UiO : University of Oslo

FYS3240- 4240
Data acquisition & control

Signal sampling

Spring 2019 – Lecture #5

Bekkeng, 28.12.2018

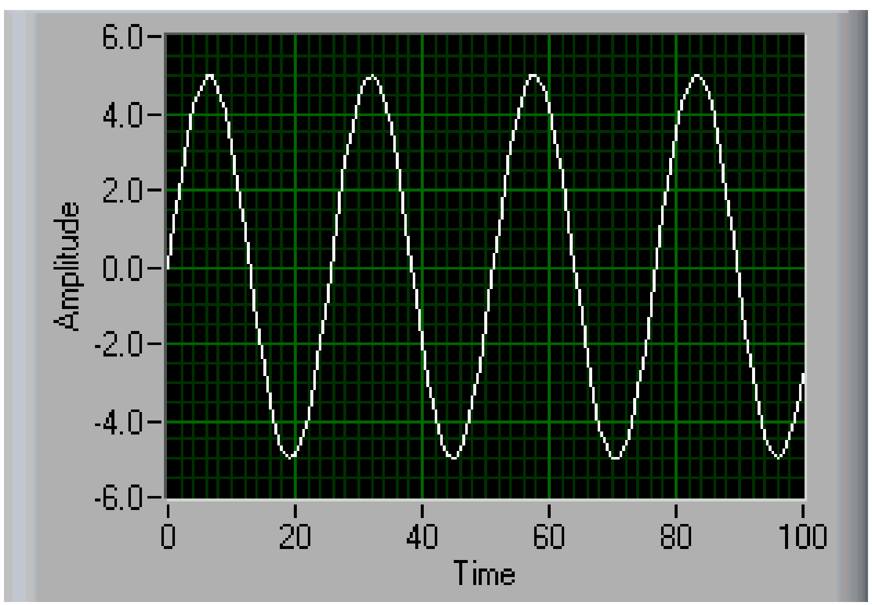
Content

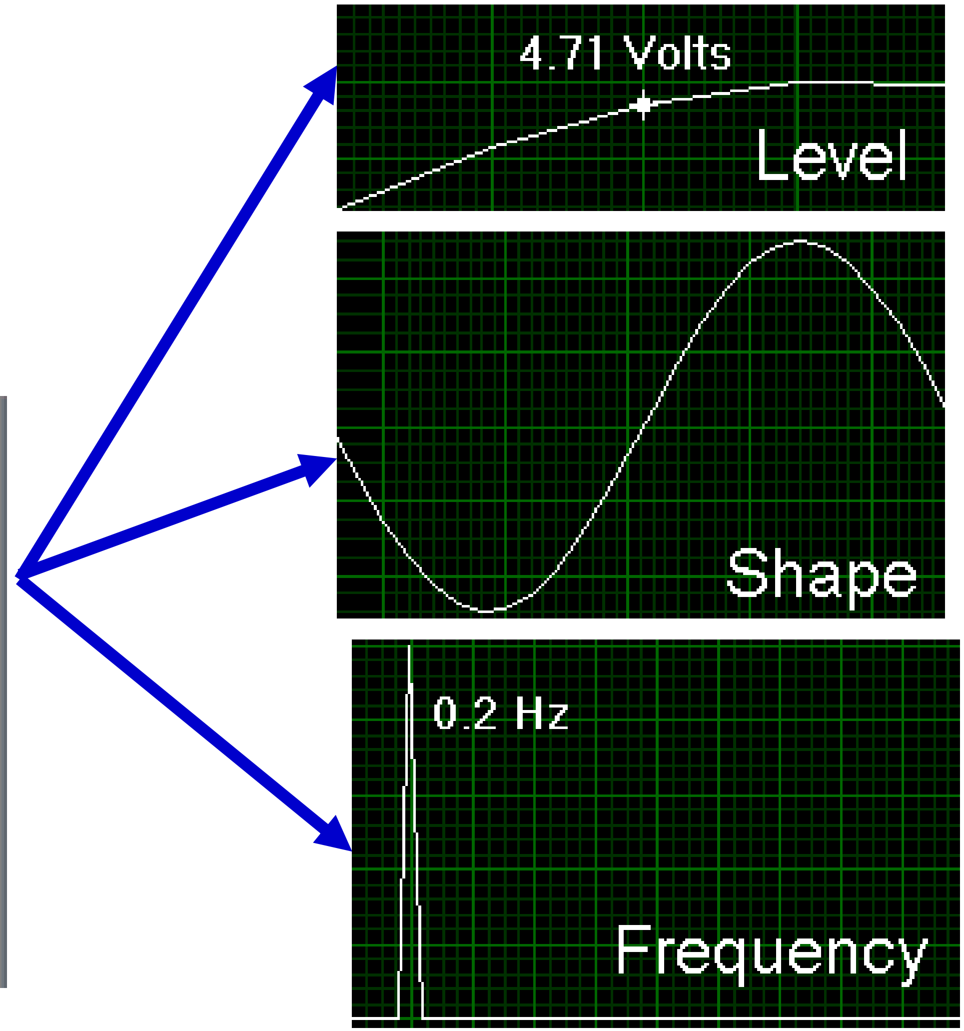
- Aliasing
- Sampling
- Analog to Digital Conversion (ADC)
- Filtering
- Oversampling
- Triggering

