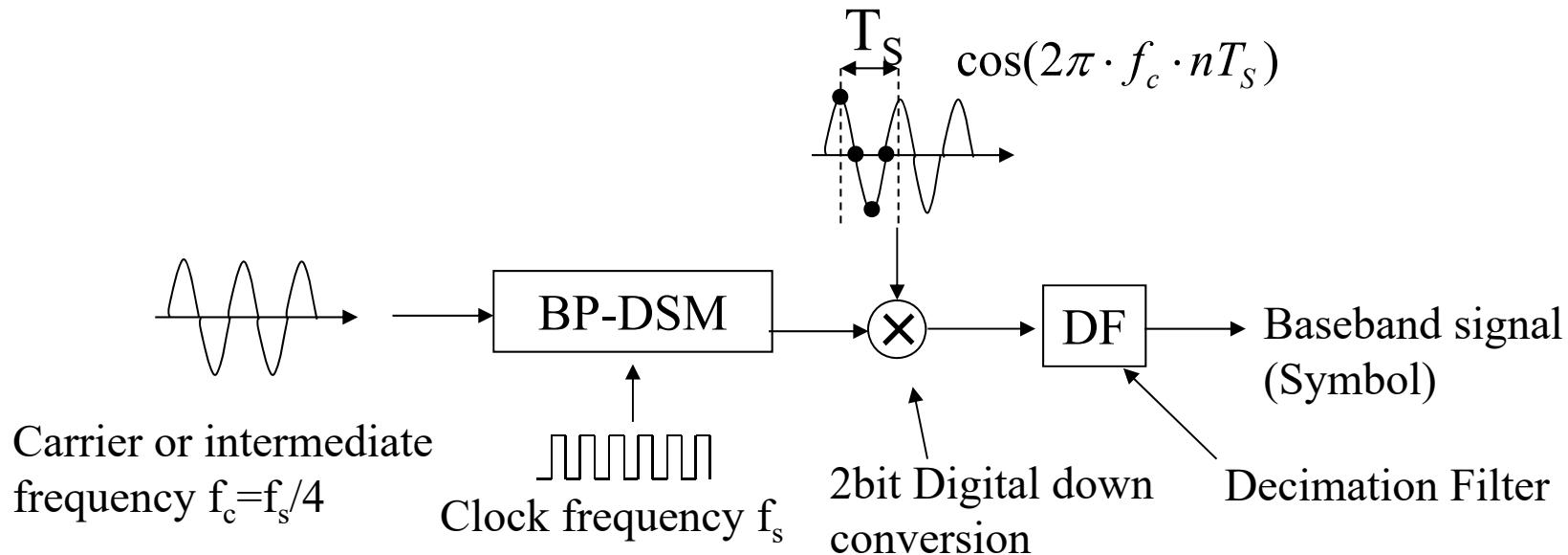


# Receiver architecture



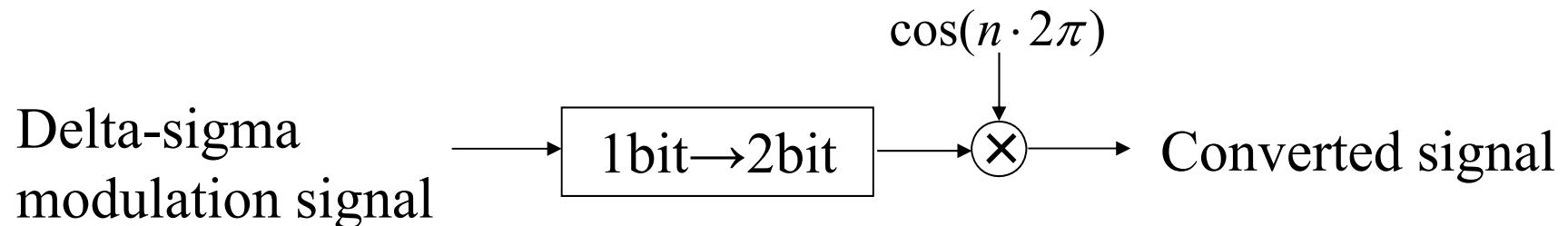
# Digital down-conversion

When  $f_s = 4f_c$ , the digital down conversion can be implemented by 2-bit logic.



