EE 240C
Analog-Digital Interface Integrated Circuits

Topics and Motivation

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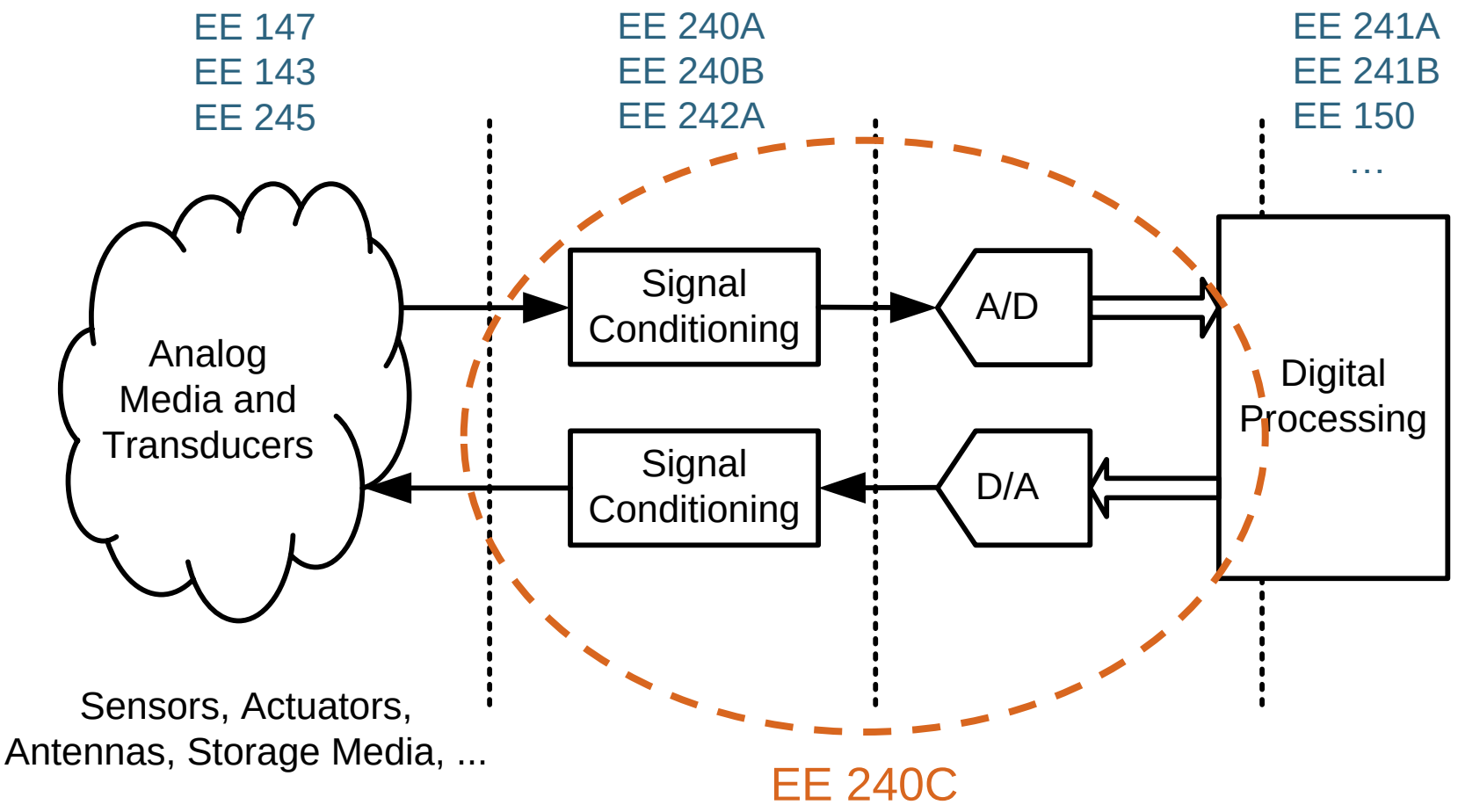
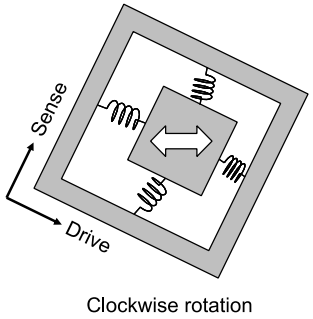
(Based on slides by Bernhard Boser)

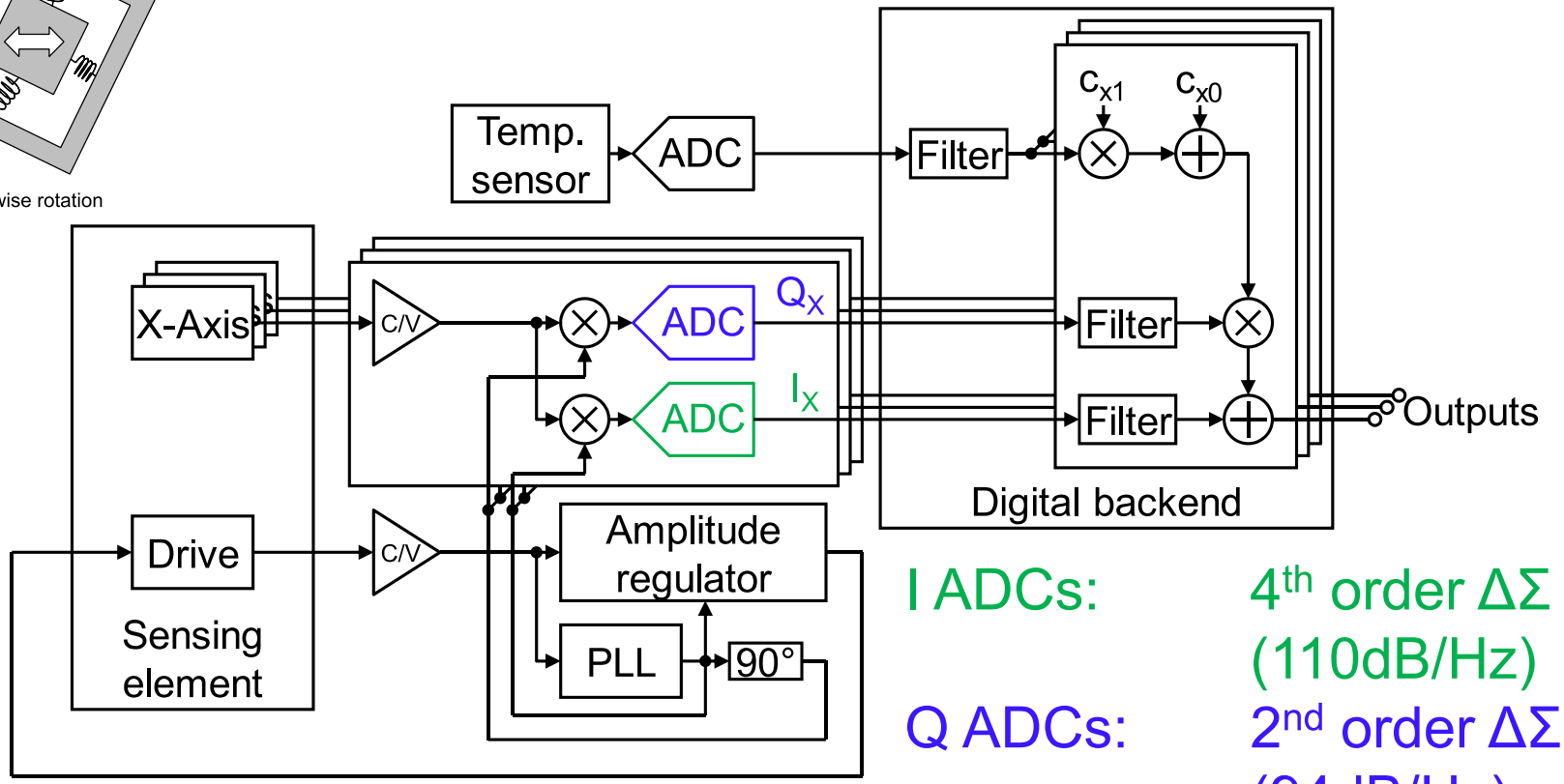
Mixed Signal System



Example: Gyroscope Front-End



Clockwise rotation

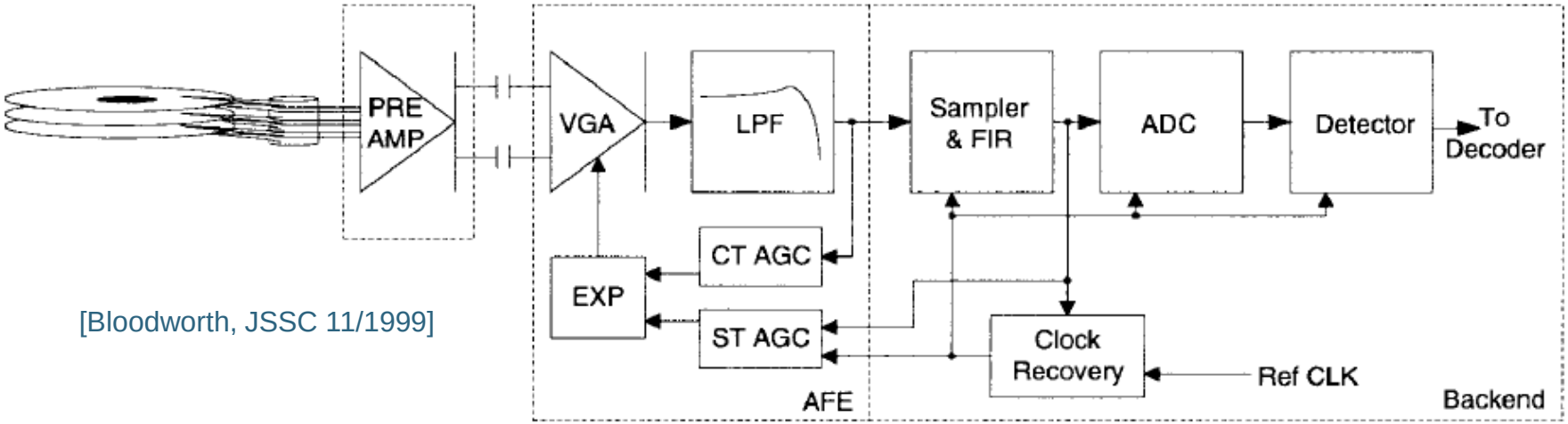


I ADCs: 4^{th} order ΔΣ (110dB/Hz)

Q ADCs: 2^{nd} order ΔΣ (94dB/Hz)

Ezekwe, Chinwuba D., Wolfram Geiger, and Torsten Ohms. "27.3 A 3-axis open-loop gyroscope with demodulation phase error correction." Solid-State Circuits Conference-(ISSCC), 2015 IEEE International. IEEE, 2015.

Example : Hard Disk Drive

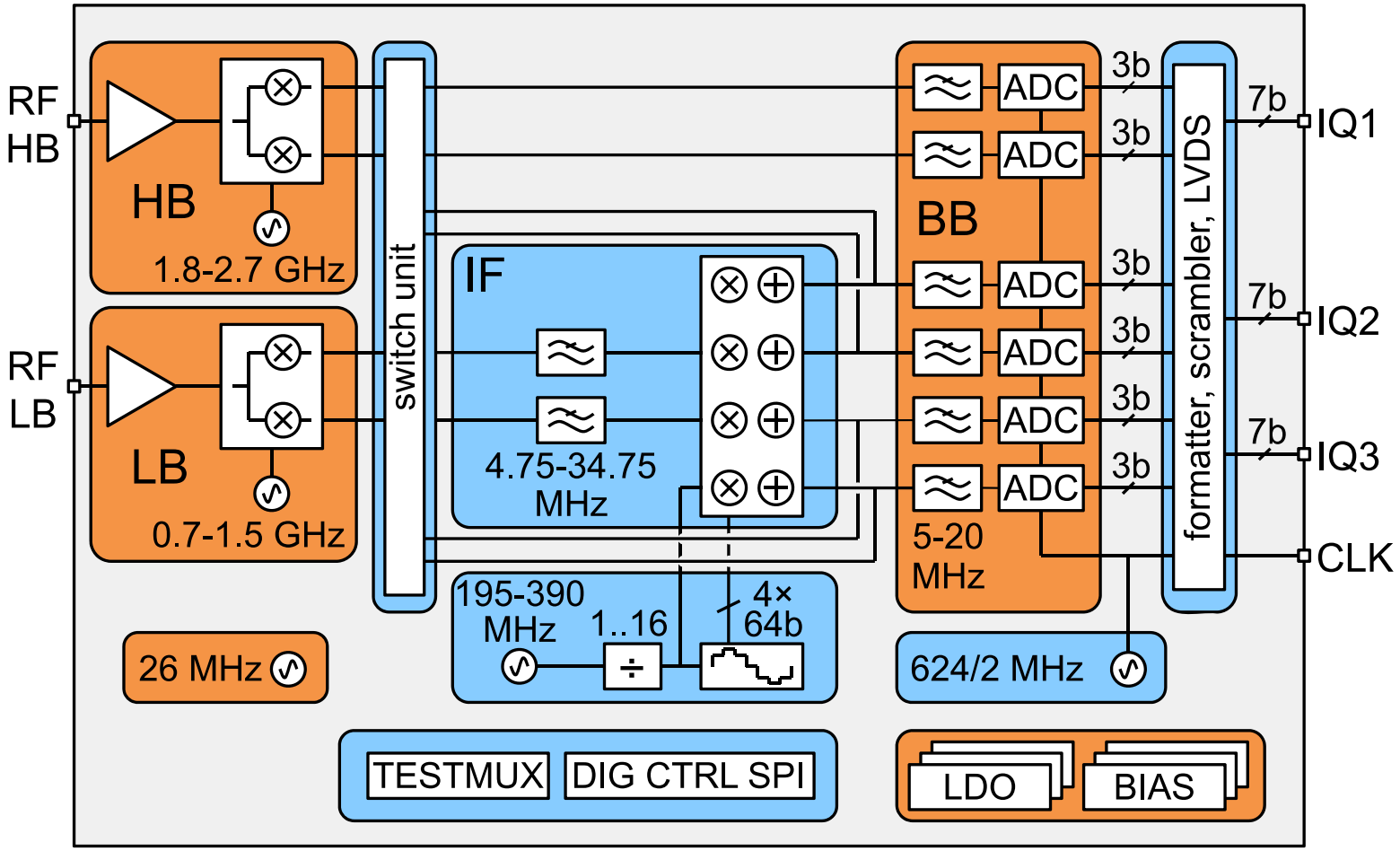


[Bloodworth, JSSC 11/1999]

AGC Specifications	
Input Signal Range	45 mVppd < V_{IN} < 500 mVppd
Output Target	1.400 Vppd
PGA Gain Settings	-3 dB, 3 dB, 9 dB
VGA Gain Range	-24 dB ≤ A_V ≤ 0 dB
w/Extended Range	-30 dB ≤ A_V ≤ 6 dB
AFE THD	< 1.0 %
AFE SNR	> 35 dB
AFE Output Offset	< 5.0 mV

LPF Specifications	
LPF Bandwidth	20 MHz ≤ f_c ≤ 120 MHz
Cutoff Variation	< ± 2.5 %
Group Delay Variation ($0.30f_c$ to f_c)	< 5 %
Group Delay Variation (f_c to $1.75f_c$)	< 8 %
Group Delay Adjustment	± 30 %
Boost	0 dB to 15 dB at f_c
CT Acquisition Time	< 10 bytes
Power Dissipation	< 250 mW

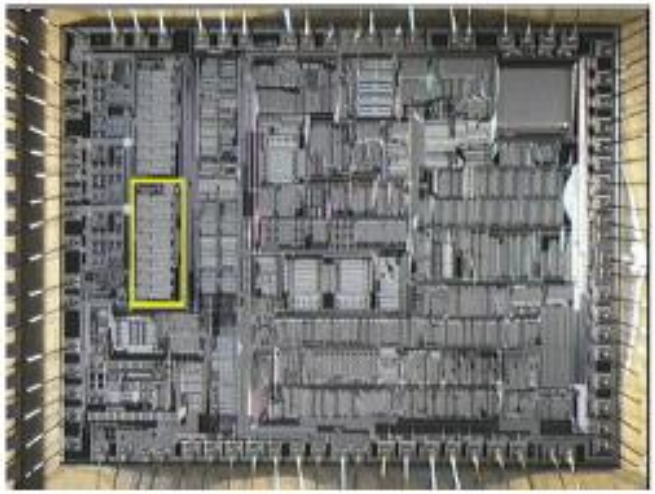
Exampe: Cellular LTE Front-End



Sundström, Lars, et al. "A receiver for LTE Rel-11 and beyond supporting non-contiguous carrier aggregation." *Solid-State Circuits Conference Digest of Technical Papers (ISSCC), 2013 IEEE International*. IEEE, 2013.

