

# ADCs for High-Speed Applications

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# ADCs for High-Speed Applications

## ADC Concepts:

- quantization
- sampling

## ADC Specifications:

- static specifications
- dynamic specifications

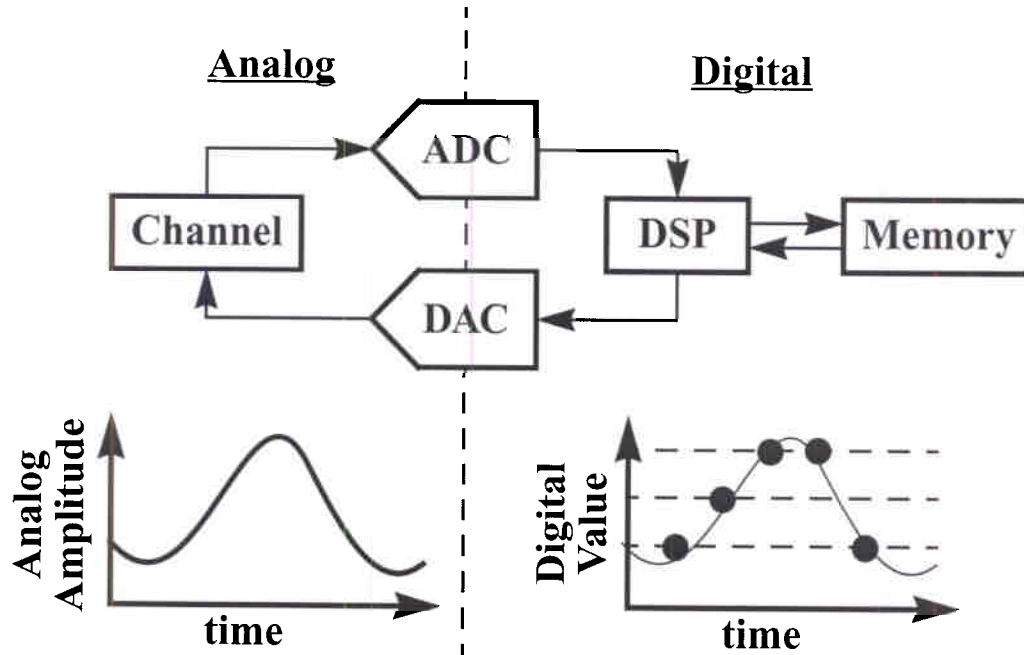
## ADC Architectures:

- classical architectures
- “new” architectures

## Summary



# ADC Concepts



- continuous time
- continuous amplitude

- discrete time
- discrete values



# Quantization and Resolution

## Quantization:

- continuous amplitudes represented with discrete levels

FS = Full Scale

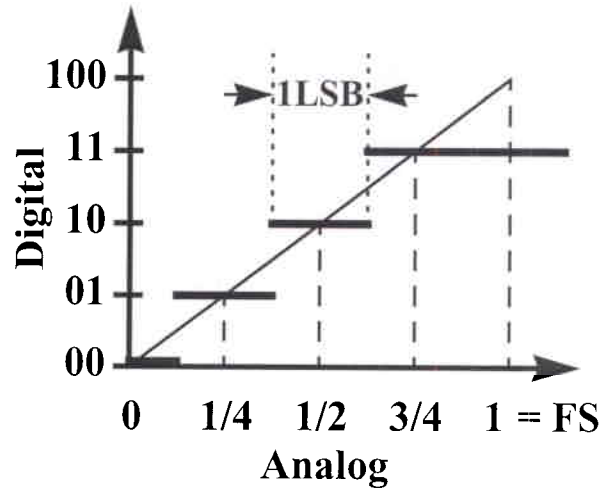
LSB = Least Significant Bit

## Resolution (1 LSB):

- smallest noticeable change

$$V_{LSB} = \frac{V_{FS}}{2^N}$$

Converter Resolution = N

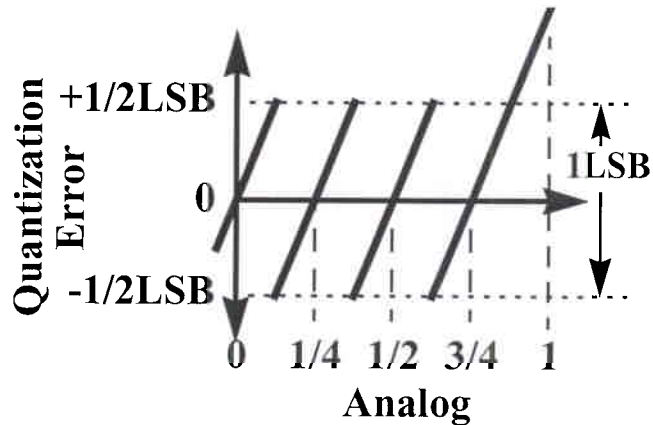


# Quantization Noise & SNR

## Quantization noise:

- quantization errors appear as noise with an RMS level of

$$\sqrt{v_{qn}^2} = \frac{V_{LSB}}{\sqrt{12}}$$



## Signal-to-Noise Ratio (SNR):

- full scale sine wave  $V_{sig} = 2^N V_{LSB} / 2\sqrt{2}$

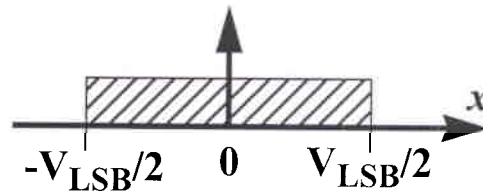
$$SNR = 20\log(V_{sig} / \sqrt{v_{qn}^2}) = [6.02N + 1.76] \text{ dB}$$

$$SNR \approx 6N \text{ dB}$$



## Notes for quantization noise:

Quantization noise can be calculated by assuming the signal within any quantization step has a uniform distribution, as shown in the sketch to the right. The error for a particular sample will simply be the distance between  $x$  and the origin where  $-V_{LSB}/2 < x < V_{LSB}/2$ . The rms error or the quantization noise can then be calculated as:

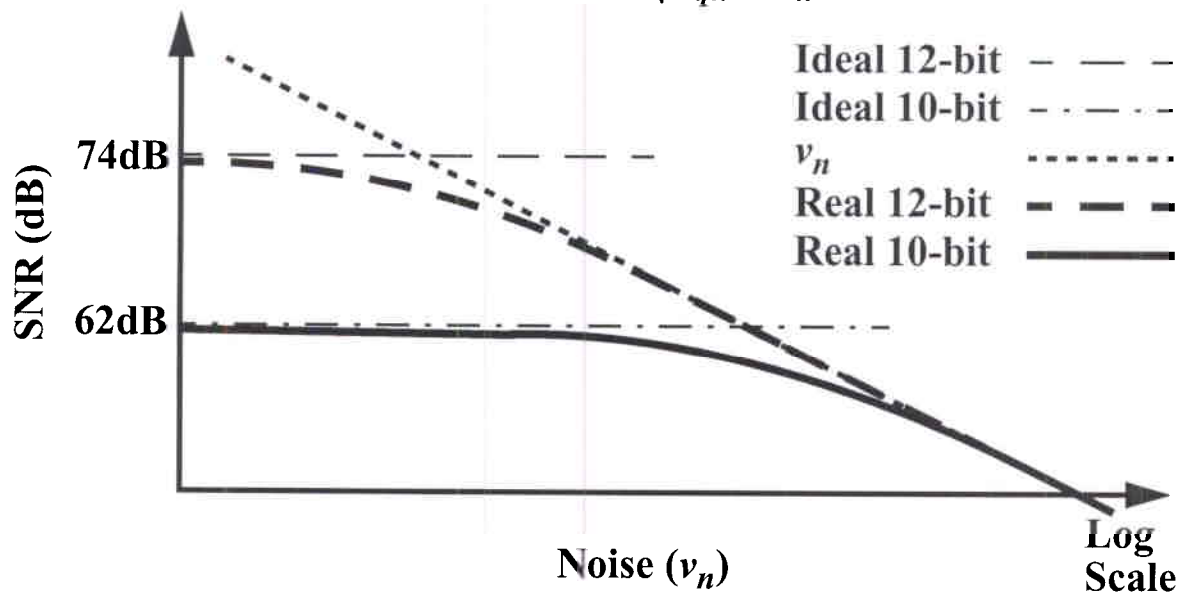


$$v_{qn} = \sqrt{\frac{1}{V_{LSB}} \int_{-V_{LSB}/2}^{V_{LSB}/2} x^2 dx} = \frac{V_{LSB}}{\sqrt{12}}$$