$$\cos(2\pi \cdot f_c \cdot nT_s) = \cos\left(2\pi \frac{f_s}{4} n \frac{1}{f_s}\right) = \cos\left(n \frac{\pi}{2}\right) = \{1, 0, -1, 0\} \quad \text{2-bit-at-a-time}$$

# Example of 2-bit multiplier

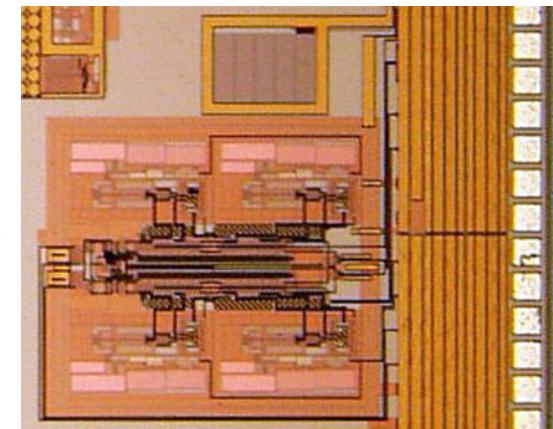


1bit→2bit extension	
a	A
0	11(-1)
1	01(1)

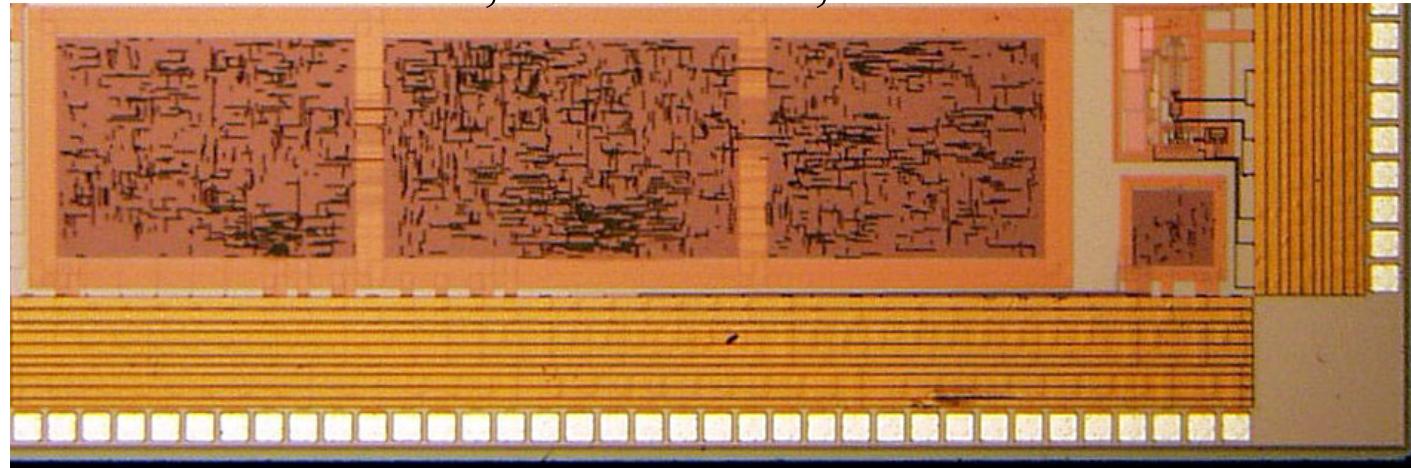
2bit multiplier		
A	B	OUT
11 (-1)	00 (0)	00 (0)
01 (1)	00 (0)	00 (0)
11 (-1)	01 (1)	11 (-1)
01 (1)	01 (1)	01 (1)
11 (-1)	11 (-1)	01 (1)
01 (1)	11 (-1)	11 (-1)
.	.	DC
.	.	DC
.	.	DC

# Example of band-pass DSM

CMOS 0.25um,  
OSR = 256,  
4th-order noise shaping,  
ENOB = 14bit



Down conversion, Decimation, and Demodulation



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