Why Digital Processing?

- Digital circuitry:
  - Cost/function decreases by 29% each year → 30X in 10 years
- Analog circuitry:
  - Cost/function is constant
  - Dropping supply voltages threaten feasibility



Digital Television IC:
1um (1995) vs 0.35um (2002)
- Digital: 10x smaller
- Analog: approximately same size

[M. Pelgrom, *Analog-to-Digital Conversion*, 2nd Ed., Springer, 2013]

Why Analog Processing?

- The “real” or “physical” world is analog

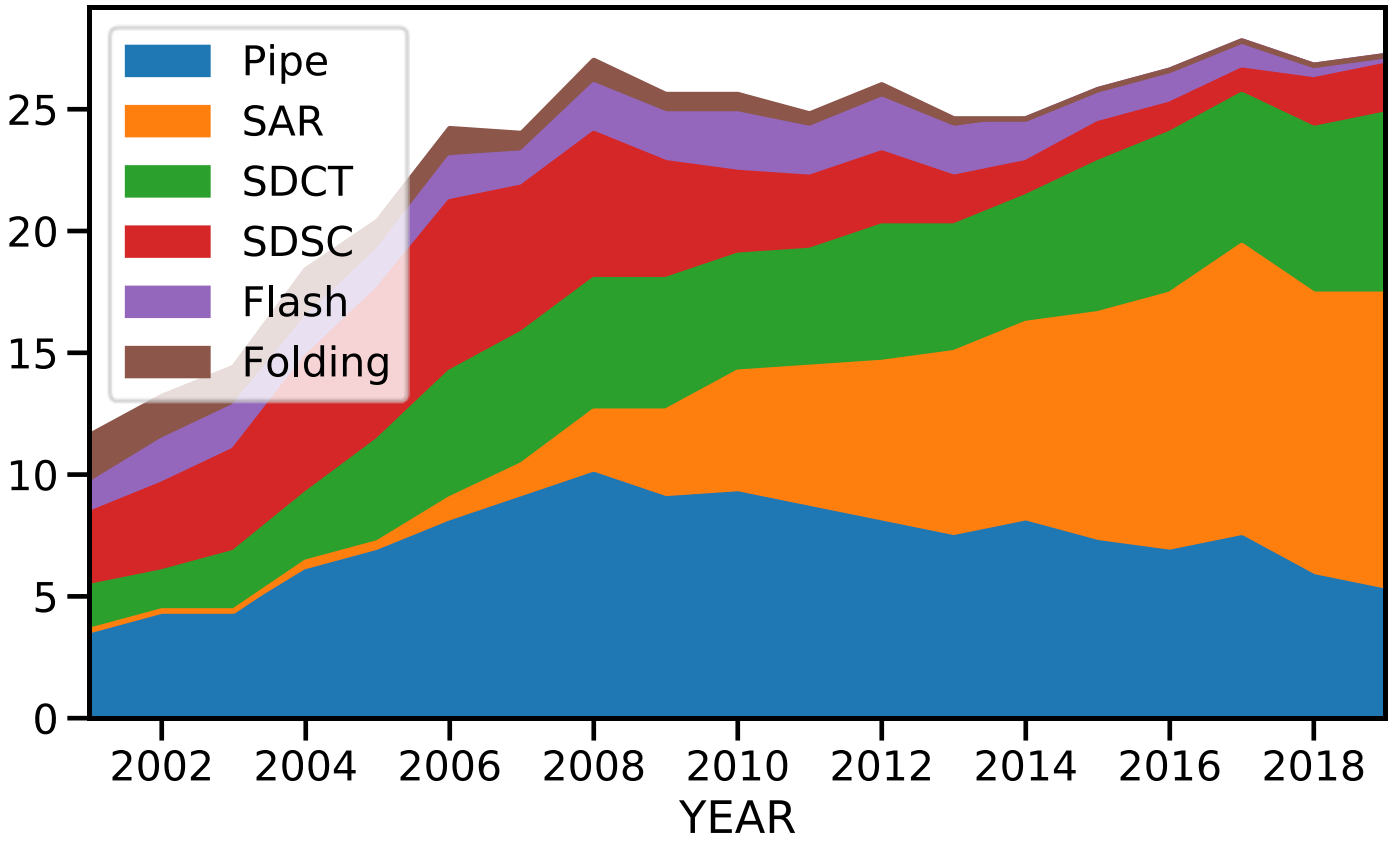
- Examples:
  - Digital Audio
  - RF receiver
  - Wireline communications

Course Topics

- Analog-Digital Interfaces
  - A/D Converters & D/A Converters
  - Sampling, Quantization
  - Performance Metrics & Limits
  - Architecture Tradeoffs

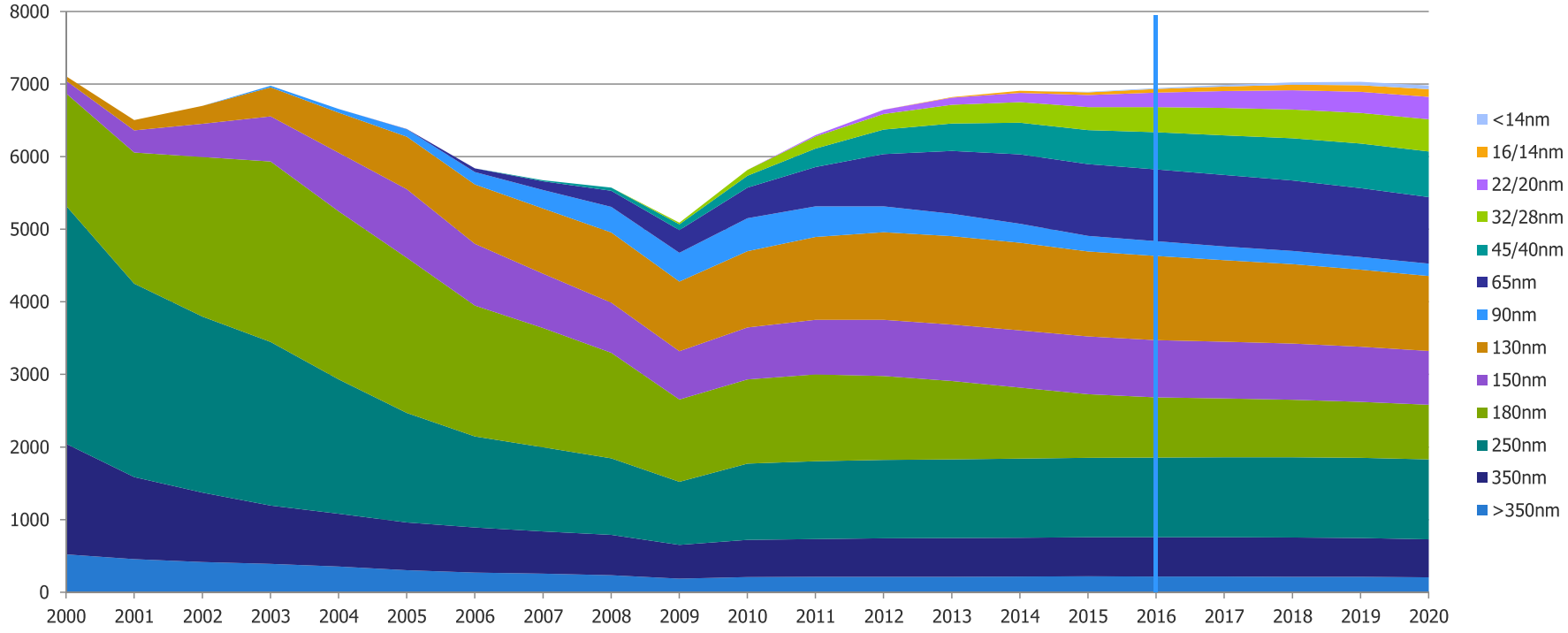
- Filters
  - Types of filters
  - Synthesis
  - Implementations:
    - Biquads versus Ladders
    - Continuous Time versus Switched Capacitor

Architectures of Published ADCs (ISSCC and VLSI)



- Evolution due to ADC improvements, technology shift, and focus on low power
- Source data: B. Murmann, "ADC Performance Survey 1997-2019," [Online]. Available: http://web.stanford.edu/~murmann/adcsurvey.html
- TI arch. combined with core arch. (e.g. SAR and TI SAR); other arch, (10-15%) not incl.

IC Design Start per Technology Node



Source: International Business Strategies, Inc. 2015

R. Subramanian- Semicon West 2016

Complexity in IC Design

- Model hierarchy
  - Behavioral — MATLAB, Simulink, CppSim, System-Verilog, Verilog-A(MS)
  - Structural (wo/ and w/ non-idealities) — MATLAB, Simulink, CppSim, System-Verilog, Verilog-A(MS)
  - Circuits — SPICE, Spectre
  - Silicon
- Development practices
  - Version control
  - Unit tests

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Analog-Digital Interface Integrated Circuits

Organization

Johan Vanderhaegen
jpv@berkeley.edu

Instructor

Johan Vanderhaegen

- IC Design Lead,
  Google / Verily Life Sciences,
  2015 – Present
- IC Designer (ADC, sensor, and sensor signal path topics)
  Robert Bosch Research and Technology Center, 2005 – 2015
- Graduate Student,
  UC Berkeley

- jpv@berkeley.edu

- Lectures:
  Wed 9am–12pm 299 Cory
- Office Hours:
  Wed 8am – 9am 299 Cory
  Mon 3:30pm – 4:30pm
  https://meet.google.com/ffa-uvqk-ayy

Course Website

- bcourses (slides)
- http://inst.eecs.berkeley.edu/~ee240c

References

- Books (not required)
  - *Converters*
    - M. Pelgrom, *Analog-to-Digital Conversion*, Springer, 3rd Ed., 2017
    - F. Maloberti, *Data Converters*, Springer, 2007
    - M. Gustavsson, J. Wikner, N. Tan, *CMOS Data Converters for Communications*, Kluwer, 2002
    - R. v. d. Plassche, *CMOS Integrated Analog-to-Digital and Digital-to-Analog Converters*, 2nd Ed., Kluwer, 2003
  - *Oversampling Converters*
    - S. Pavan, R. Schreier, G. Temes, *Understanding Delta-Sigma Data Converters*, 2nd Ed., Wiley-IEEE Press, 2017
    - S. Norsworthy, R. Schreier, G. Temes, *Delta-Sigma Data Converters*, Wiley-IEEE Press, 1996

References

- Books
  - *Filters*
    - K. Su, *Analog Filters*, 2nd Ed., Kluwer, 2002
    - R. Schaumann, H. Xiao, M. Van Valkenburg, *Design of Analog Filters*, 2nd Ed., Oxford University Press, 2009
    - A. Zverev, *Handbook of Filter Synthesis*, Wiley-Interscience, 1963

- Journals and Conferences, http://ieeexplore.ieee.org
  - IEEE Journal of Solid-State Circuits (JSSC)
  - IEEE International Solid-State Circuits Conference (ISSCC)
  - …

Prerequisites

- Transistor-level circuit design course
  - Device physics and models
  - Transistor level analog circuits, elementary gain stages
  - Frequency response, feedback, noise
  - E.g. EE 140 or 240A, 240 B
    - Gray, Hurst, Lewis, Meyer, *Analysis and Design of Analog Integrated Circuits*, 5th Ed, Wiley, 2011
    - Razavi, *Design of Analog CMOS Integrated Circuits*, 2nd Ed. McGraw-Hill.
- Familiarity with Spectre, Matlab, Simulink, Verilog-A
- Basic signals and systems
  - Laplace and z-transforms
  - Noise power spectral density

Assignments

- Homework: (20%)
  - Lowest HW score is dropped in final grade calculation
- Project: (30%)
  - Design of a high performance interface
  - Class project
  - Project report in the format of an IEEE journal paper
- Midterm & Final Exam (50%)
  - Midterm: Oct 16th (in class)
  - Final: Thu Dec 19th

Acknowledgements

- Bernhard Boser, UC Berkeley
- Paul Gray, UC, Berkeley
- Boris Murmann, Stanford University
- Eric Swanson, Crystal Semiconductor
- Haideh Khorramabadi
- Aaron Buchwald, Mobius Semiconductor
- Countless books and articles