Analog Signal Information

Three types of information:

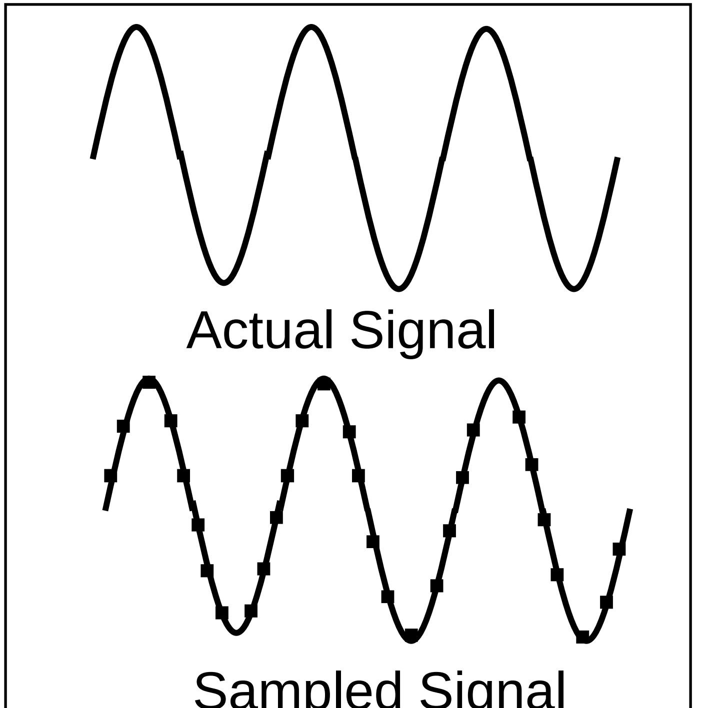
- Level
- Shape
- Frequency





Sampling Considerations

- An analog signal is continuous
- A sampled signal is a series of discrete samples acquired at a specified sampling rate
- The faster we sample the more our sampled signal will look like our actual signal
- If not sampled fast enough a problem known as aliasing will occur



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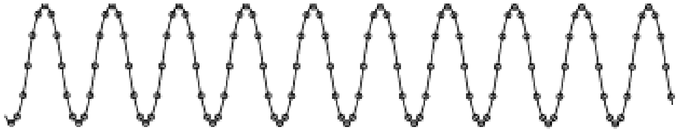
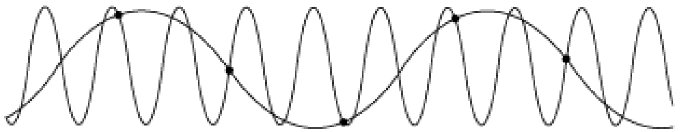
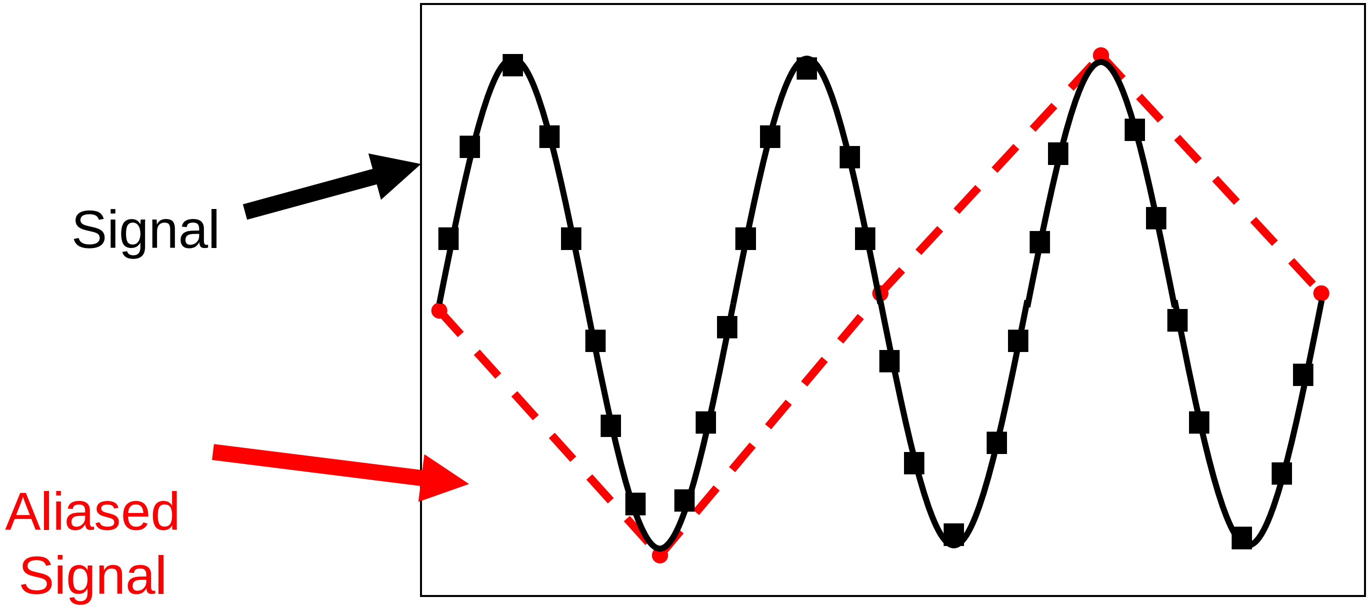

