#### EE 505

Lecture 4

#### Quantization Noise Spectral Characterization

#### **CORRECTION** from Last Lecture

## **Differential Nonlinearity (DAC)**

Nonideal DAC



Theorem: The INL<sub>k</sub> of a DAC (when corrected for gain error and offset) can be obtained from the DNL by the expression  $INL_{k} = \sum_{i=1}^{k} DNL(i)$ 

Caution: Be careful about using this theorem to measure the INL since errors in DNL measurement (or simulation) can accumulate

Corollary: The DNL of a DAC (when corrected for gain error and offset) can be expressed as

DNL(k)=INL<sub>k</sub>-INL<sub>k-1</sub>

#### Commercial Data Converter Update (Jan 25, 2023)

Analog Dev	vices		
Flash			7
MS			2
Pipe			566
SAR			863
Delta Sigr		255	
		Total	1693
Texas			
Instruments	S		
Flash			3
Two- Step			6
Folding			64
Pipe			296
SAR			428
Delta Sigr		195	
Not Specified			2
		Total	994

Note: Based upon reported part numbers. ADI lists some parts in multiple performance categories so some are listed more than once. Both have variants of one component listed with unique part numbers

**Consider ADC** 



Linearity testing often based upon code density testing

Code density testing:



Ramp or multiple ramps often used for excitation Linearity of test signal is critical (typically 3 or 4 bits more linear than DUT)



• First and last bins generally have many extra counts (and thus no useful information)

• Typically average 16 or 32 hits per code

Code density testing:

$$\overline{C} = \frac{\sum_{i=1}^{N-2} \widehat{C}_i}{N-2}$$





 $\begin{bmatrix} \underline{L} & \underline{K} = 1 \\ \overline{C} \end{bmatrix}$   $DNL = \max_{1 \le i \le N-2} \{ |DNL_i| \}$   $INL = \max_{1 \le i \le N-3} \{ |INL_i| \}$ 



- Code Density Measurements are Indirect Measurements of the INL and DNL
- Can give very wrong information under some nonmonotone missing code scenarios
- Often use an average of 16 or 32 samples per code
- Measurement noise often 1 lsb or larger but averages out
- Sometimes use good sinusoidal waveform but must correct code density for this distinction
- Full code-density testing is costly for high-resolution low-speed data converters because of data acquisition costs
- Reduced code testing using servo methods is often a less costly alternative but may miss some errors

#### Performance Characterization of Data Converters

- Static characteristics
  - Resolution
  - Y− Least Significant Bit (LSB)
  - Offset and Gain Errors
  - Absolute Accuracy
  - Relative Accuracy
  - Integral Nonlinearity (INL)
  - Differential Nonlinearity (DNL)
  - Monotonicity (DAC)
  - Missing Codes (ADC)
    - Quantization Noise
  - Low-f Spurious Free Dynamic Range (SFDR)
  - Low-f Total Harmonic Distortion (THD)
  - Effective Number of Bits (ENOB)
  - Power Dissipation

- DACs and ADCs generally quantize both amplitude and time
- If converting a continuous-time signal (ADC) or generating a desired continuoustime signal (DAC) these quantizations cause a difference in time and amplitude from the desired signal
- First a few comments about Noise

#### What is Noise in a data converter?

Noise is a term applied to some nonideal effects of a data converter

Precise definition of noise is probably not useful

Some differences in views about what nonideal characteristics of a data converter should be referred to as noise

#### Types of noise:

- Random perturbations in V or I due to movement of electrons in electronic circuits
- Interfering signals generated by other systems
- Interfering signals generated by a circuit or system itself
- Error signals associated with imperfect signal processing algorithms or circuits \_ Quantization noise
  - Sample Jitter
  - Harmonic Distortion

### Noise

All of these types of noise are present in data converters and are of concern when designing most data converters

Can not eliminate any of these noise types but with careful design can manage their effects to certain levels

Noise (in particular the random noise) is often the major factor limiting the ultimate performance potential of many if not most data converters