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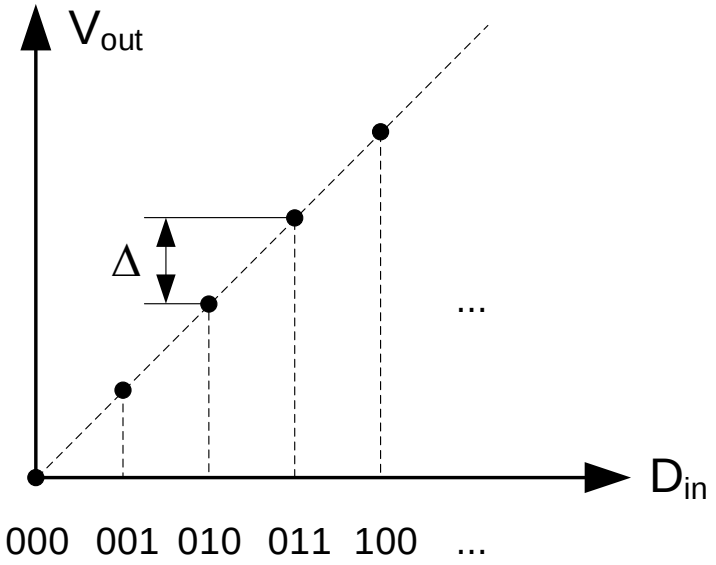
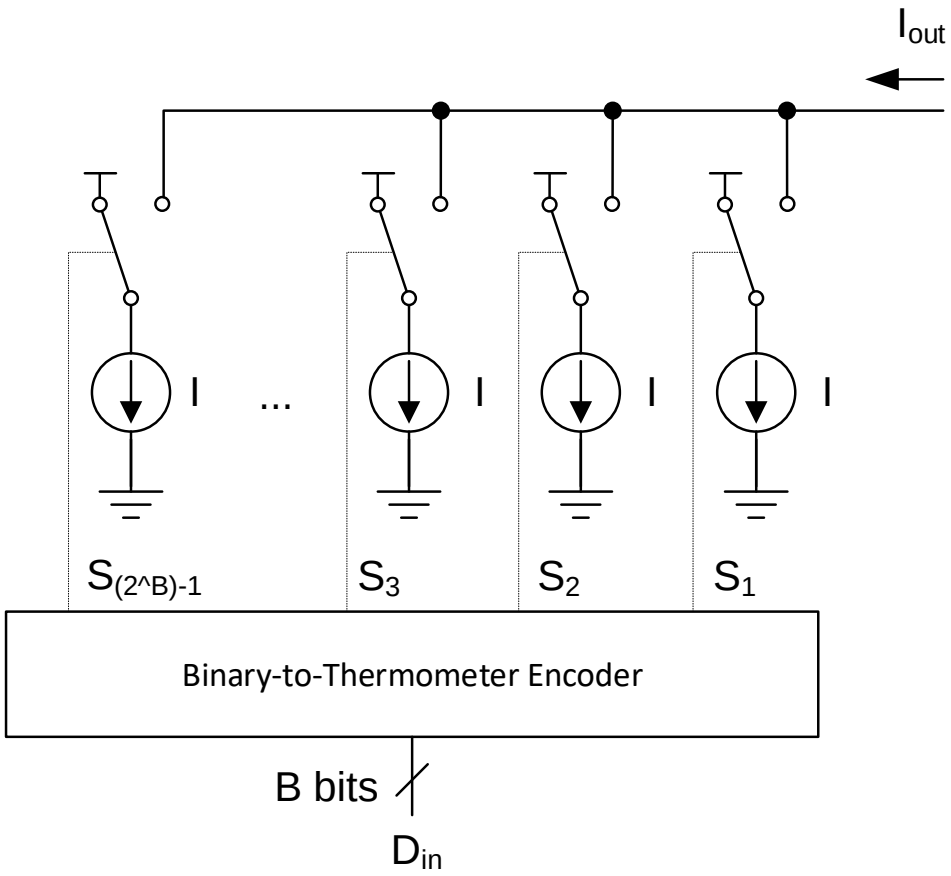
Digital-to-Analog Converter

Digital-to-Analog Converter

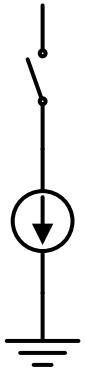


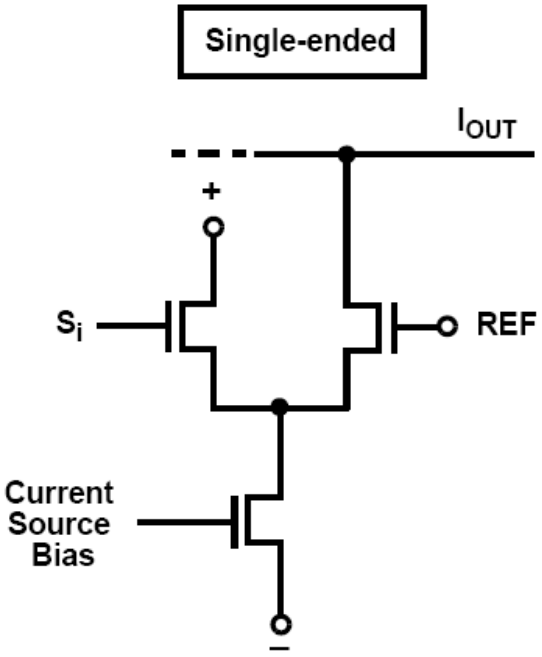
Realization?

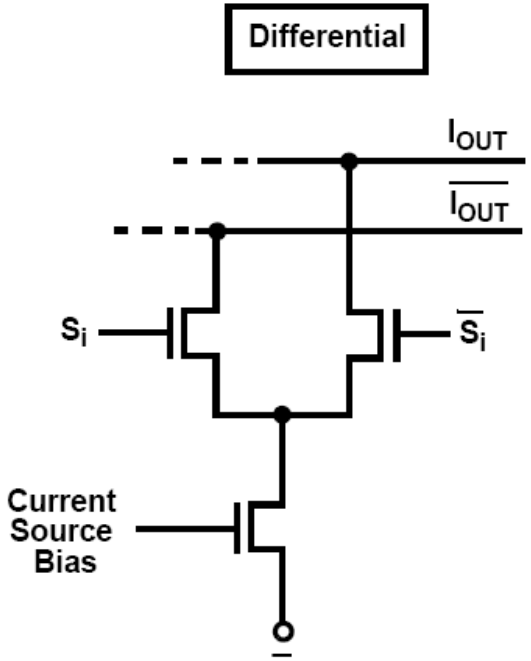
Current DAC ("M-DAC")



Circuit Realization



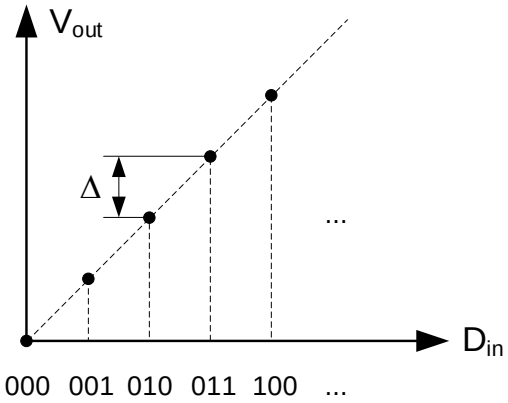


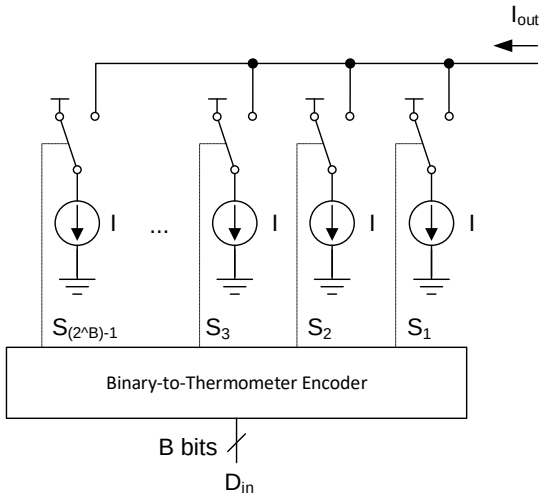


Other DAC implementations:

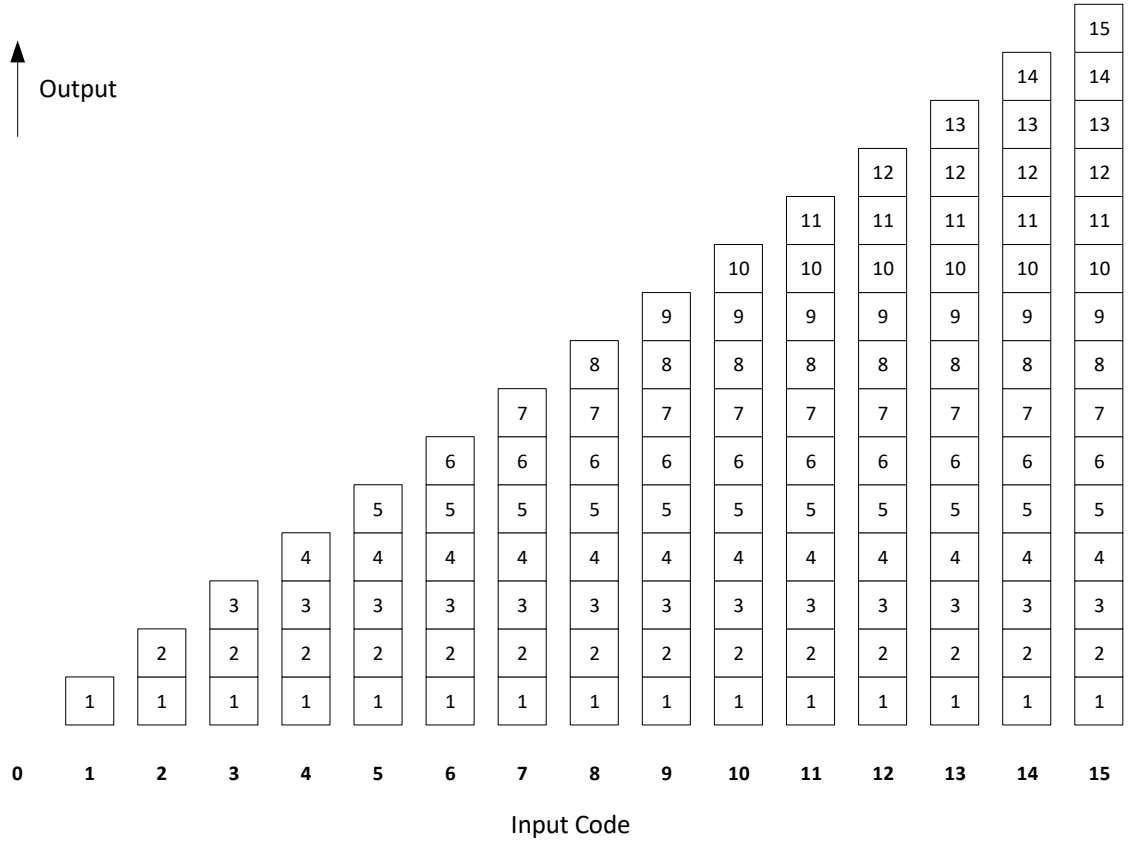
- Resistor string DAC
- Charge redistribution DAC

How Many Current Sources?





Thermometer DAC Operation



[Boris Murmann]

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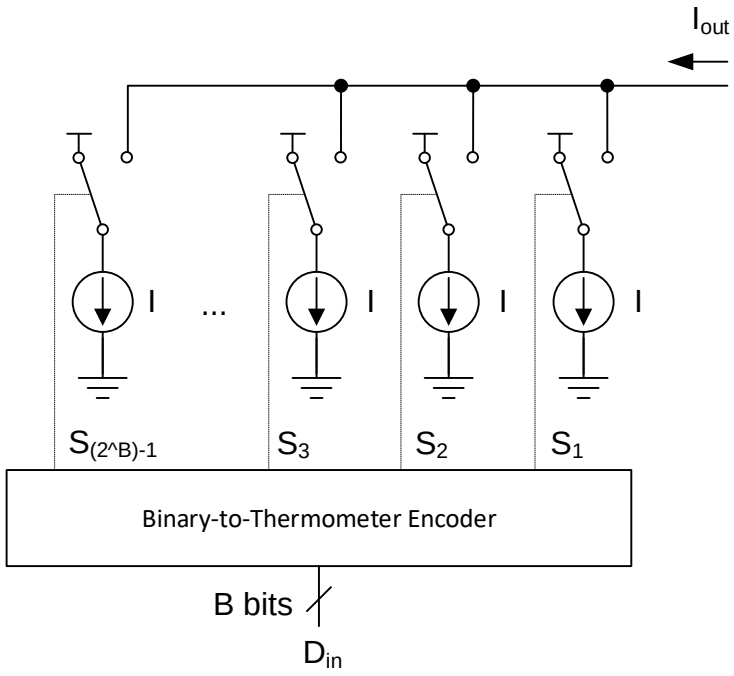
Digital-to-Analog Converter

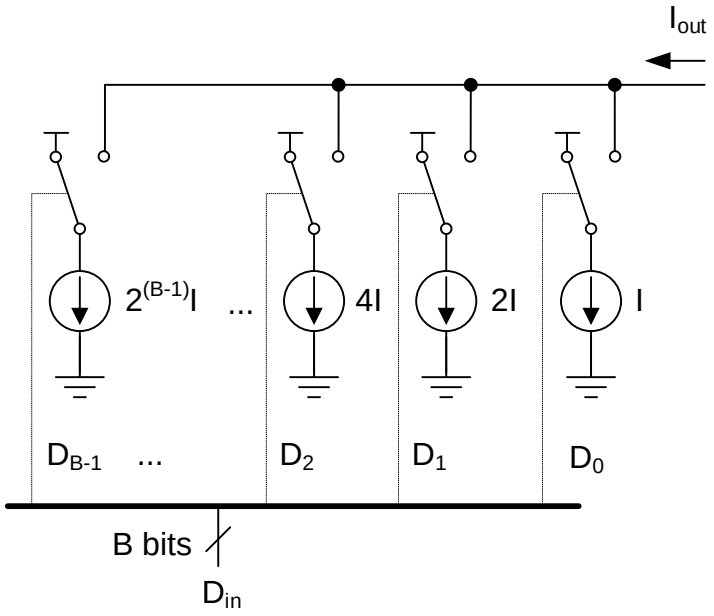
Current DAC ("M-DAC")