Adequately Sampled Signal

Aliasing

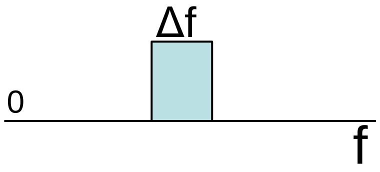

Aliased Signal Due to Undersampling



Aliasing refers to a misrepresentation of the signal frequency due to **undersampling** of the signal.

Sampling & Nyquist's Theorem

- **Nyquist's sampling theorem:**
  - **The sample frequency should be at least twice the highest frequency contained in the signal**
- Or, more correctly: The sample frequency f_s should be at least twice the bandwidth Δf of your signal
  - In mathematical terms: $f_s \geq 2 * \Delta f$, where $\Delta f = f_{high} - f_{low}$



- However, to accurately represent the shape of the signal, or to determine peak maximum and peak locations, a higher sampling rate is required
  - Typically a sample rate of 10 times the bandwidth of the signal is required.

ECG signal

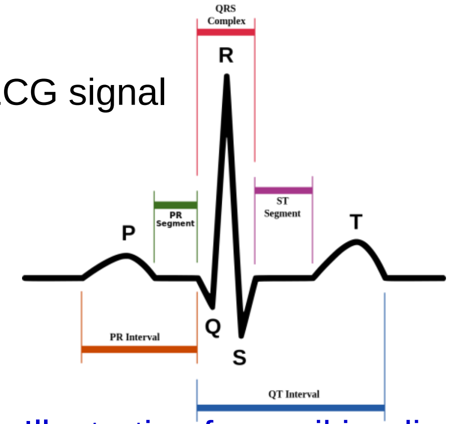
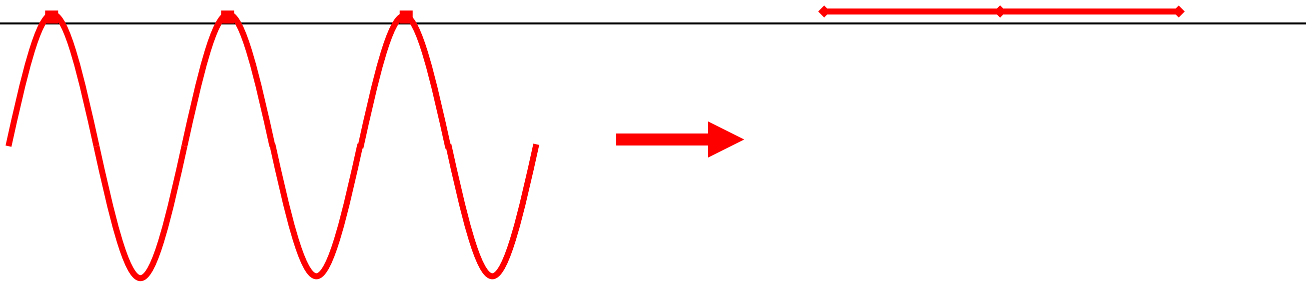
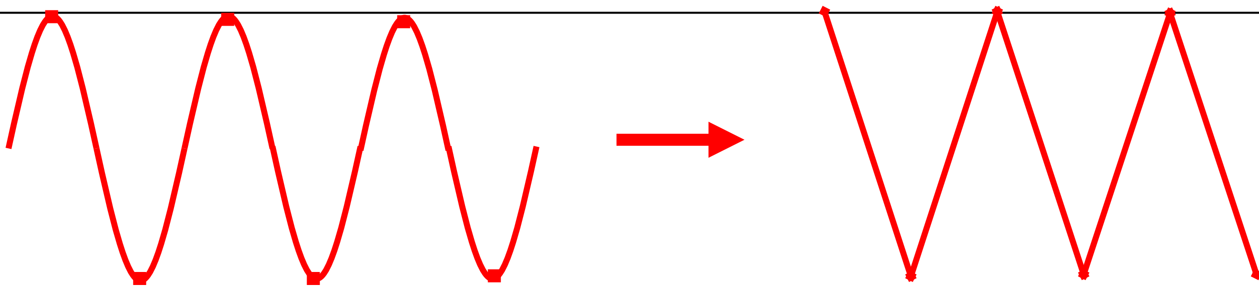
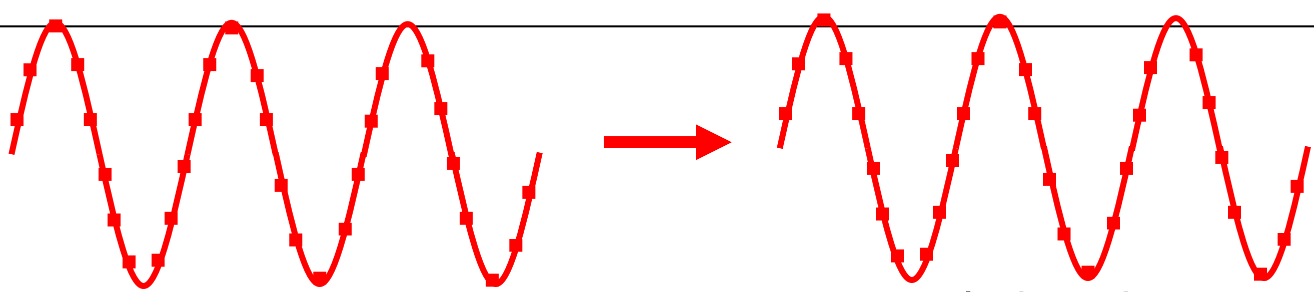
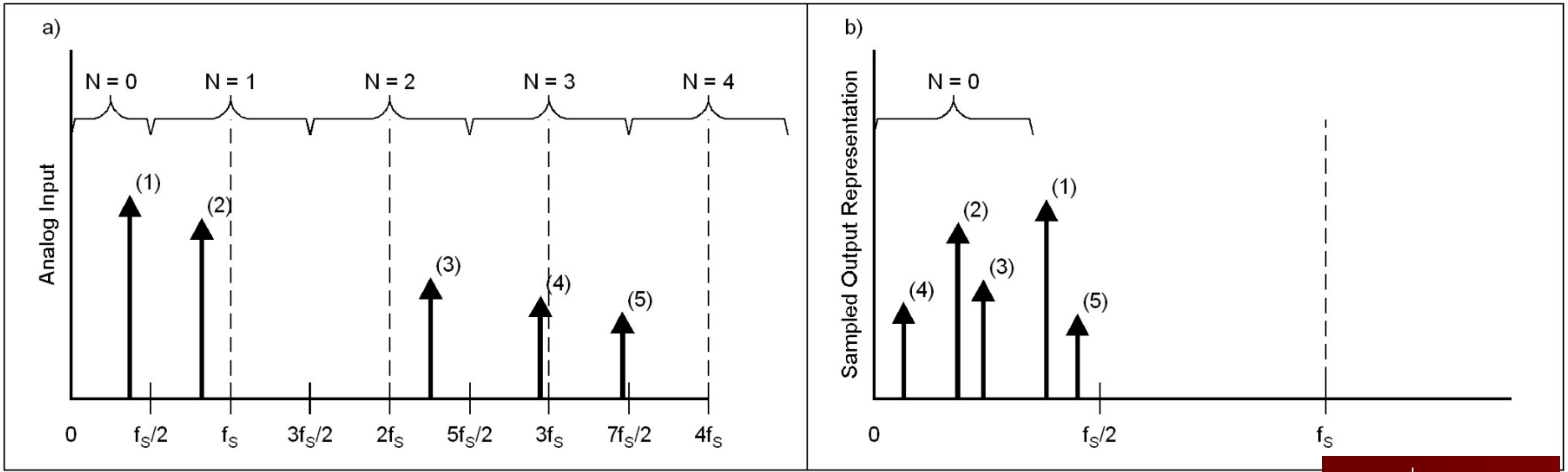