# SNR and Resolution: the Practical Case

Quantization is not the only noise source:

$$SNR = 20\log\left(\frac{V_{sig}}{\sqrt{v_{qn}^2 + v_n^2}}\right)$$

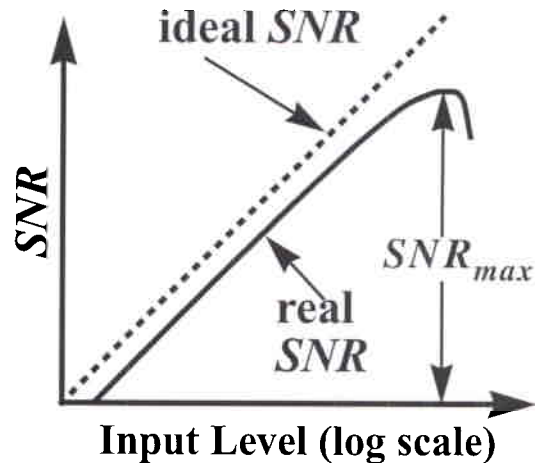


# SNR & Effective Number of Bits

## Measuring SNR:

- $f_{in} < f_s/2$
- noise bandwidth =  $f_s/2$
- SNR increases with the signal
- ideal SNR

$$SNR_{max} = (6N + 1.76) \text{ dB}$$



## Effective Number of Bits:

- ideally ENOB = N
- circuit nonlinearities and noise reduce ENOB to:

$$ENOB = \frac{SNR_{max} - 1.76}{6.02}$$



## Notes for signal-to-noise:

The maximum SNR for an ideal N-bit ADC can be calculated by assuming the input is a sine wave and the only noise source is quantization noise. The largest peak-to-peak value of the sine wave is  $2^N V_{LSB}$  which has an RMS value of  $2^N V_{LSB} / 2\sqrt{2}$ . Knowing the quantization noise is  $V_{LSB} / \sqrt{12}$ , one can express the SNR as:

$$SNR = 20 \log \left( \frac{2^N V_{LSB} / 2\sqrt{2}}{V_{LSB} / \sqrt{12}} \right)$$

which reduces to:

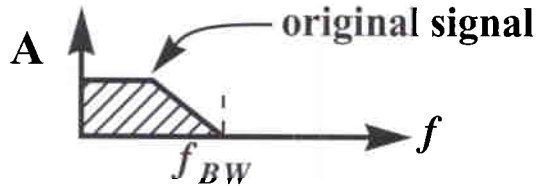
$$SNR = 6.02N + 1.76 \text{ dB}$$

Solving the above equation for N, yields the effective number of bits from the maximum SNR

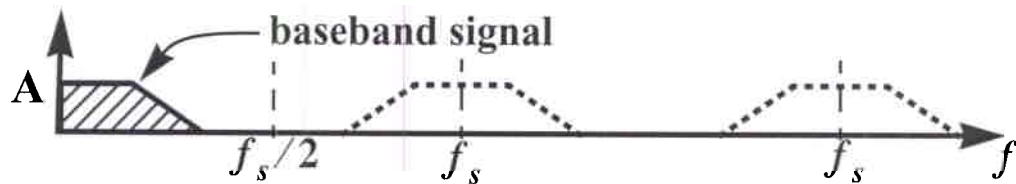
$$ENOB = (SNR_{max} - 1.76) / 6.02$$

# Sampling, Nyquist and Aliasing

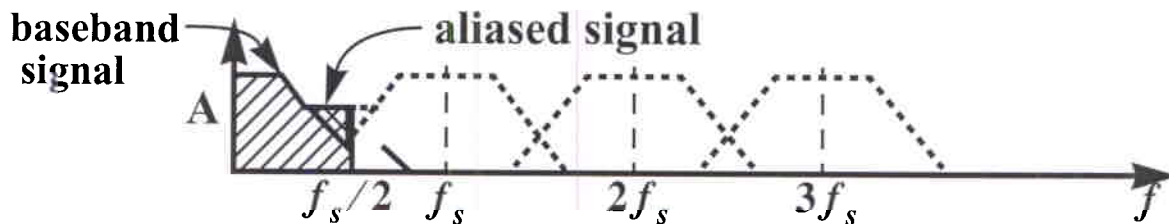
Signal spectrum:



Sampled spectrum with  $f_s > 2f_{BW}$ :