Unit Element



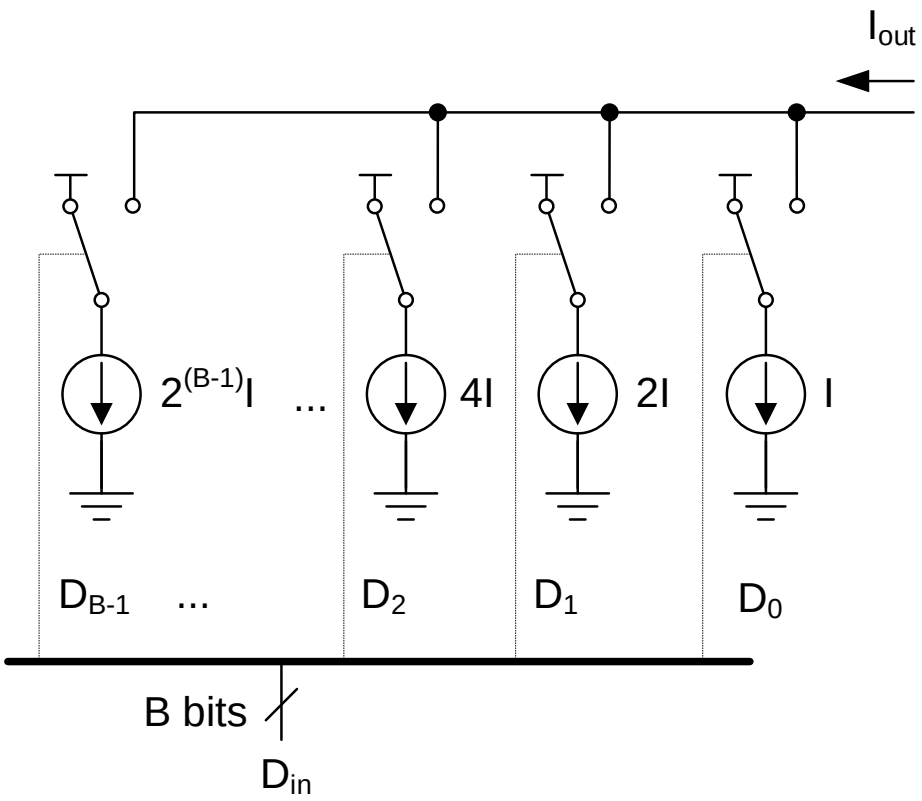
2^B-1 switches

Binary Weighted



B switches

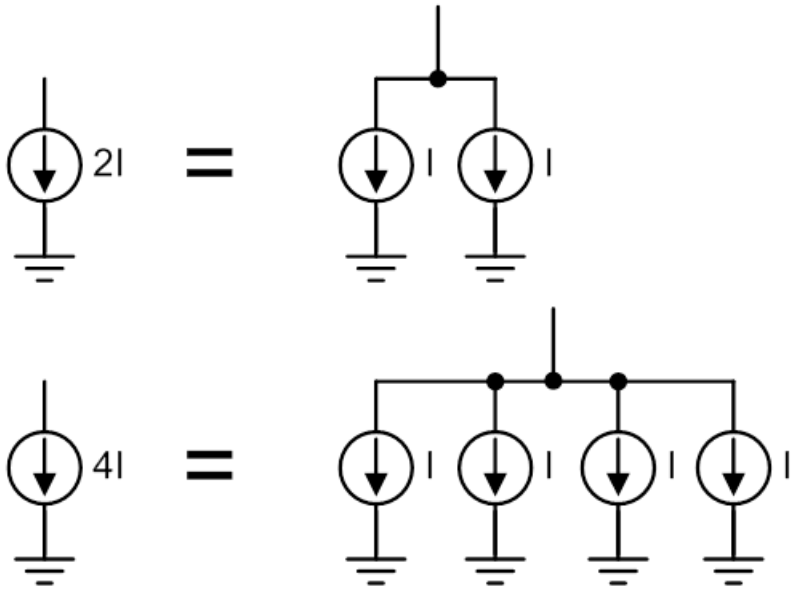
Binary Weighted DAC



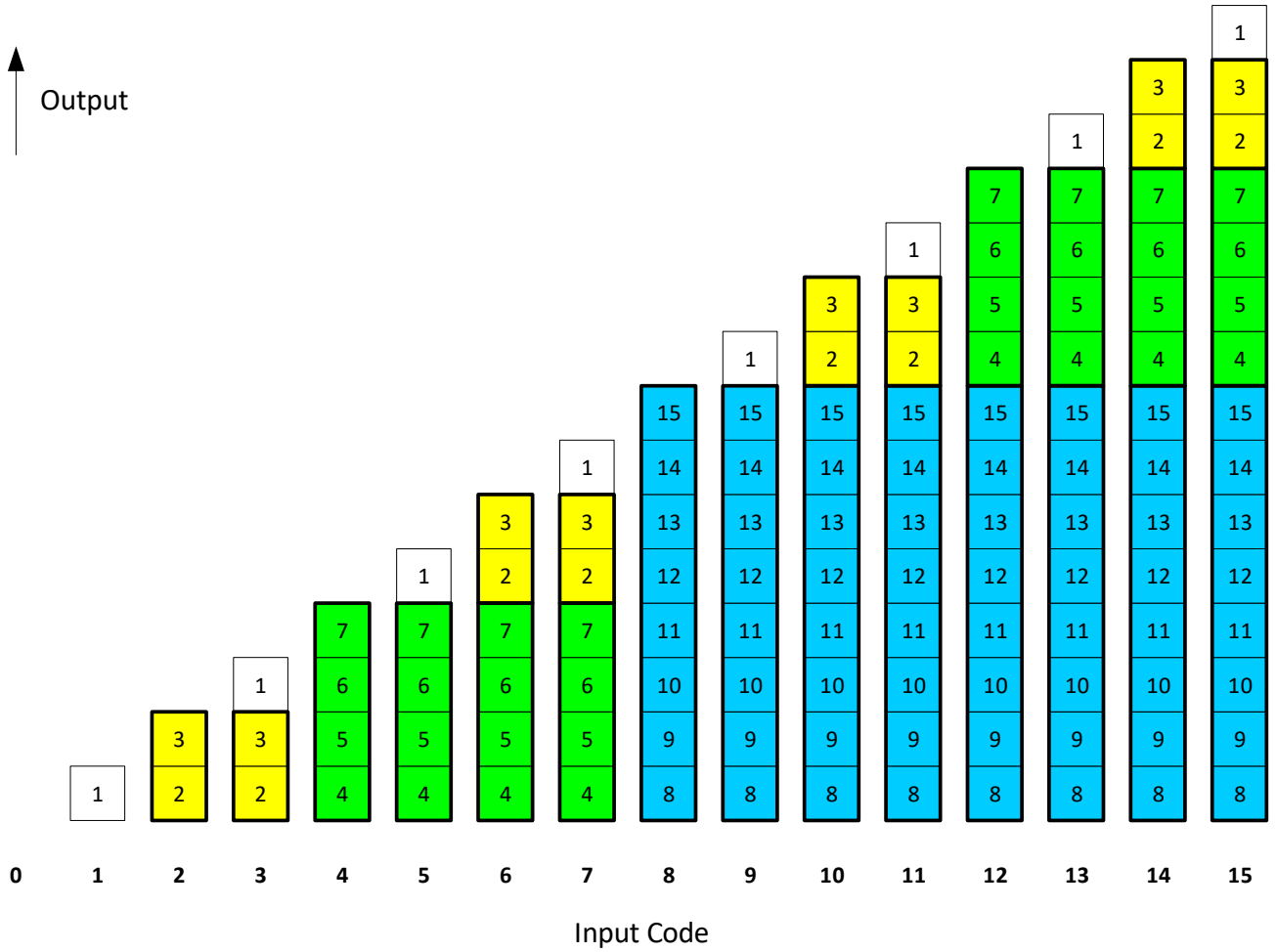
- No encoder needed
- B switches
- How many current sources?

Implementation of Weighted Sources

Transistor-level design:



Weighted DAC Operation



[Boris Murmann]

DAC Comparison

Thermometer

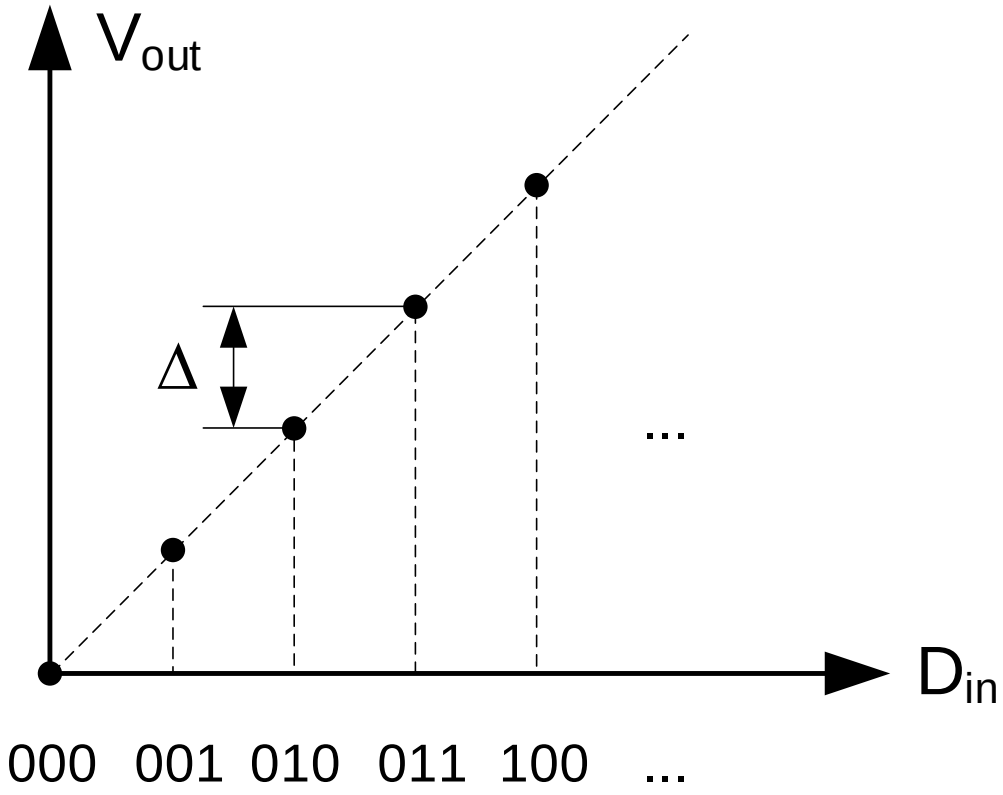
- 2^B-1 switches
- Encoder

Binary Weighted

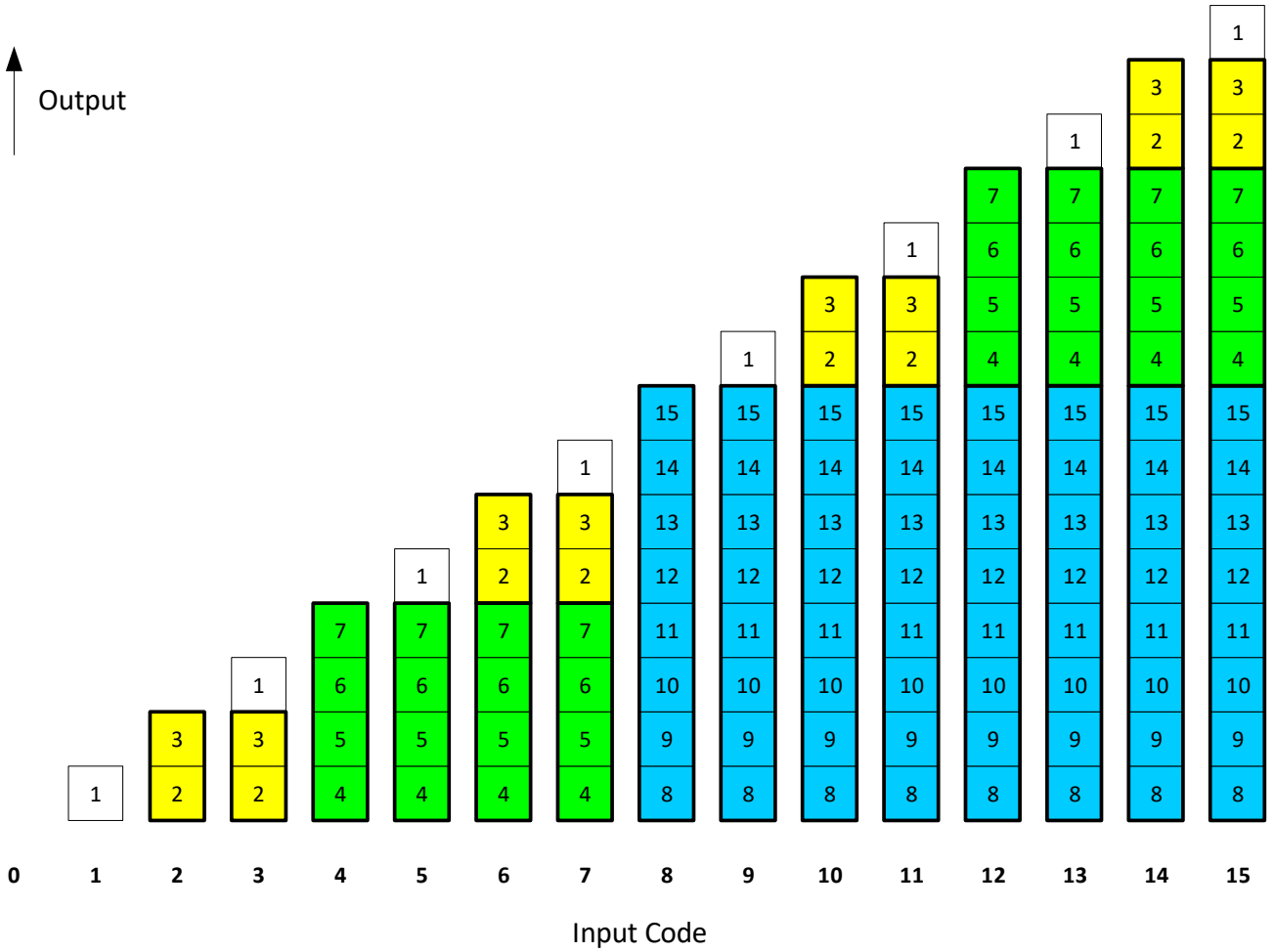
- B switches
- No encoder

Which one is “better”?

Monotonicity



Non-Monotonic Weighted DAC



[Boris Murmann]

Questions

- What is causing imperfections such as non-monotonicity?
- What metrics can we use to quantify the errors?

- Let's look at these questions in detail.
- This will take several lectures!