Illustration from wikipedia

Sampling Examples

 100Hz Sine Wave — Sampled at 100Hz	Aliased Signal
 100Hz Sine Wave — Sampled at 200Hz	Adequately Sampled for Frequency Only (Same # of cycles)
 100Hz Sine Wave — Sampled at 1kHz	Adequately Sampled for Frequency and Shape

Aliasing shown in the frequency domain



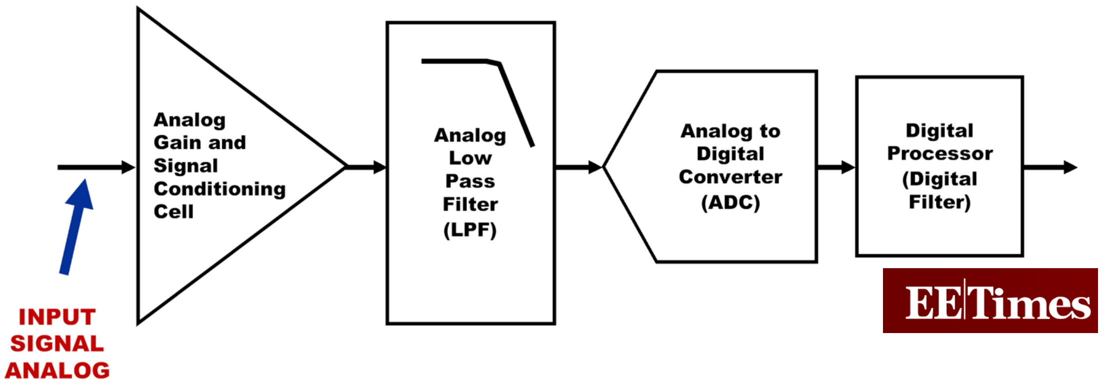
EETimes

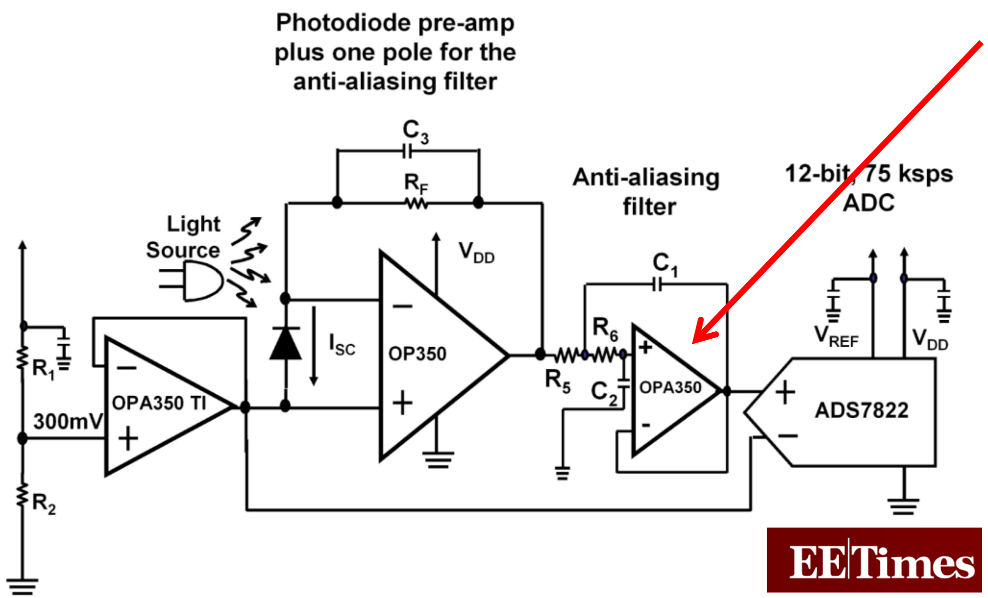
$f_{ALIASED} = |f_{IN} - Nf_S|$

A system that has a sampling frequency f_s (a) will digitize signals with frequencies below $f_s/2$ as well as above. Input signals below $f_s/2$ will be reliably digitized while signals above fs/2 will be folded back (b) and appear as lower frequencies in the digital output according to $f_{aliased} = |f_{in} - N*fs|$

Need to remove all signal frequencies above fs/2 using an analog low-pass filter before the sampling in the ADC

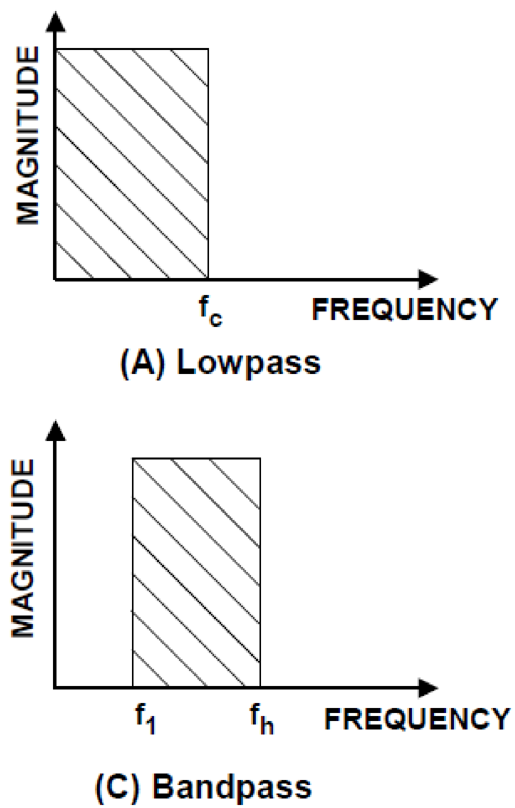
Example of DAQ with anti-aliasing filter