## Noise

#### Types of noise:

- Perturbations in V or I due to movement of electrons in electronic circuits
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- Interfering signals generated by a circuit or system itself
- Error signals associated with imperfect signal processing algorithms or circuits
  - 🔶 Quantization noise
    - Sample Jitter
    - Harmonic Distortion
      - Quantization noise is a significant
        component of this noise in ADCs and
        DACs and is present even if the ADC
        or DAC is ideal

Only the first type is associated with random variations but from a performance limitation viewpoint, all

#### Quantization Noise in ADC (same concepts apply to DACs)

Consider an Ideal ADC with first transition point at 0.5X<sub>LSB</sub>



If the input is a low frequency sawtooth waveform of period T that goes from 0 to  $X_{REF}$ , the error signal in the time domain will be:



This time-domain waveform (after dc offset is removed) is termed the Quantization Noise for the ADC with a sawtooth (or triangular) input



For large n, this periodic waveform "behaves" much like a random noise source that is uncorrelated with the input and can be characterized by its RMS value which can be obtained by integrating over any interval of length  $T_{1.}$  For notational convenience, shift the waveform to the left by  $T_{1/2}$  units

$$\mathsf{E}_{\mathsf{RMS}} = \sqrt{\frac{1}{\mathsf{T}_{1}} \int_{-\mathsf{T}_{1}/2}^{\mathsf{T}_{1}/2} \varepsilon_{Q}^{2}(t) dt}$$



In this interval,  $\epsilon_{\text{Q}}$  can be expressed as

$$\varepsilon_{\mathcal{Q}}(t) = -\left(\frac{\mathsf{X}_{\mathsf{LSB}}}{\mathsf{T}_{\mathsf{1}}}\right)\mathsf{t}$$

$$\mathsf{E}_{\mathsf{RMS}} = \sqrt{\frac{1}{\mathsf{T}_{1}} \int_{-\mathsf{T}_{1}/2}^{\mathsf{T}_{1}/2} \varepsilon_{Q}^{2}(t) dt}$$

$$E_{RMS} = \sqrt{\frac{1}{T_1} \int_{-T_1/2}^{T_1/2} \left(-\frac{\mathcal{X}_{LSB}}{T_1}\right)^2 t^2 dt}$$

$$E_{RMS} = \mathcal{X}_{LSB} \sqrt{\frac{1}{T_1^3} \left. \frac{t^3}{3} \right|_{-T_1/2}^{T_1/2}}$$

$$\mathsf{E}_{\mathsf{RMS}} = \frac{\mathcal{X}_{\mathsf{LSB}}}{\sqrt{12}}$$



$$E_{RMS} = \frac{\mathcal{X}_{LSB}}{\sqrt{12}}$$

The signal to quantization noise ratio (SNR) can now be determined. Since the input signal is a sawtooth waveform of period T and amplitude  $X_{REF}$ , it follows by the same analysis that it has an RMS value of

$$\mathcal{X}_{\text{RMS}} = \frac{\mathcal{X}_{\text{REF}}}{\sqrt{12}}$$

Thus the SNR is given by

$$SNR = \frac{\mathcal{X}_{RMS}}{E_{RMS}} = \frac{\mathcal{X}_{RMS}}{\mathcal{X}_{LSB}} = 2^{n}$$

or, in dB,

#### $SNR_{dB} = 20(n \cdot \log 2) = 6.02n$

Note: dB subscript often neglected when not concerned about confusion

How does the SNR change if the input is a sinusoid that goes from 0 to  $\mathcal{X}_{\text{REF}}$  centered at  $\mathcal{X}_{\text{REF}}/2?$ 



 $SNR = 20(n \cdot \log 2) = 6.02n$ 

How does the SNR change if the input is a sinusoid that goes from 0 to  $\mathcal{X}_{\text{REF}}$  centered at  $\mathcal{X}_{\text{REF}}/2?$ 