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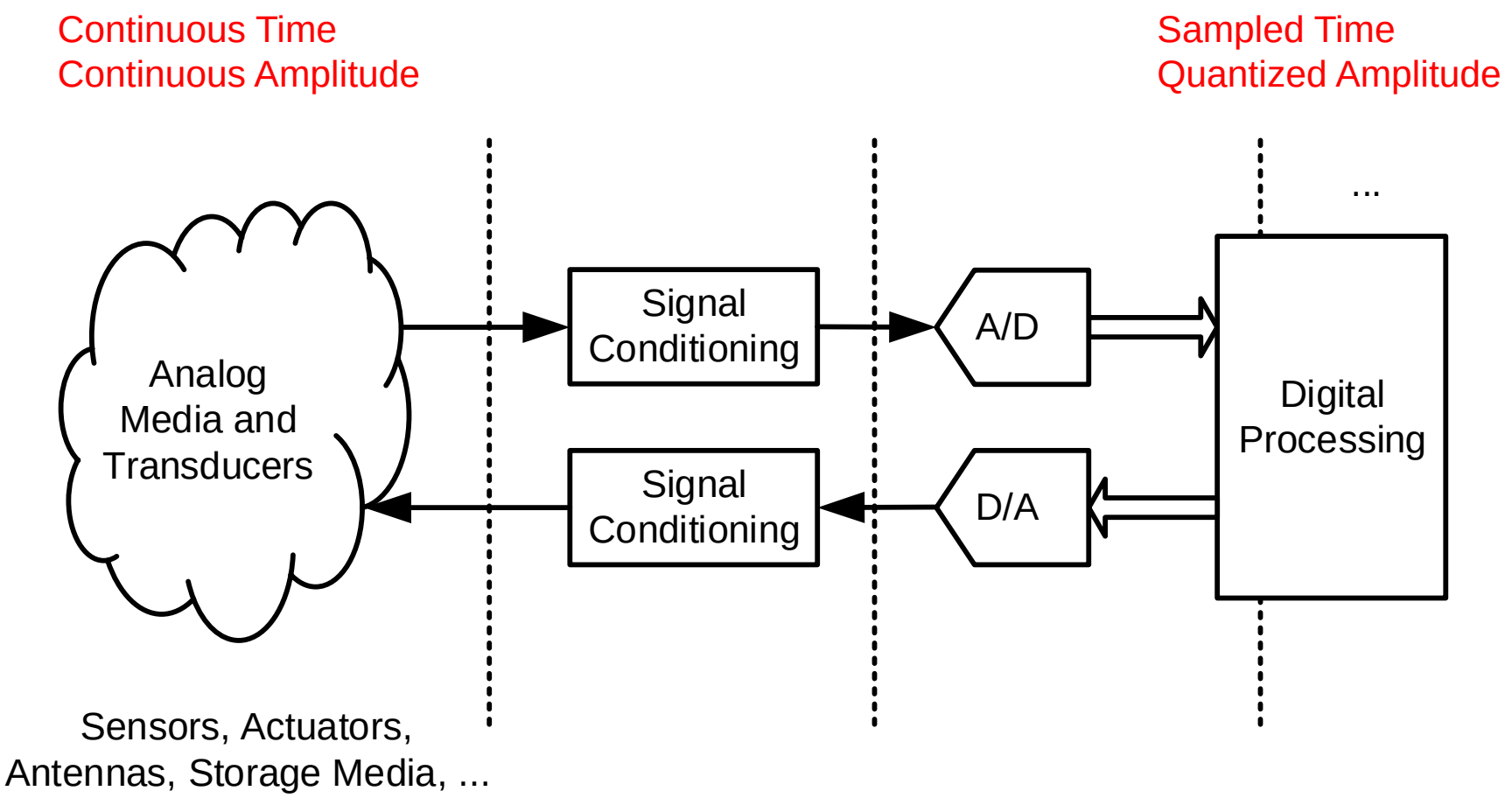
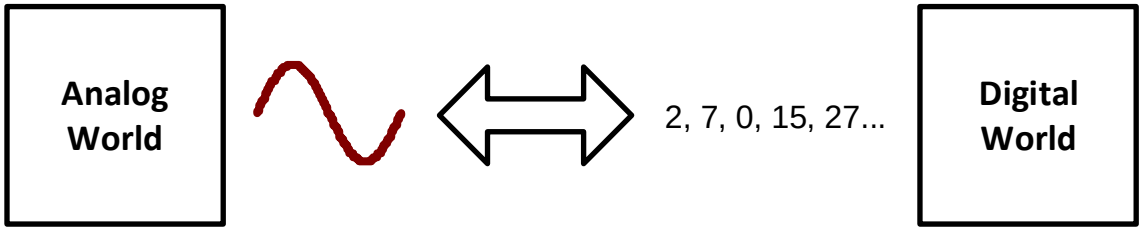
Sampling and Quantization

Mixed Signal System

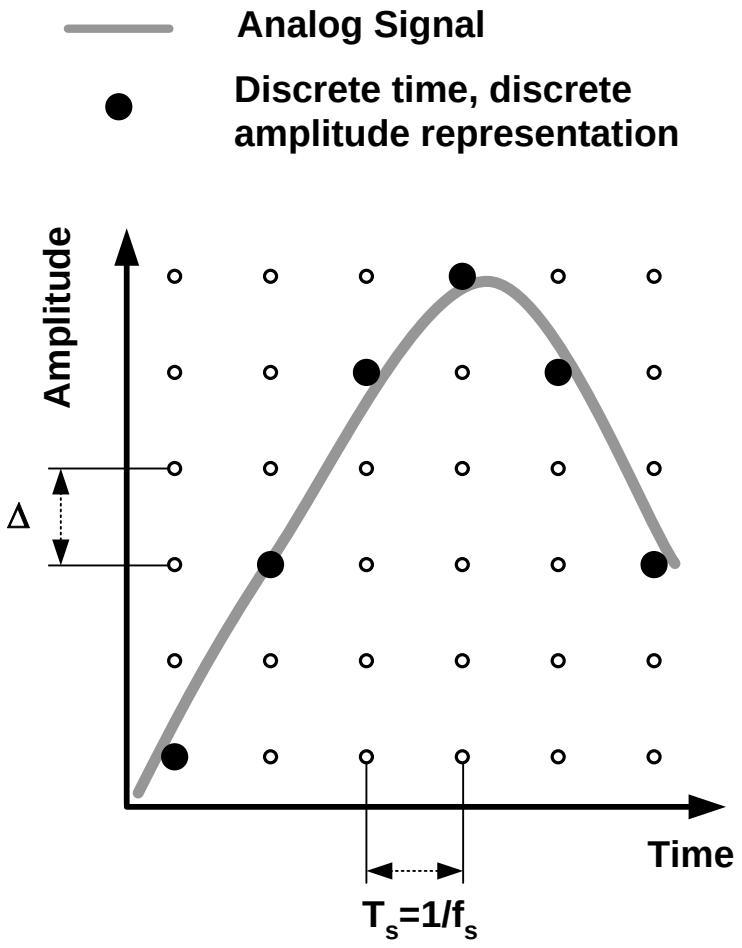


Quantized Signals



- Real world signals
  - Continuous time, continuous amplitude
- Digital abstraction
  - Discrete time, discrete amplitude
- Two problems – how to discretize / ”undescretize”
  - in time (sampling / reconstruction)
  - in amplitude (quantization / …)
- Often performed in conjunction
  - A/D conversion: discretize in time and amplitude
  - D/A conversion: “undiscretize” in time and amplitude
  - Other valid combinations, especially discrete time & continuous amplitude
    - Examples?

Uniform Sampling and Quantization



Most common way of performing A/D conversion

- Sample signal uniformly in time
- Quantize signal uniformly in amplitude

Questions

- How fast do we need to sample ?
- How fine do we need to quantize ?

Questions

1. How fast do we need to sample an audio signal?
2. What is the criterion?
3. How about an RF signal modulated at a 1GHz carrier frequency with 1MHz bandwidth?
4. How do we prevent “other RF channels” from corrupting the signal?
5. How do we determine the “number of bits” of an A/D converter?
6. How do errors in the quantizer thresholds affect signal quality?
7. Is there a formal procedure for answering these and similar questions?
8. Your question?

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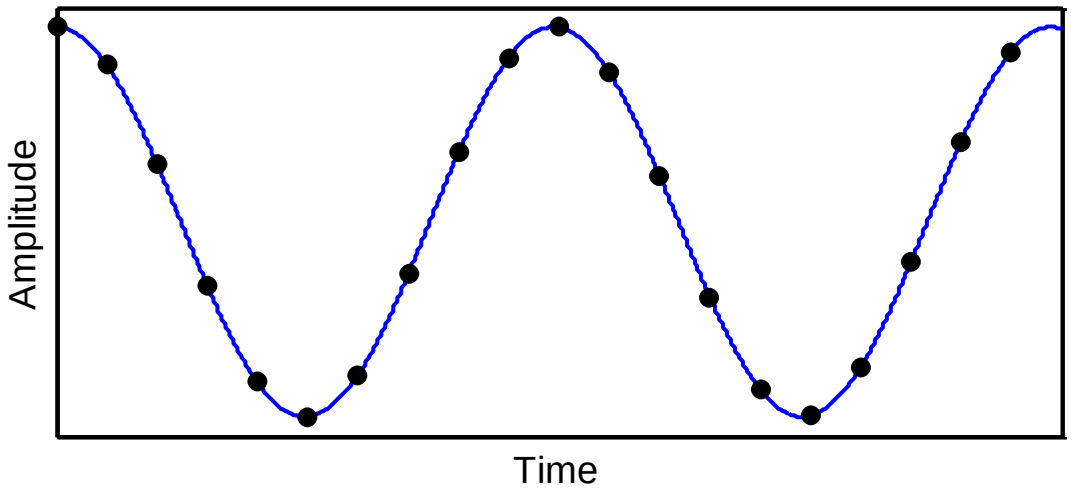
Sampling

Sampling Case (1)



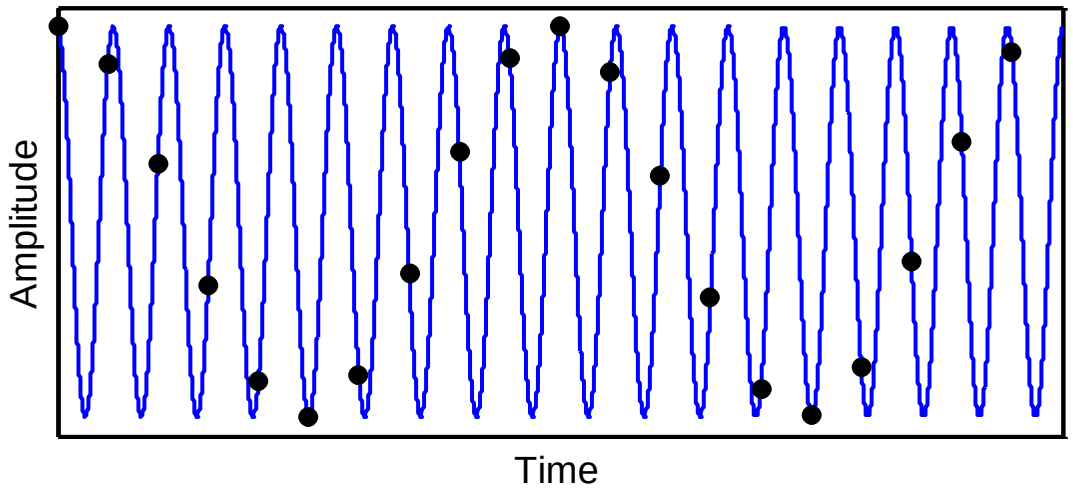
$$f_s = \frac{1}{T_s} = 1000\ kHz$$
$$f_{sig} = 101\ kHz$$

$$v_{sig}(t) = \cos(2\pi \cdot f_{in} \cdot t)$$

$$t \rightarrow n \cdot T_s = \frac{n}{f_s}$$

$$v_{sig}(n) = \cos\left(2\pi \cdot \frac{f_{in}}{f_s} \cdot n\right)$$
$$= \cos\left(2\pi \cdot \frac{101}{1000} \cdot n\right)$$

Sampling Case (2)

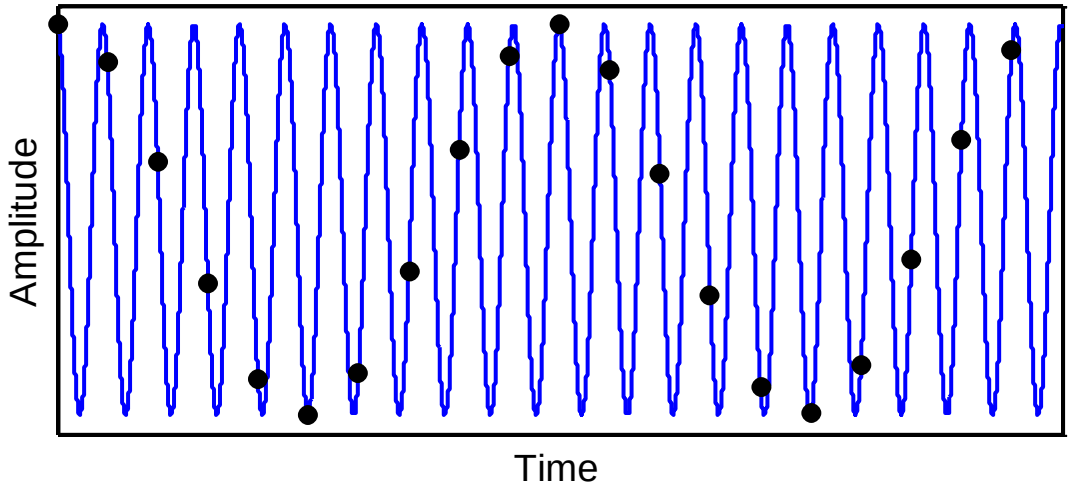


$$f_s = \frac{1}{T_s} = 1000\ kHz$$
$$f_{sig} = 899\ kHz$$

$$v_{sig}(n) = \cos\left(2\pi \cdot \frac{899}{1000} \cdot n\right) = \cos\left(2\pi \cdot \left[\frac{899}{1000} - 1\right] \cdot n\right) = \cos\left(2\pi \cdot \frac{101}{1000} \cdot n\right)$$

Sampled signal indistinguishable from Case 1 (101*kHz*)

Sampling Case (3)

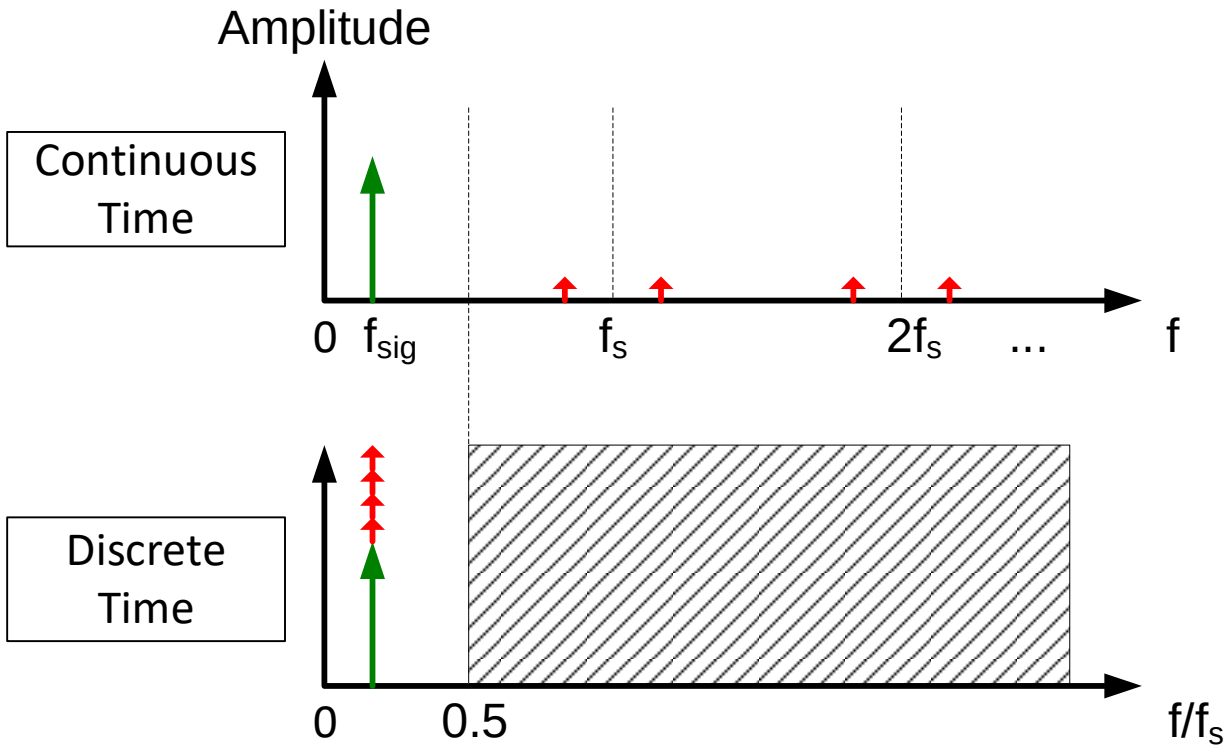


$$f_s = \frac{1}{T_s} = 1000\ kHz$$
$$f_{sig} = 1101\ kHz$$

$$v_{sig}(n) = \cos\left(2\pi \cdot \frac{1101}{1000} \cdot n\right) = \cos\left(2\pi \cdot \left[\frac{1101}{1000} - 1\right] \cdot n\right) = \cos\left(2\pi \cdot \frac{101}{1000} \cdot n\right)$$