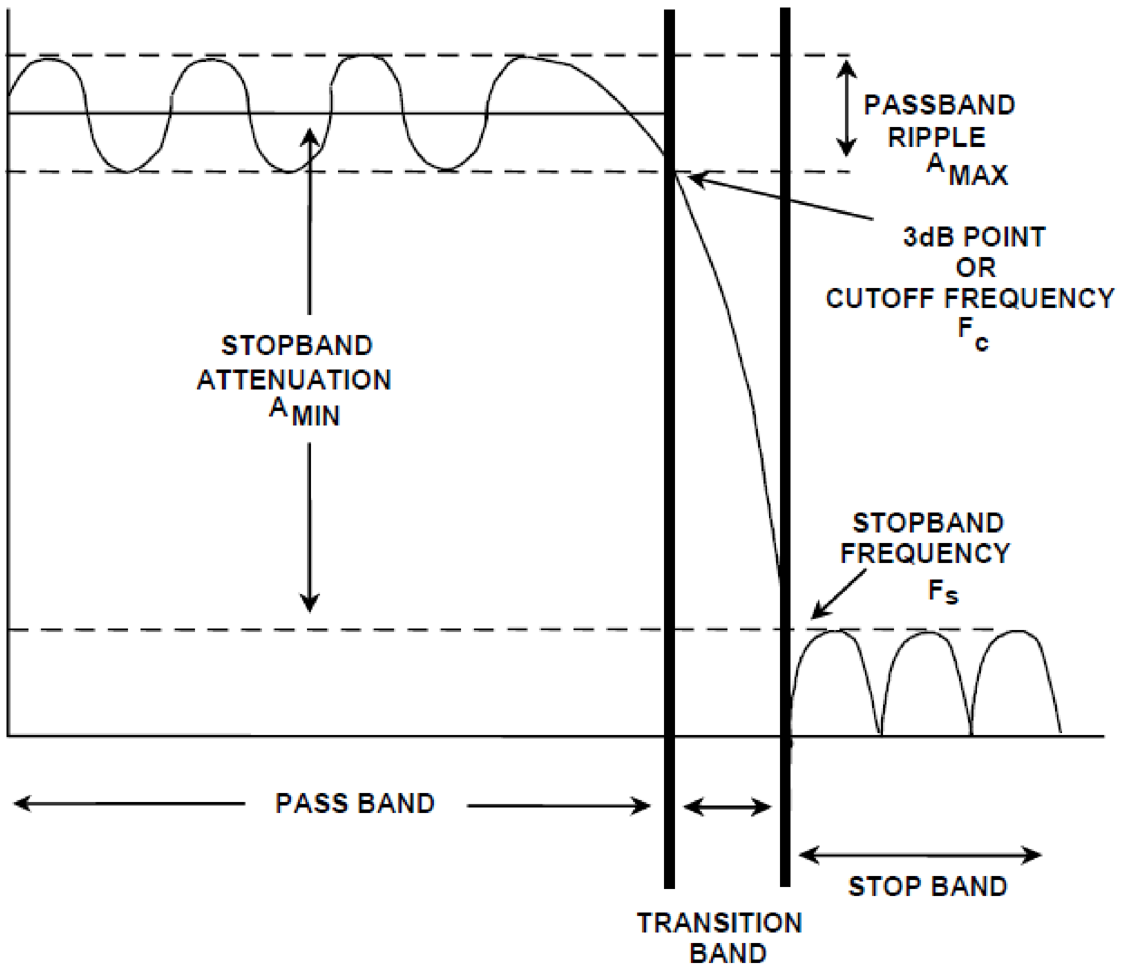


The required filter order is determined by the number of bits in the ADC and the DAQ system specification

Idealized Filter Responses

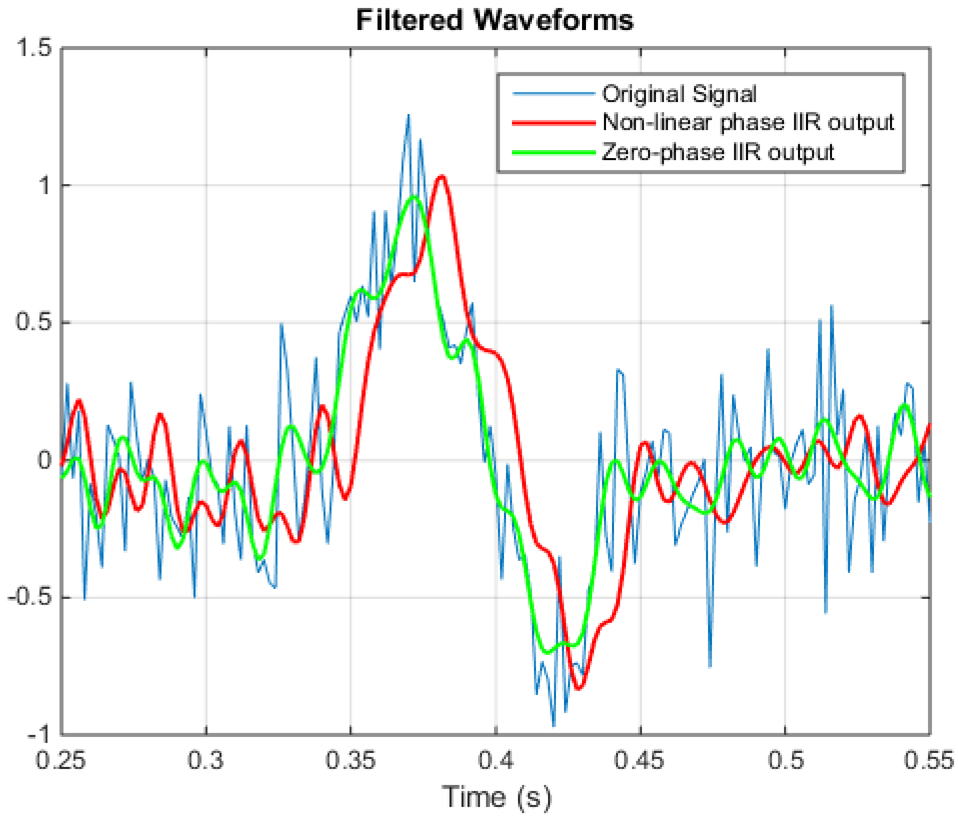


Filter parameters



A filter will affect the phase of a signal, as well as the amplitude!