Time and amplitude quantization points



Time and Amplitude Quantized Waveform



How does the SNR change if the input is a sinusoid that goes from 0 to  $\mathscr{X}_{\text{REF}}$  centered at  $\mathscr{X}_{\text{REF}}/2?$ 



- Appears to be highly uncorrelated with input even though deterministic
- Mathematical expression for  $\epsilon_Q$  very messy
- Excursions exceed  $X_{LSB}$  (but will be smaller and bounded by ±  $X_{LSB}$ /2 for lower frequency signal/frequency clock ratios)
- For lower frequency inputs and higher resolution, at any time, errors are approximately uniformly distributed between  $-X_{LSB}/2$  and  $X_{LSB}/2$
- Analytical form for  $\epsilon_{QRMS}$  essentially impossible to obtain from  $\epsilon_Q(t)$

How does the SNR change if the input is a sinusoid that goes from 0 to  $\mathcal{X}_{\text{REF}}$  centered at  $\mathcal{X}_{\text{REF}}/2?$ 



For low  $f_{SIG}/f_{CL}$  ratios, bounded by ±XLB and at any point in time, behaves almost as if a uniformly distributed random variable

$$\varepsilon_Q \sim U[-0.5X_{LSB}, 0.5X_{LSB}]$$

Recall:

If the random variable f is uniformly distributed in the interval [A,B] f: U[A,B] then the mean and standard deviation of f are given by  $\mu_f = \frac{A+B}{2} \qquad \sigma_f = \frac{B-A}{\sqrt{12}}$ 

Theorem: If n(t) is a random process, then for large T,  $V_{RMS} = \sqrt{\frac{1}{T} \int_{t_1}^{t_1+T} n^2(t) dt} = \sqrt{\sigma_n^2 + \mu_n^2}$ 

How does the SNR change if the input is a sinusoid that goes from 0 to  $\mathcal{X}_{\text{REF}}$  centered at  $\mathcal{X}_{\text{REF}}/2?$ 



Note this is the same RMS noise that was present with a triangular input

How does the SNR change if the input is a sinusoid that goes from 0 to  $\mathcal{X}_{\text{REF}}$  centered at  $\mathcal{X}_{\text{REF}}/2?$ 



#### **ENOB** based upon Quantization Noise

SNR = 6.02 n + 1.76

Solving for n, obtain

$$\mathsf{ENOB} = \frac{\mathsf{SNR}_{\mathsf{dB}} - 1.76}{6.02}$$

Note: could have used the  $SNR_{dB}$  for a triangle input and would have obtained the expression

$$\mathsf{ENOB} = \frac{\mathsf{SNR}_{\mathsf{dB}}}{6.02}$$

But the earlier expression is more widely used when specifying the ENOB based upon the noise level present in a data converter

#### **ENOB** based upon Quantization Noise

For very low resolution levels, the assumption that the quantization noise is uncorrelated with the signal is not valid and the ENOB expression will cause a modest error  $\sqrt{1 + \sqrt{2}}$ 

from van de Plassche (p13)

$$SNR_{corr} \cong \left(2^{n}-2+\frac{4}{\pi}\right)\sqrt{\frac{3}{2}}$$

Res (n)	SNR <sub>corr</sub>	SNR	
1	3.86	7.78	
2	12.06	13.8	
3	19.0	19.82	SNR = 6.02 n +1.76
4	25.44	25.84	
5	31.66	31.86	
6	37.79	37.88	
8	49.90	49.92	
10	61.95	61.96	

Table values in dB

Almost no difference for  $n \ge 3$ 

Effects of quantization noise can be very significant, even at high resolution, when signals are not of maximum magnitude



Quantization noise remains constant but signal level is reduced

The desire to use a data converter at a small fraction of full range is one of the major reasons high resolution is required in many applications

Effects of quantization noise can be very significant, even at high resolution, when signals are not of maximum magnitude



Example: If a 14-bit audio output is derived from a DAC designed for providing an output of 100W but the normal listening level is at 50mW, what is the SNR due to quantization noise at maximum output and at the normal listening level? What is the ENOB of the audio system when operating at 50mW?