Sinusoids with frequencies 101, 899, 1101*kHz* etc
sampled at 1000*kHz* are indistinguishable

Spectrum



- The frequencies f_{sig} and $N \cdot f_s \pm f_{sig}$ (N integer), are indistinguishable in the discrete time domain

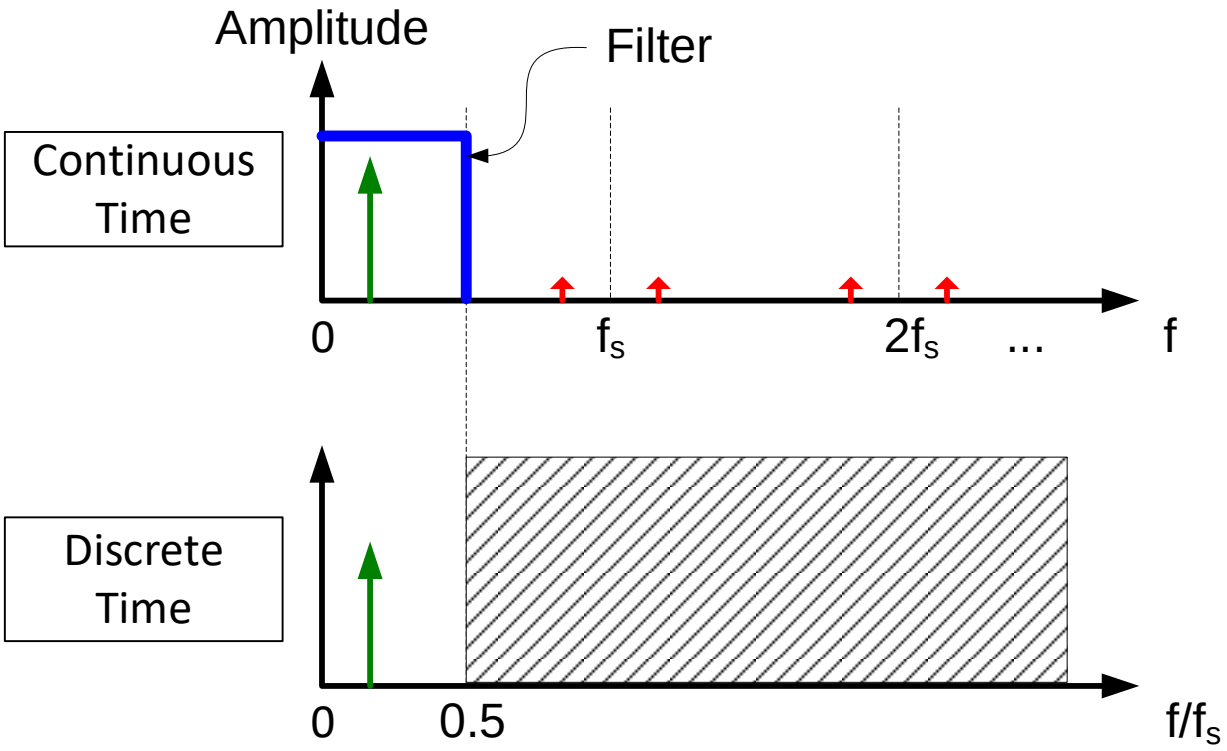
Sampling Theorem

- In order to prevent aliasing, we need

$$B_{sig} < \frac{f_s}{2}$$

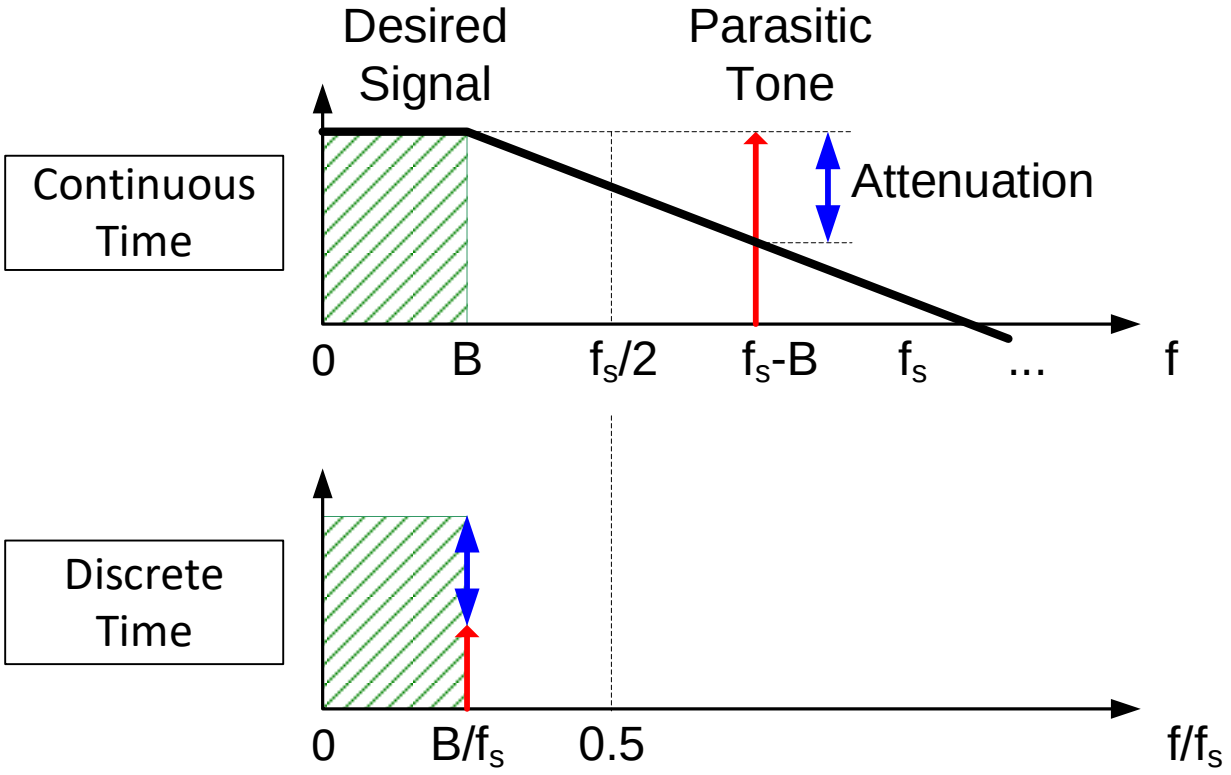
- B_{sig} is the signal bandwidth
  - For “baseband signals”, B_{sig} equals the “highest” signal frequency $f_{sig,max}$
- The sampling rate $f_s = 2 \cdot B_{sig}$ is called the Nyquist rate
- Two options:
  - Sample fast enough to cover all spectral components, including "parasitic" ones outside band of interest
  - Limit B_{sig} through filtering

Brick Wall Anti-Alias Filter



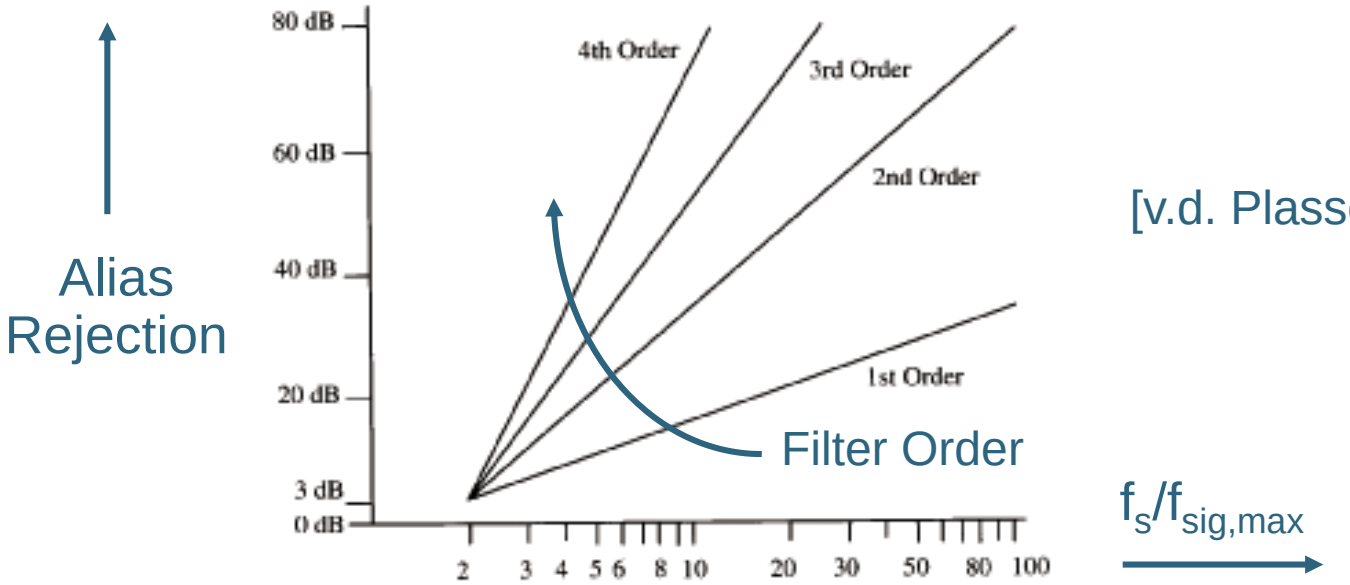
Most ideal contraptions cannot be realized in practice. Brick-Wall filters are no exception …

Practical Anti-Alias Filter



- Need to sample faster than Nyquist rate to get good attenuation
  - "Oversampling"

How much Oversampling?



[v.d. Plassche, p.41]

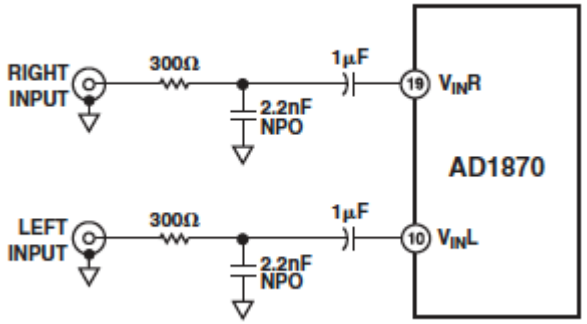
- Tradeoff sampling rate versus filter order
- In high speed converters, $f_s/f_{sig,max}>10$ is usually impossible or too costly → need fairly high order filters
- Question: how much alias rejection is required? Give an example!

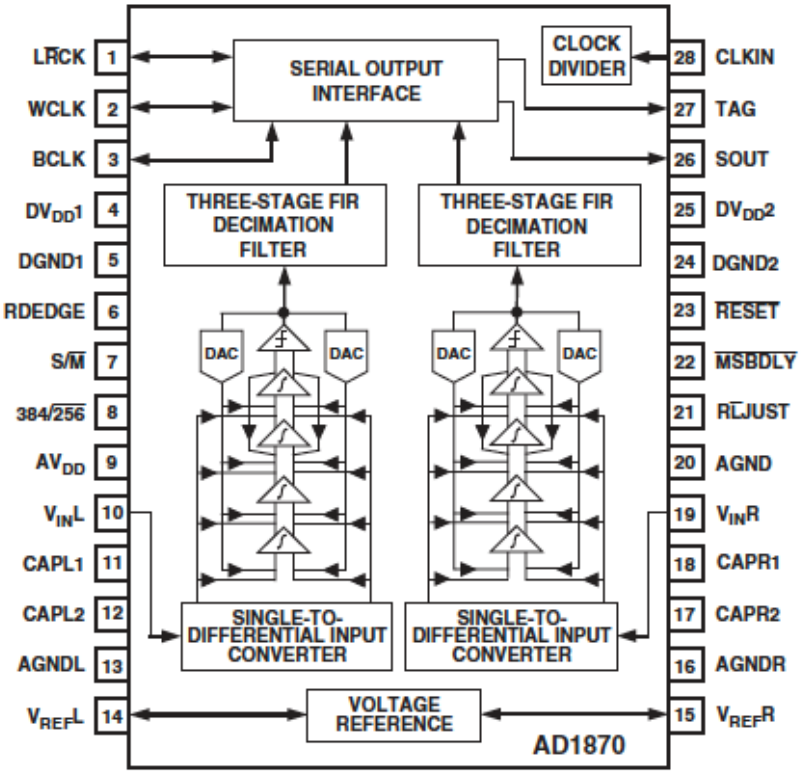
Types of Sampling

1. Nyquist-rate sampling ($f_s > 2 \cdot f_{sig,max}$)
   - Nyquist data converters
   - In practice always slightly oversampled
2. Oversampling ($f_s >> 2 \cdot f_{sig,max}$)
   - Slowly varying signals (e.g. temperature)
   - Oversampled data converters
   - Anti-alias filtering is often trivial or not needed
   - Explain "trivial"
3. Undersampling, subsampling ($f_s < 2 \cdot f_{sig,max}$)
   - Exploit aliasing to mix RF/IF signals down to baseband
   - See e.g. Pekau & Haslett, JSSC 11/2005

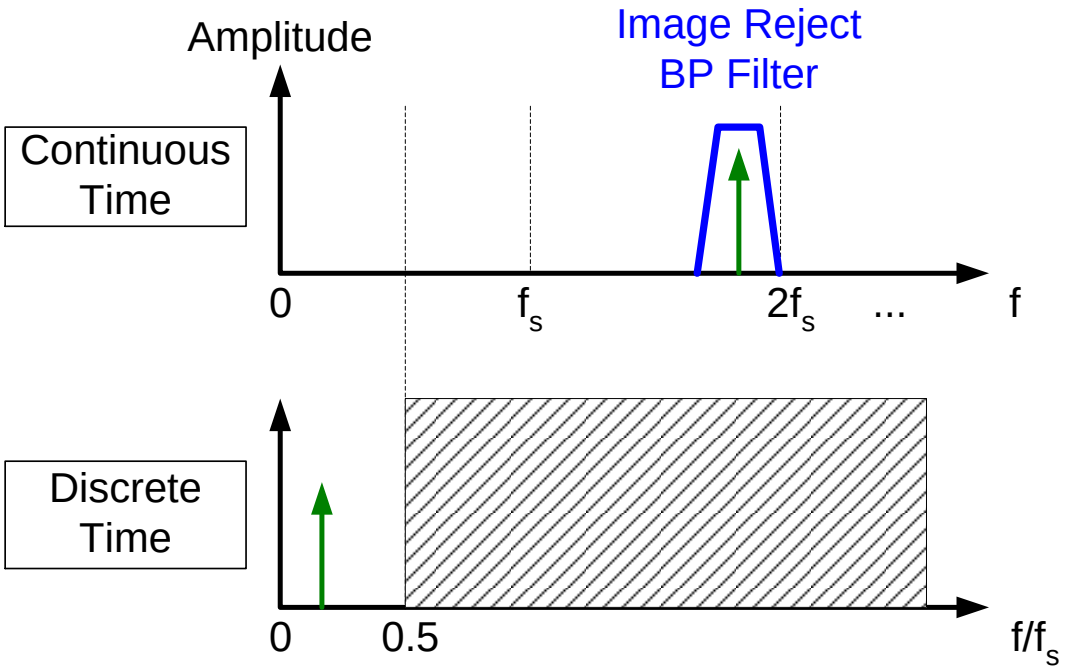
Oversampling Anti-Aliasing Filter

- CD Audio Signals: B = 20 KHz and f_s = 44.1 KHz
- Example: AD1870 Single-Supply 16-bit ΣΔ Stereo ADC
  ADC f_s = 64 x 44.1KHz = 2.8 MHz
  - First-order passive analog anti-alias filter with 22dB rejection at 3 MHz
  - Digital filter with 90dB stopband rejection