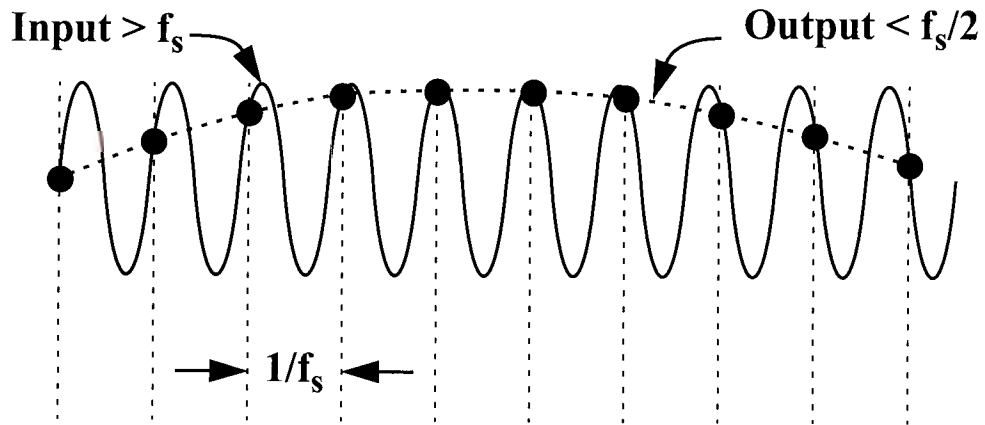


Sampled spectrum with  $f_s < 2f_{BW}$ :





## Aliasing in the Time Domain



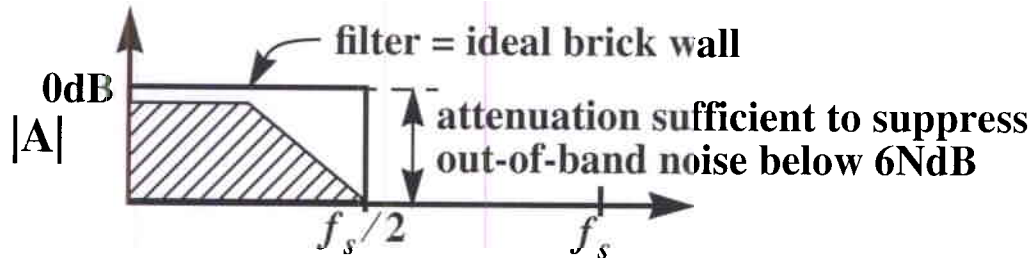
- signals beyond  $f_s / 2$  are aliased to below  $f_s/2$
- useful for down conversion from IF to baseband



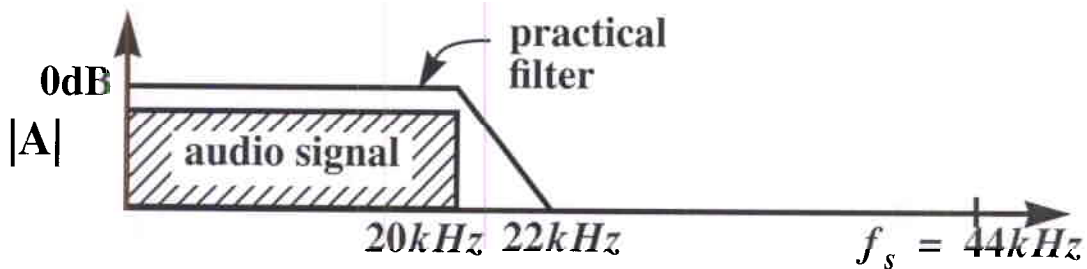
# Sampling & Anti-Aliasing Filtering (ADCs)

$$\text{Nyquist Criterion } f_s > 2f_{BW}$$

**Ideal Case:**  $f_s = 2f_{BW}$ <sup>1</sup>



**Practical Case: digital audio**

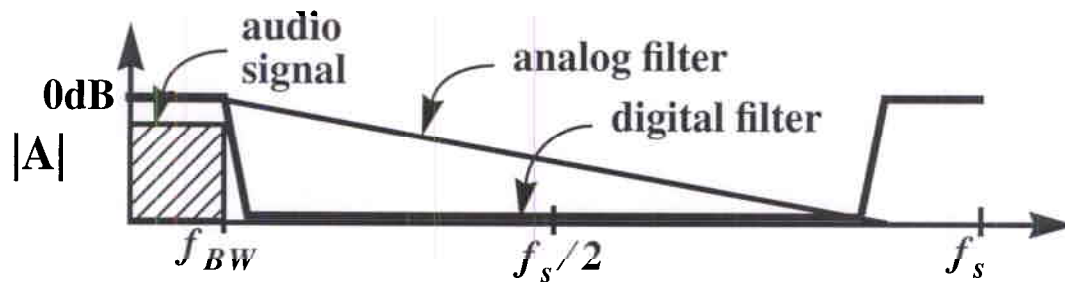


1. Strictly, there should be NO signal energy at half the sampling frequency

# Oversampling & Anti-Aliasing Filters

Oversampling eases filtering requirements:

- simple analog filter



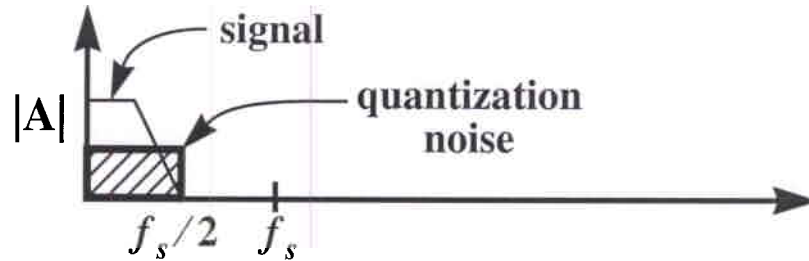
- high order digital filters can have sharp transitions & linear phase



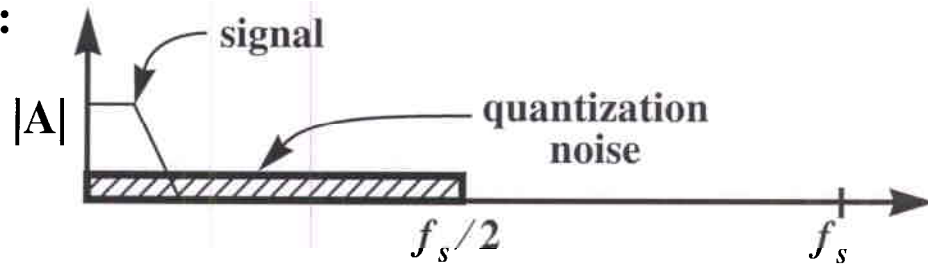
# Oversampling & Quantization Noise

Quantization noise is uniform between 0 and  $f_s/2$

$$f_s = 2f_{BW} :$$



$$f_s = 8(2f_{BW}) :$$



**In-band SNR::**

$$SNR = \left[ 6.02N + 1.76 + 10\log\left(\frac{f_s}{2f_{BW}}\right) \right] \text{dB}$$



# ADC Concepts

## Summary

### Quantization determines resolution:

- smallest discernible step = 1 LSB
- more bits = more resolution
- $SNR \approx 6N\text{dB}$

### Sampling rate determines useful bandwidth:

- $f_s > 2f_{BW}$  - Nyquist criterion
- $f_s \gg 2f_{BW}$  - eases analog filtering

$$\bullet SNR = \left[ 6N + 1.76 + 10\log\left(\frac{f_s}{2f_{BW}}\right) \right] \text{dB}$$



# Specifying ADCs

## DC Specifications:

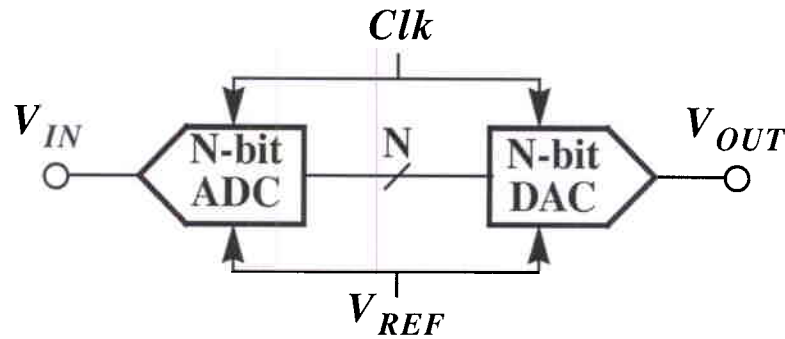
- $f_s \ll f_{clk}$
- “optimal” specifications

## Dynamic Specifications:

- $f_s \sim f_{clk}$
- indicates performance degradation for high signal frequencies
- “realistic” specifications



# DC Specifications



**Ideal output:**  $V_{OUT} = V_{REF} \left[ \frac{b_{n-1}}{2} + \dots + \frac{b_0}{2^N} \right]$

- MSB (most significant bit) =  $b_{n-1}$

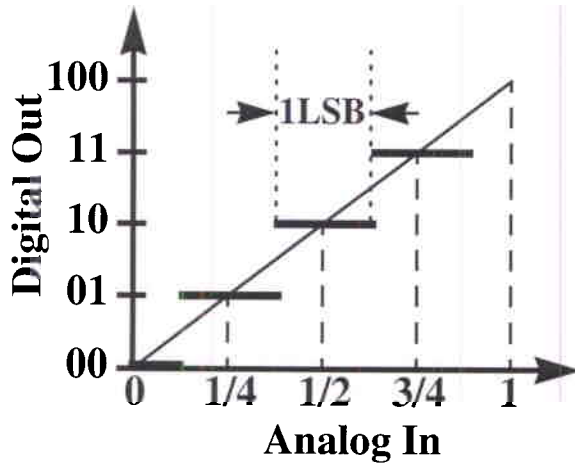
- LSB (least significant bit) =  $b_0$

- Full Scale input (or output) =  $V_{REF} \left[ \frac{2^N - 1}{2^N} \right] \approx V_{REF}$