Filtering example

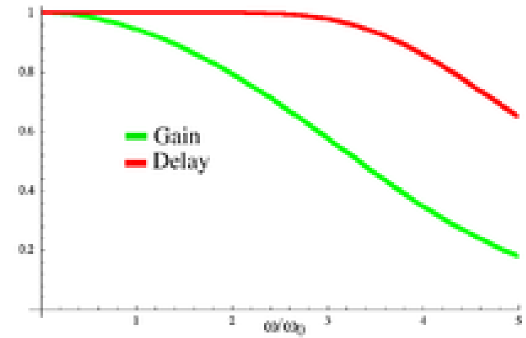


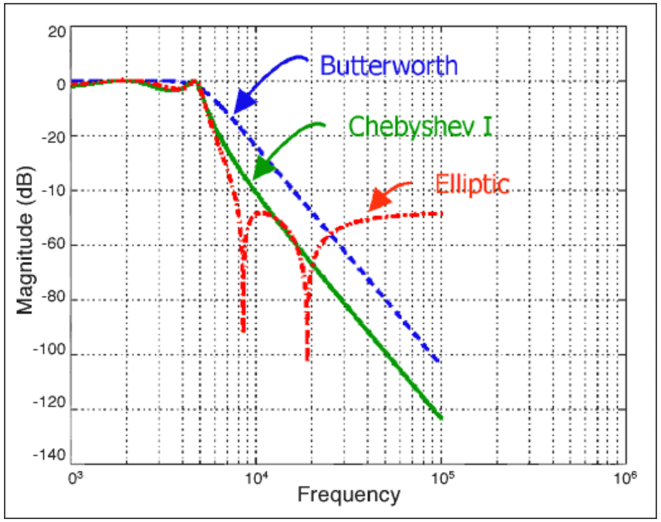
Example from MathWorks

In post-processing (non-real time) a **zero-phase digital filter** can be used, by processing the input data in both the forward and reverse directions

Analog filters

Bessel

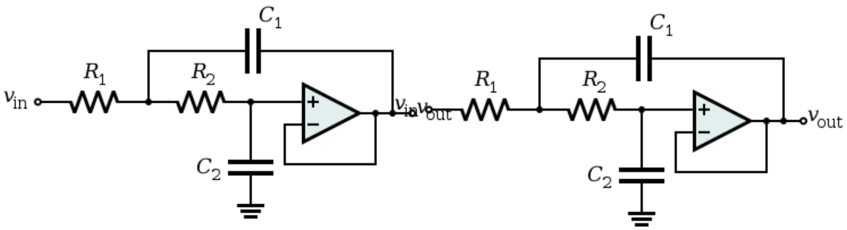




- Some common filter characteristics
  - **Butterworth** (no rippel)
  - **Chebyshev**
  - **Bessel** (constant group delay = linear phase in pass band)
  - **Elliptic**

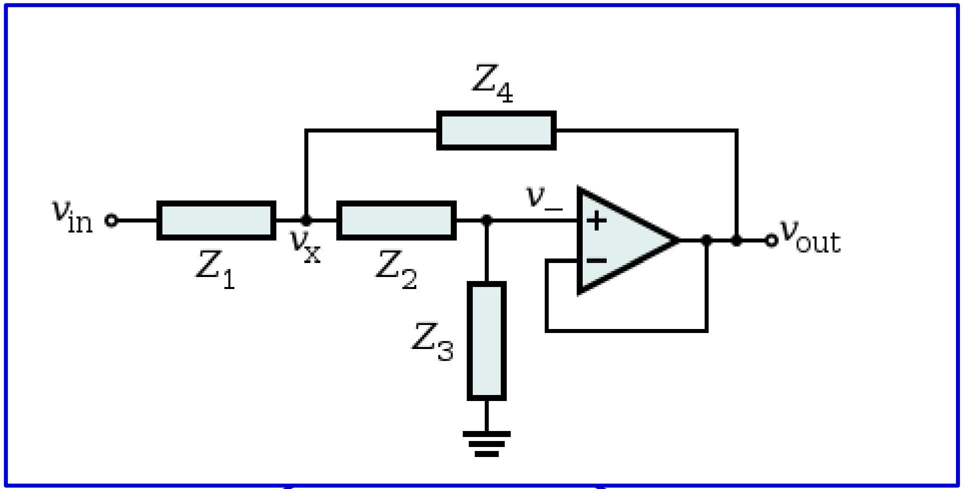
- Select filter characteristics according to DAQ system specification /requirements

- Analog filters can be made using a Sallen-Key architecture (see next slide)
  - Multiple 2. order elements can be connected together to create a n. order filter.