Subsampling

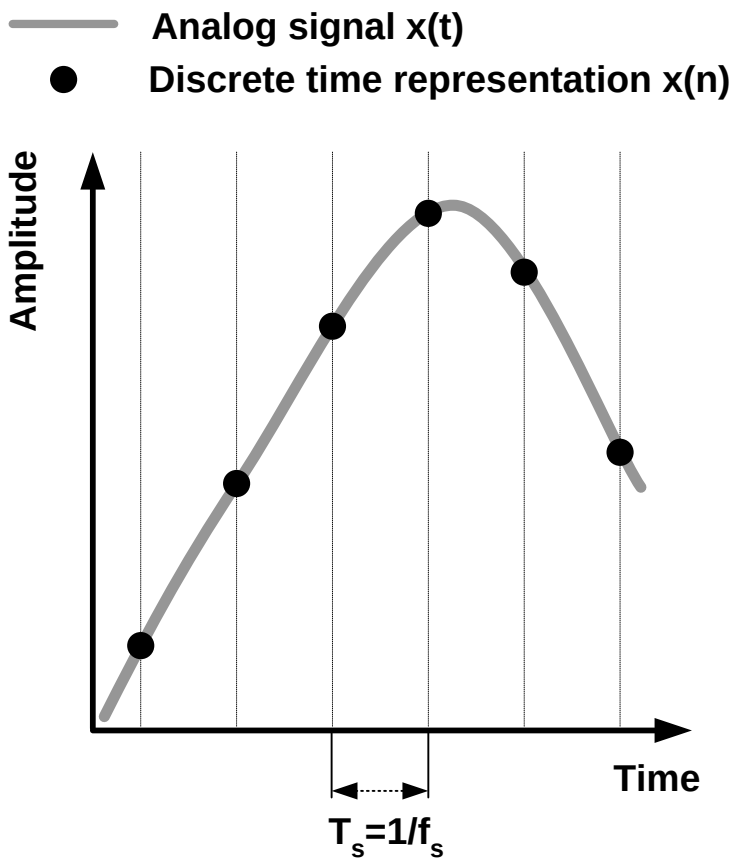


- Aliasing is "non-destructive" if signal is band limited around some carrier frequency
- Downfolding of noise is a severe issue in practical subsampling mixers
  - Typically achieve noise figure no better than 20 dB (!)

EE 240C
Analog-Digital Interface Integrated Circuits

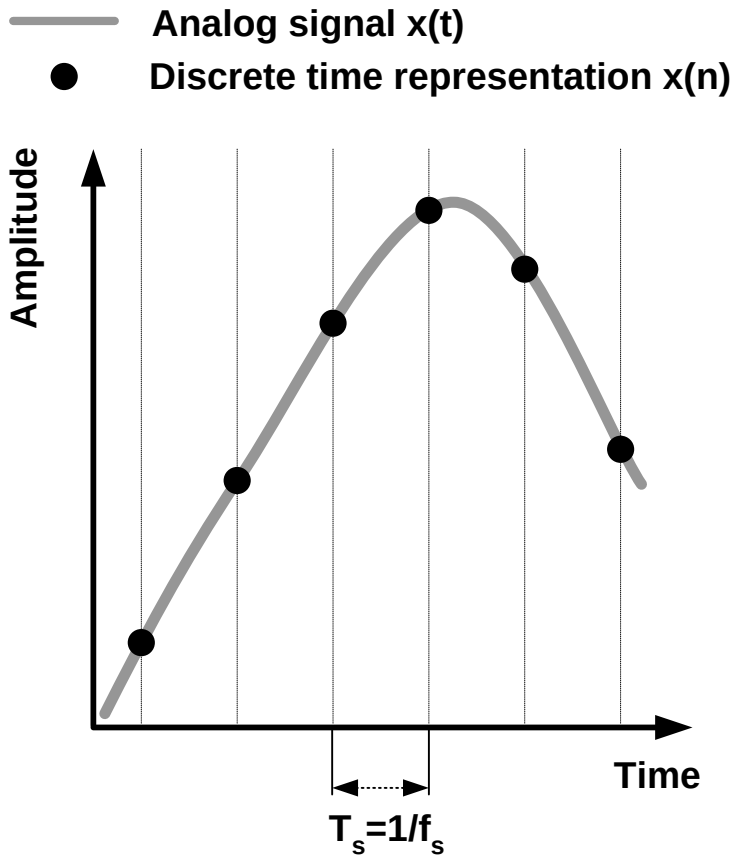
Reconstruction

Reconstruction Problem



- Known: discrete time samples x(n)
- Objective: recover x(t)
- Sampling theorem: signal spectrum preserved provided that $f_s > 2 \cdot B$
- Should be possible to perfectly recover original analog signal x(t)
- How?

Ideal Reconstruction



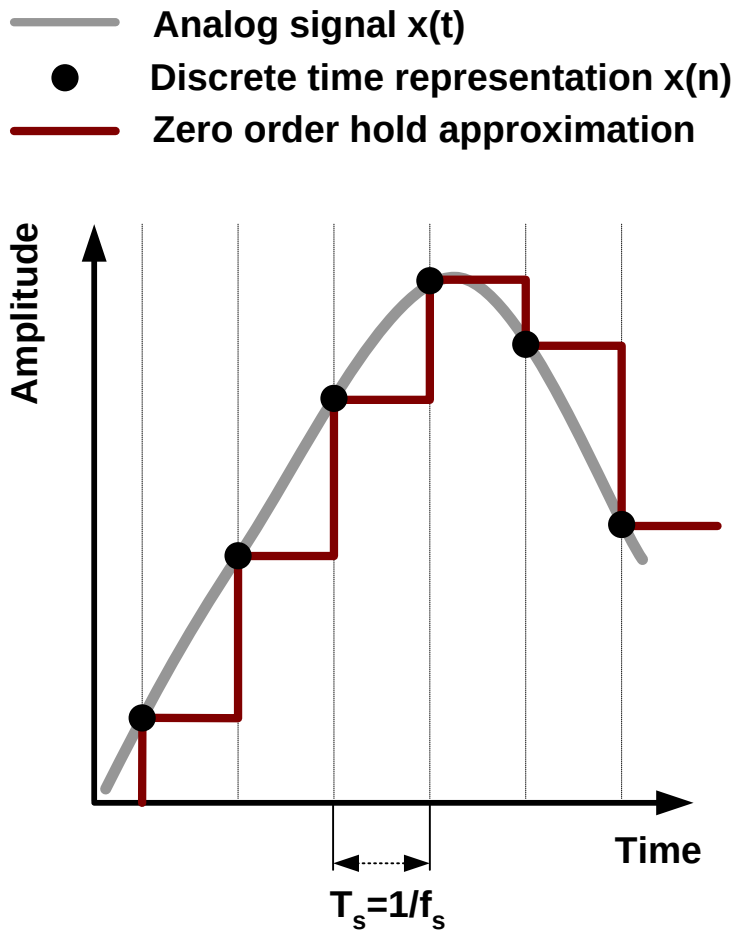
- Ideal interpolation

$$x(t) = \sum_{n=-\infty}^{\infty} x(n) \cdot g(t - nT_s)$$

$$g(t) = \frac{sin(\pi f_s t)}{\pi f_s t}$$

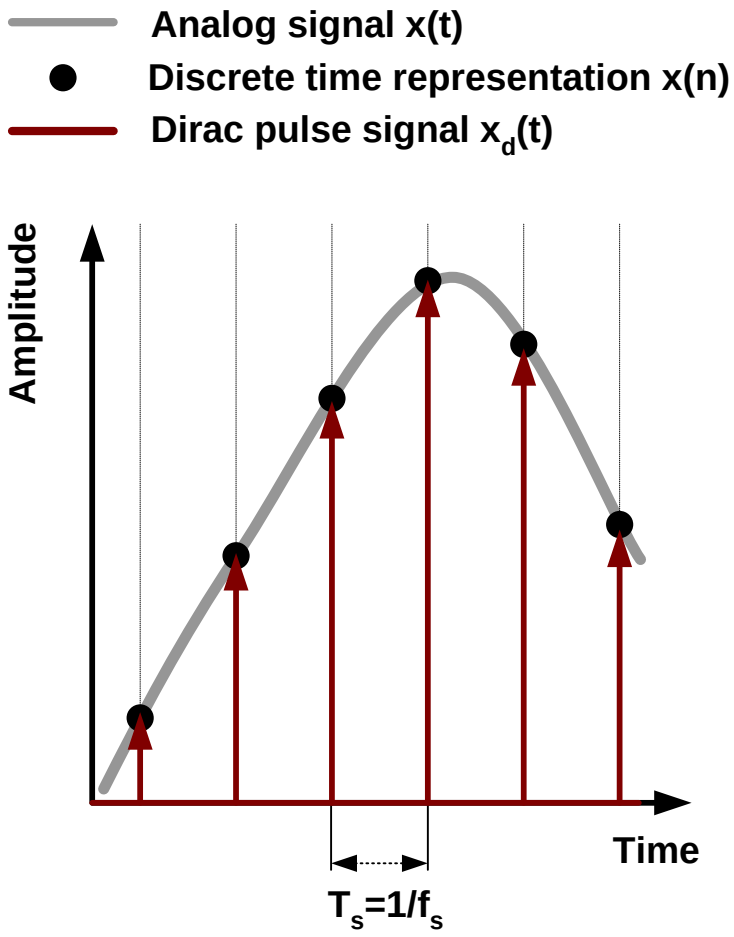
- Not practical
- Question:
  - What happens if each sampling point is replaced with an ideal Dirac impulse?
  - (not very practical, either)

Zero-Order Hold (ZOH) Reconstruction



- The most practical way of reconstructing the continuous time signal is to simply "hold" the discrete time values
  - Either for full period T_s or a fraction thereof
  - Other schemes exist, e.g. first- or partial-order hold [Jha, TCAS II, 11/2008]
- What does this do to the signal spectrum?
- Analyze in two steps
  - First look at infinitely narrow reconstruction pulses

Dirac Pulses



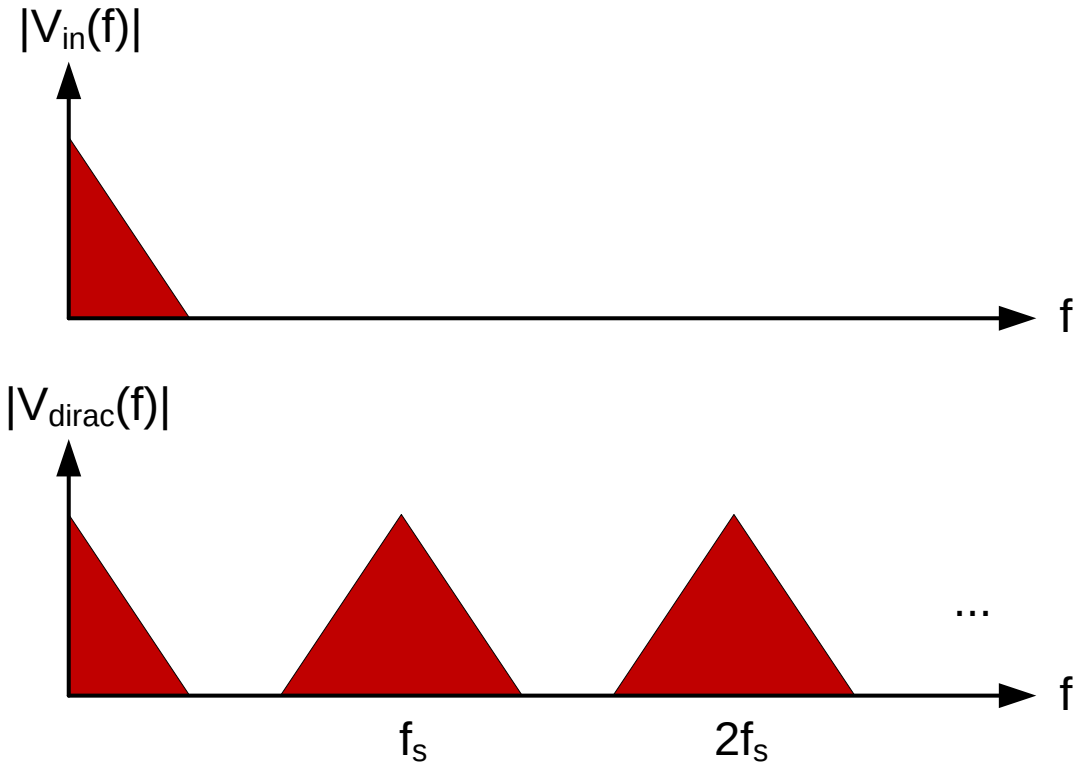
- $x_d(t)$ is zero between pulses
  - Note that x(n) does not exist at these times

$$x_d(t) = x(t) \cdot \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

- Multiplication in time means convolution in frequency
  - Resulting spectrum

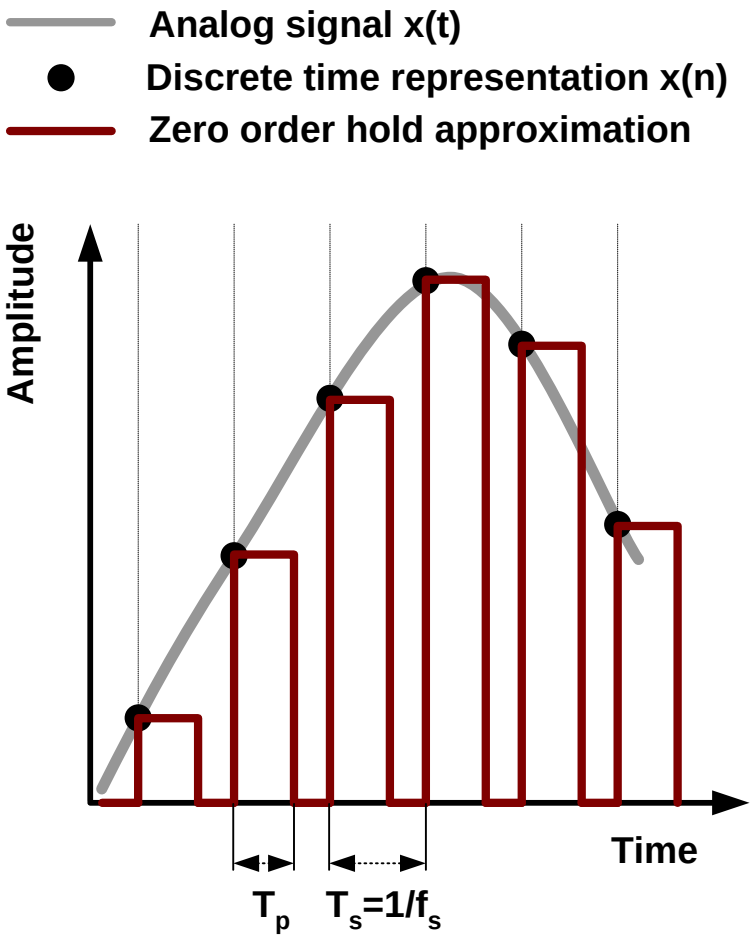
$$X_d(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X\left(f - \frac{n}{T_s}\right)$$