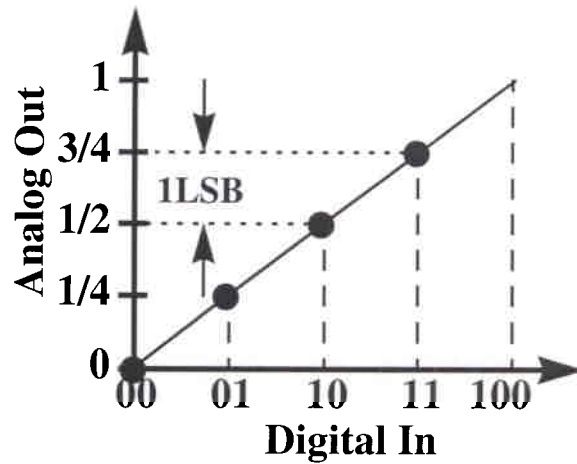


# Ideal ADCs and DACs

Analog-to-Digital Converter



Digital-to-Analog Converter



- levels are uniformly spaced:

$$1\text{LSB} = \frac{V_{REF}}{2^N}$$

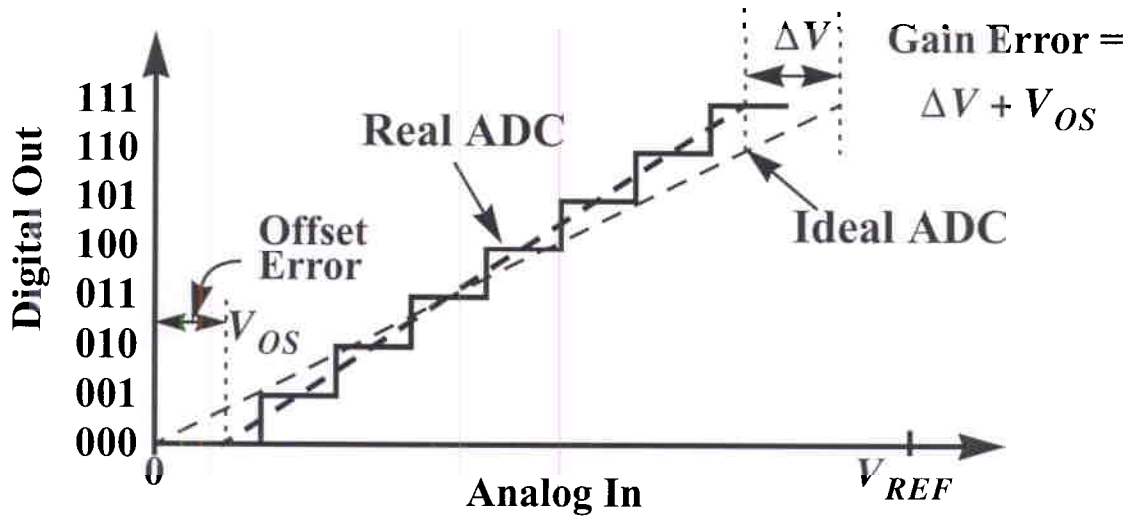
- slope =  $45^\circ$ , intercept = origin





# Gain and Offset Errors

3-bit ADC example:



**Offset Error:** difference between zero intercept and the origin  
- expressed in LSBs

**Gain Error:** difference between real and ideal slopes  
- often expressed in LSBs from ideal full scale



# Linearity Errors

## Differential Non-Linearity (DNL):

- difference between real & ideal step size

## Integral Non-Linearity (INL):

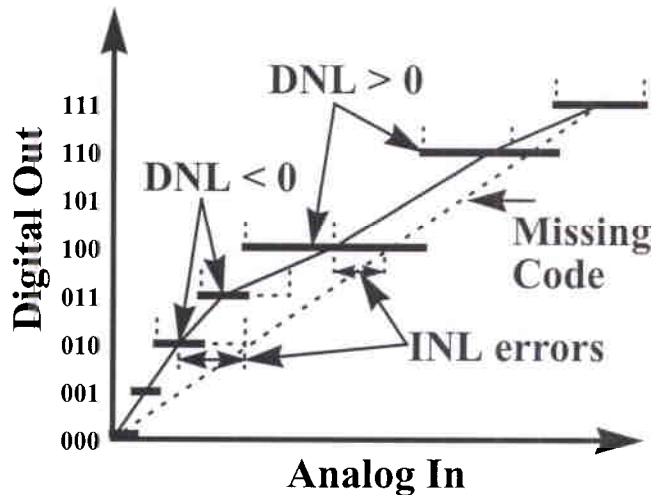
- difference between step midpoint and line joining end points