At 100W output, SNR=6.02n+1.76 = 90.6dB

$$\frac{V^2}{R_L}$$
=100W  $\frac{V_1^2}{R_L}$ =50mW  $V_1 = \frac{V}{44.7}$ 

 $20 \log_{10}V_1 = 20 \log_{10}V - 20 \log_{10}44.7 = 20 \log_{10}V - 33 dB$ 

At 50mW output, SNR reduced by 33dB to 57.6dB

ENOB = 
$$\frac{\text{SNR}_{\text{dB}} - 1.76}{6.02} = \frac{57.6 - 1.76}{6.02} = -9.3$$

Note the dramatic reduction in the effective resolution of the DAC when operated at only a small fraction of full-scale.

## **ENOB** Summary

Resolution:

$$\mathsf{ENOB} = \frac{\mathsf{log}_{10}\mathsf{N}_{\mathsf{ACT}}}{\mathsf{log}_{10}2} = \mathsf{log}_2\mathsf{N}_{\mathsf{ACT}}$$

INL:

ENOB = 
$$n_R - \log_2(v) - 1$$
  $n_R$  specified res, v INL in LSB

$$ENOB = -log_2(INL_{REF}) - 1$$

 $INL_{REF}$  INL rel to  $X_{REF}$ 

DNL:

HW problem

Quantization noise:

rel to triangle/sawtooth



Additional ENOB will be introduced when discussing dynamic characteristics

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#### **Absolute Accuracy**

Absolute Accuracy is the difference between the actual output and the ideal or desired output of a data converter

The ideal or desired output is in reference to an absolute standard (often maintained by the National Institute of Standards and Technology – NIST) (renamed from National Bureau of Standards in 1988) and could be volts, amps, time, weight, distance, or one of a large number of other physical quantities)

Absolute accuracy provides no tolerance to offset errors, gain errors, nonlinearity errors, quantization errors, frequency rolloff, or noise

In many applications, absolute accuracy is not of a major concern

Absolute accuracy generally dominated by the nonidealities of the reference (a data converter is a ratio-metric device so no fundamental limit on ratio portion)

but ... scales, meters, etc. may be more concerned about absolute accuracy than any other parameter

#### **Relative Accuracy**

In the context of data converters, pseudo-static Relative Accuracy is the difference between the actual output and an appropriate fit-line to overall output of the data converter

INL is often used as a measure of the relative accuracy

In many, if not most, applications, relative accuracy is of much more concern than absolute accuracy

Some architectures with good relative accuracy will have very small deviations in the outputs for closely-spaced inputs whereas others may have relatively large deviations in outputs for closely-spaced inputs

DNL provides some measure of how outputs for closely-spaced inputs compare



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- INL is a key parameter that is attempting to characterize the overall linearity of a DAC !
- INL is a key parameter that is attempting to characterize the overall linearity of an ADC !
- DNL is a key parameter that is attempts to characterize the local linearity of a DAC !
- DNL is a key parameter that is attempts to characterize the local linearity of an ADC !

Are INL and DNL effective at characterizing the linearity of a data converter?

Consider the following 4 transfer characteristics, all of which have the same INL







Although same INL, dramatic difference in performance particularly when inputs are sinusoidal-type excitations

INL also gives little indication of how performance degrades at higher frequencies Spectral Analysis often used as an alternative (and often more useful in many applications) linearity measure for data converters

## Linearity Issues

- INL is often not adequate for predicting the linearity performance of a data converter
- Distortion (or lack thereof) is of major concern in many applications
- Distortion is generally characterized in terms of the harmonics that may appear in a waveform

Spectral Analysis often used as an alternative (and often more useful in many applications) linearity measure for data converters Performance Characterization of Data Converters

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Spectral Characterization



If f(t) is periodic

$$f(t) = A_0 + \sum_{k=1}^{\infty} A_k \sin(k\omega t + \theta_k)$$

alternately

$$f(t) = A_0 + \sum_{k=1}^{\infty} a_k \sin(k\omega t) + \sum_{k=1}^{\infty} b_k \cos(k\omega t) \qquad \omega = \frac{2\pi}{T}$$
$$A_k = \sqrt{a_k^2 + b_k^2}$$

Termed the Fourier Series Representation of f(t)

### **Spectral Analysis**



Often the system of interest is ideally linear but practically it is weakly nonlinear.