Spectrum for "Dirac Interpolation"



Spectrum of Dirac signal includes replicas of $V_{in}(f)$ at integer multiples of the sampling frequency

Finite Hold Duration



- General case $0 < T_p \leq T_s$
- The time domain signal on the left results from convolving the Dirac sequence with a rectangular unit pulse
- Spectrum: multiplication with Fourier transform of the pulse

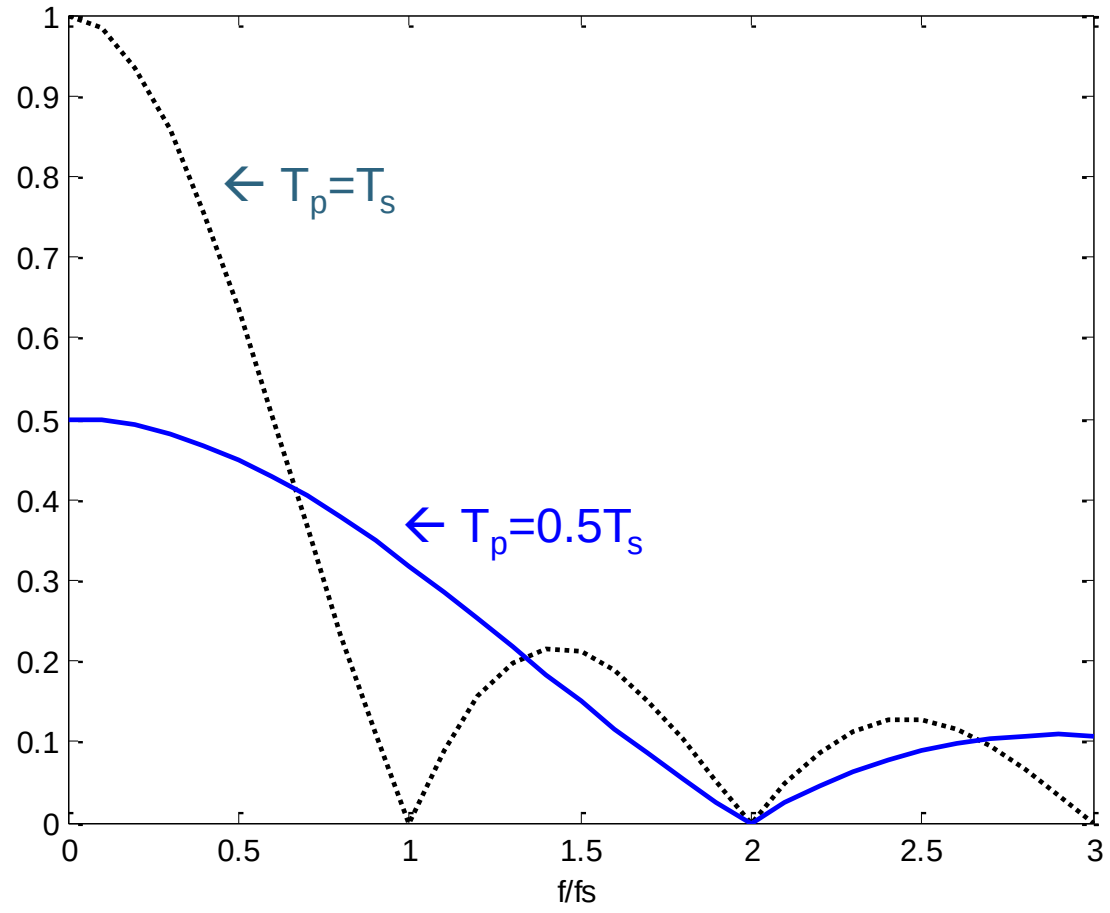
$$H_p(s) = \frac{1 - e^{-sT_p}}{s} \qquad H_p(f) = T_p \frac{\sin(\pi f T_p)}{\pi f T_p} \cdot e^{-j\pi f T_p}$$

"sinc" filter

$$X_p(f) = \underbrace{\frac{T_p}{T_s} \frac{\sin(\pi f T_p)}{\pi f T_p}}_{\text{amplitude envelope}} \cdot e^{-j\pi f T_p} \sum_{n=-\infty}^{\infty} X\left(f - \frac{n}{T_s}\right)$$

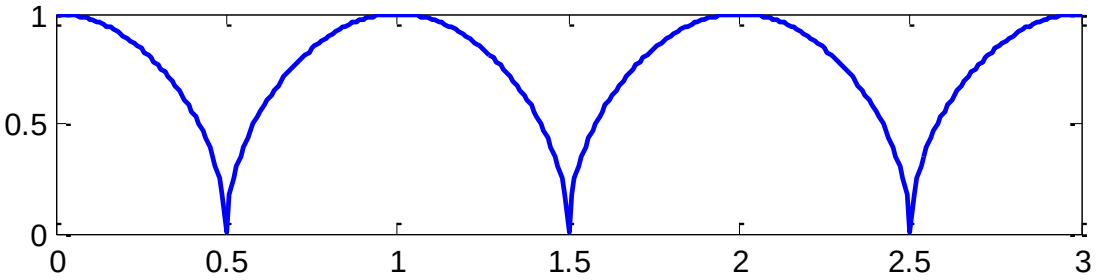
Spectra for $T_p=T_s$ and $T_p=0.5 \cdot T_s$

$$\left| \frac{T_p}{T_s} \frac{\sin(\pi f T_p)}{\pi f T_p} \right|$$



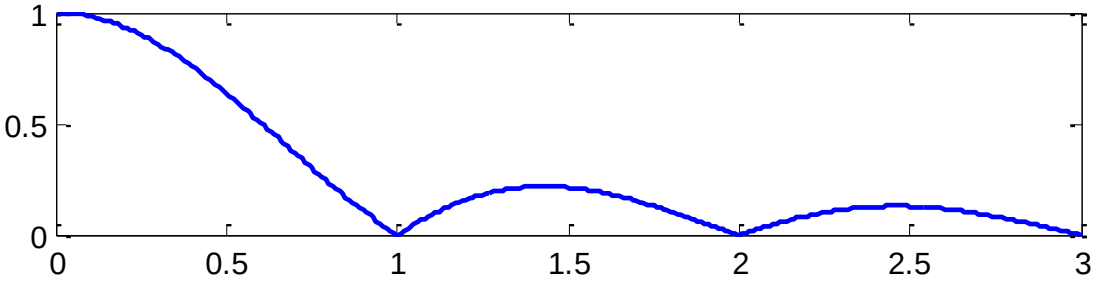
Example

Spectrum of Continuous Time Dirac Pulse Train (Arbitrary Example)

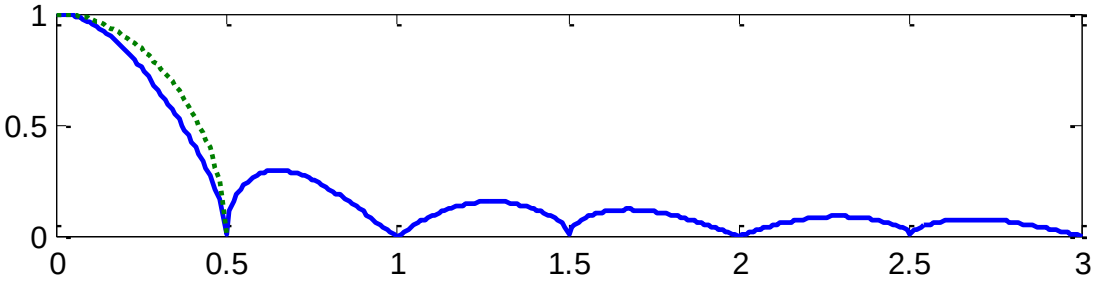


ZOH Transfer Function (sinc "Distortion")

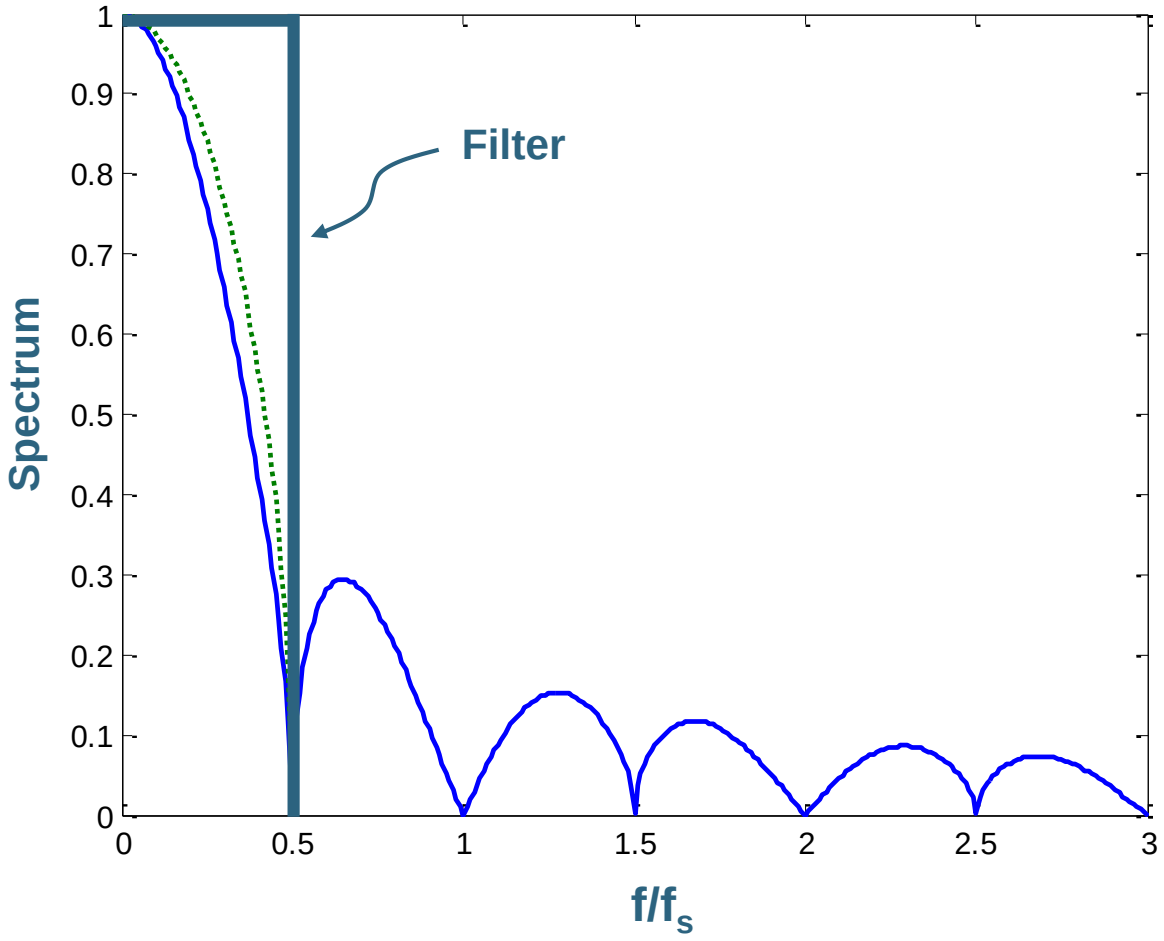
$T_p = T_s$



ZOH output, Spectrum of Staircase Approximation (image rejection & in-band attenuation)



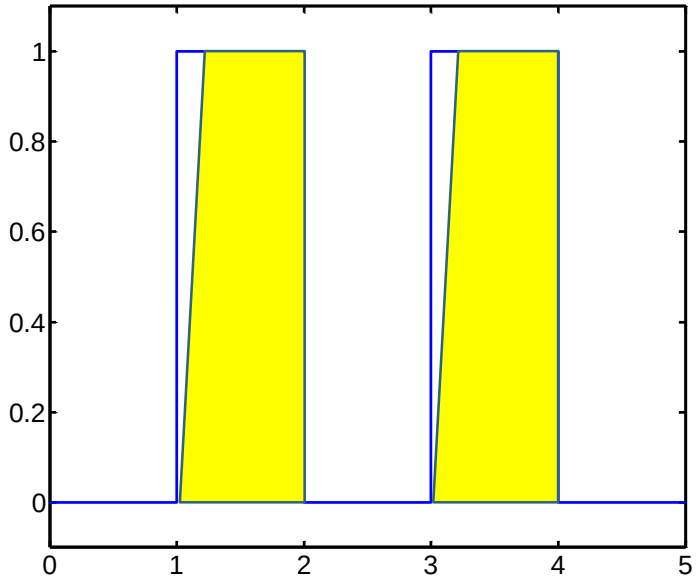
Reconstruction Filter

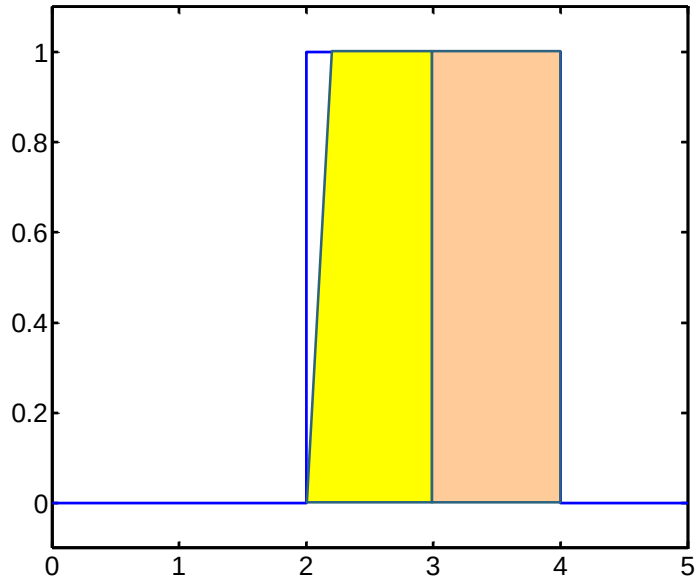


- Also called smoothing filter
- Same situation as with anti-alias filter
  - A brick wall filter with post-emphasis for sinc attenuation would be ideal …
  - What is post-emphasis?
  - Oversampling helps reduce filter order. How?

NRZ versus RZ DAC

1–Bit Example (non return to zero, NRZ):





Area: 1-Δ 1-Δ 1-Δ 1

Total: 2-2Δ 2-Δ

Areas are different for the two cases. RZ coding avoids this.