## Missing Codes:

- excessive DNL leads to missed codes in ADCs

## Non-Monotonicity:

- excessive DNL leads to non-monotonic behavior in ADCs



# Spurious Free Dynamic Range (SFDR)

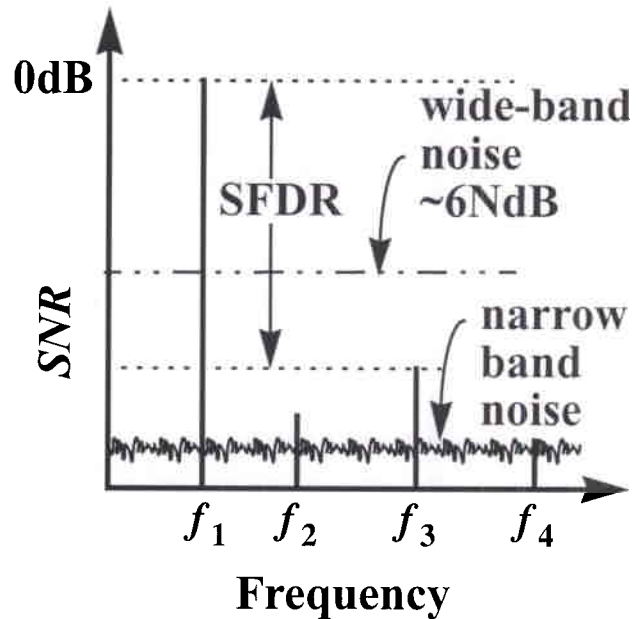
- INL, DNL and finite slew rates cause distortion

## SFDR:

- is a measure of harmonic distortion
- ideally  $SFDR > SNR$

## To Measure:

- noise bandwidth  
 $\Delta f \ll f_s/2$
- input sine wave  
 $f_{in} \ll f_s/2$

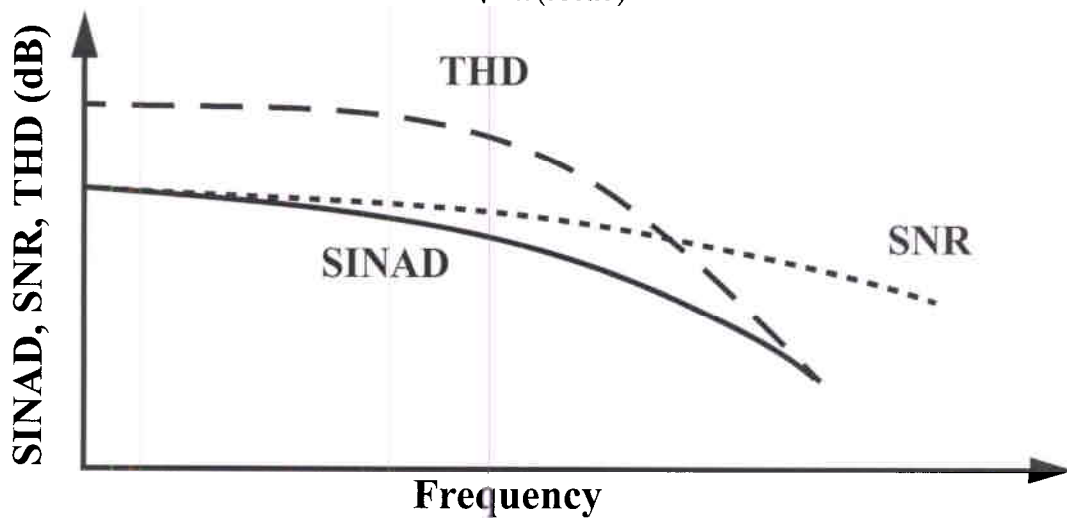


# Signal-to-Noise and Distortion (SINAD)

## SINAD:

- noise and distortion degrade an ADC's performance

$$\text{SINAD} = 20\log\left(\frac{V_{sig}}{\sqrt{v_{n(total)}^2 + THD^2}}\right)$$