Often the input is nearly periodic and often sinusoidal and in latter case desired output is also sinusoidal

Weak nonlinearity will cause harmonic distortion (often just termed distortion) of signal as it is propagated through the system

Spectral analysis often used to characterize effects of the weak nonlinearity

### **Spectral Analysis**



**Distortion Types:** 

**Frequency Distortion** 

Nonlinear Distortion (alt. harmonic distortion)

Frequency Distortion: Amplitude and phase of system is altered but output is linearly related to input

Nonlinear Distortion: System is not linear, frequency components usually appear in the output that are not present in the input

Spectral Analysis is the characterization of a system with a periodic input with the Fourier series relationships between the input and output waveforms

### **Spectral Analysis**



$$X_{OUT}(t) = A_0 + \sum_{k=1}^{\infty} A_k \sin(k\omega t + \theta_k)$$

All spectral performance metrics depend upon the sequences  $\langle A_k \rangle_{k=0}^{\infty} \langle \theta_k \rangle_{k=1}^{\infty}$  (index sequence, not time sequence)

Typical spectral performance metrics of interest: SNDR, SDR, THD, SFDR, IMOD

Alternately

$$X_{OUT}(t) = A_0 + \sum_{k=1}^{\infty} a_k \sin(k\omega t) + \sum_{k=1}^{\infty} b_k \cos(k\omega t) \qquad A_k = \sqrt{a_k^2 + b_k^2} \qquad \theta_k = \tan^{-1}\left(\frac{b_k}{a_k}\right)$$



- Often termed the DFT coefficients (will show later)
- Spectral lines, not a continuous function

 $A_1$  is termed the fundamental (when input is sinusoid or periodic)  $A_k$  is termed the kth harmonic (when input is sinusoid or periodic)



Often <u>ideal</u> response will have only fundamental present and all remaining spectral terms will vanish



For a low distortion signal, the 2<sup>nd</sup> and higher harmonics are generally much smaller than the fundamental

The magnitude of the harmonics generally decrease rapidly with k for low distortion signals



f(t) is band-limited to frequency  $2\pi$  f k<sub>x</sub> if A<sub>k</sub>=0 for all k>k<sub>x</sub>

where  $\langle A_k \rangle_{k=0}^{\infty}$  are the Fourier series coefficients of f(t)

## **Distortion Analysis**

Total Harmonic Distortion, THD

 $THD = \frac{RMS \text{ voltage in harmonics}}{RMS \text{ voltage of fundamenta l}}$ 

THD = 
$$\frac{\sqrt{\left(\frac{A_2}{\sqrt{2}}\right)^2 + \left(\frac{A_3}{\sqrt{2}}\right)^2 + \left(\frac{A_4}{\sqrt{2}}\right)^2 + \dots}}{\frac{A_1}{\sqrt{2}}}$$
$$\frac{\frac{A_1}{\sqrt{2}}}{\sqrt{\sum_{k=2}^{\infty} A_k^2}}$$
$$THD = \frac{\sqrt{\sum_{k=2}^{\infty} A_k^2}}{A_1}$$

## **Distortion Analysis**

Spurious Free Dynamic Range, SFDR

The SFDR is the difference between the fundamental and the largest harmonic



SFDR and THD are usually determined by either the second or third harmonic

## **Distortion Analysis**

**Theorem:** In a fully differential symmetric circuit, all even-order terms are absent in the Taylor's series output for symmetric differential sinusoidal excitations !



When k is even, the corresponding term in [] vanishes



## Stay Safe and Stay Healthy !

# End of Lecture 4