# ADC Bandwidth Specifications

## Nyquist:

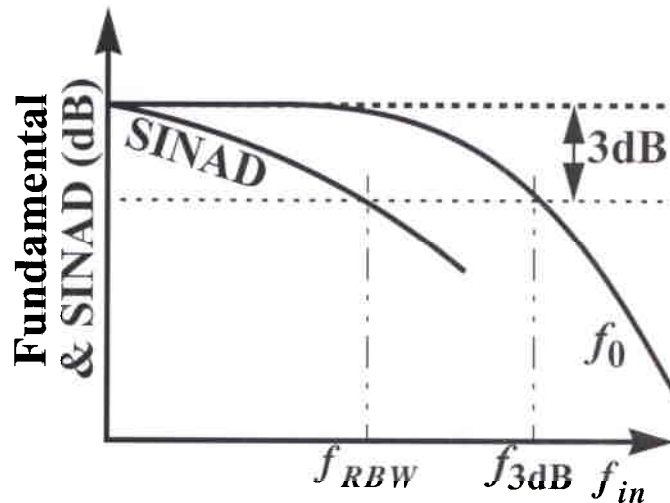
- bandwidth =  $f_s/2$

## Full Power BW ( $f_{3dB}$ ):

- fundamental down 3dB
- typically  $f_{3dB} > f_s/2$

## Resolution BW ( $f_{RBW}$ ):

- SINAD down 3dB  
(1/2 bit = 3dB)
- ideally  $f_{RBW} \gg f_s/2$



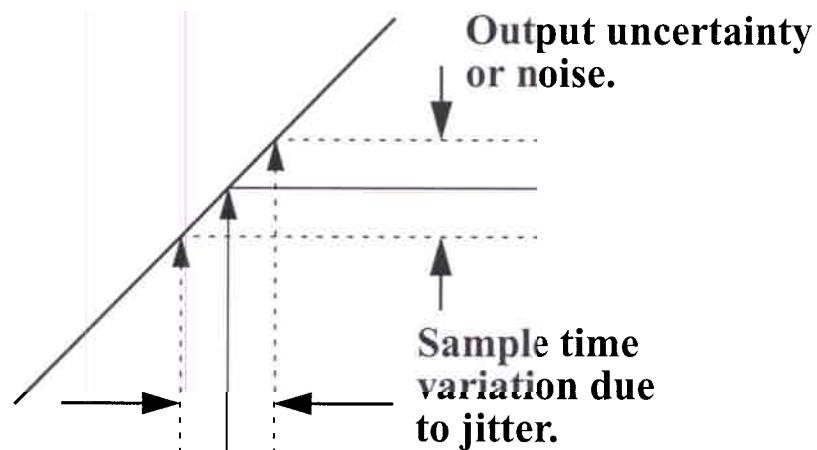
$f_{3dB}$  &  $f_{RBW}$  are usually independent of  $f_s/2$



# Sampling Errors (Jitter)

## Jitter:

- most systems assume the signal is sampled uniformly.
- clock noise leads to non-uniform sampling (*i.e.* jitter).



- leads to SNR degradation for high frequency inputs.

$$2\pi f_a T_j V_p < V_{LSB}$$



## Notes for jitter:

Jitter introduces uncertainty in the sampling instant that leads to uncertainty in the sampled value resulting in a degradation in the SNR. The effects of jitter depend on the slope of the signal at the sampling instant. A sine wave of frequency  $f_a$  with a peak value of  $V_p$  can be expressed as  $V_p \sin(2\pi f_a t)$ . The slope of this wave form at any time is given by:

$$\partial v / \partial t = 2\pi f_a V_p \cos(2\pi f_a t)$$

The worst case slope occurs at  $t = 0$  and is given by  $2\pi f_a V_p$ . If the sampling clock has an rms jitter of  $T_j$  one can substitute for  $\partial t$  to solve for the worst case  $\partial v$ :

$$\partial v = 2\pi f_a V_p T_j$$

A nominal target for  $\partial v$  is to keep it below one LSB or

$$2\pi f_a V_p T_j < V_{LSB}$$

# ADC Specification Summary

**For true N-bit performance:**

- $\text{INL} < \pm\frac{1}{2}\text{LSB}$
- $\text{DNL} < \pm\frac{1}{2}\text{LSB}$
- $\text{ENOB} \sim N$
- $\text{SFDR} > 6N\text{dB}$
- $f_{\text{RBW}} > f_s/2$