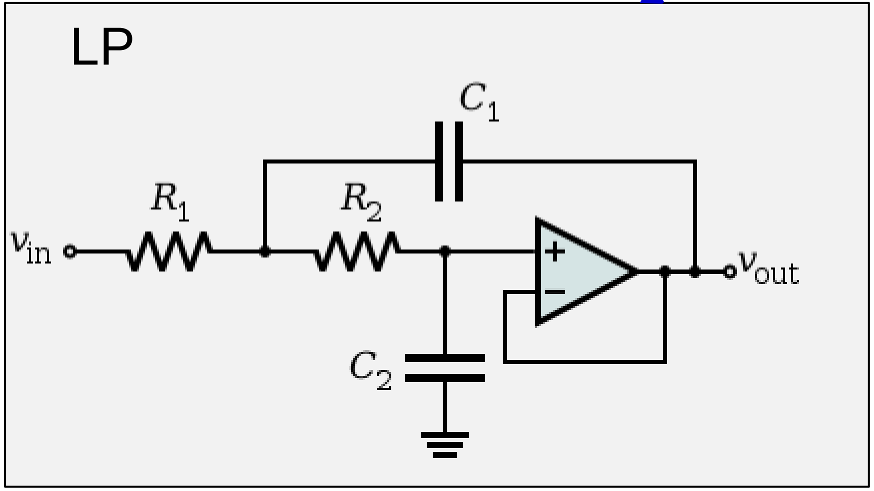


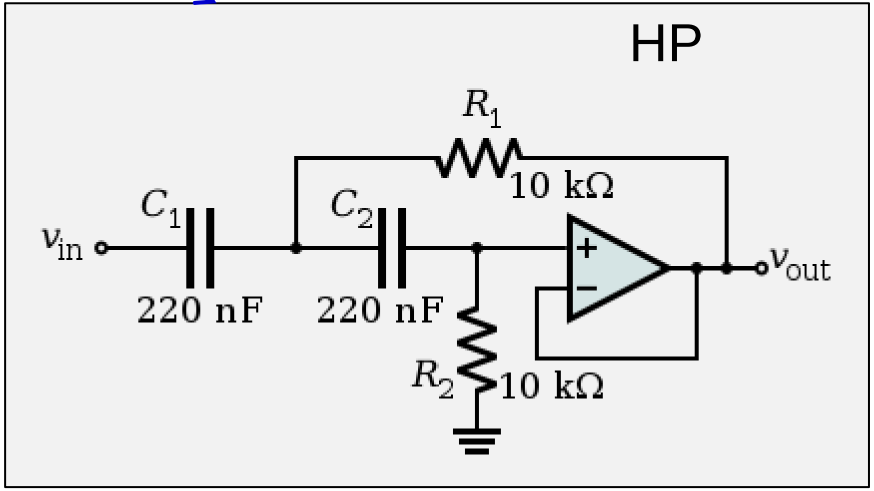
2. order Sallen-Key - Active analog filter

Structure



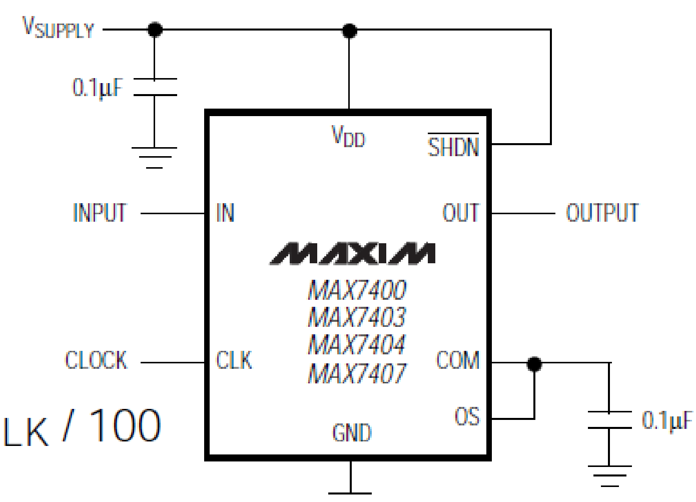
WEBENCH



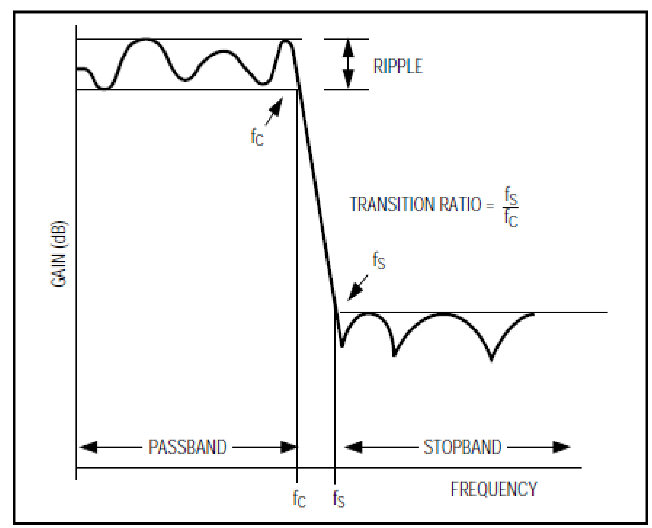


Switched-Capacitor Filter

- Can be suitable as an ADC anti-aliasing filter if you build your own electronics
- Be aware of possible clock noise (add RC-filters before and after)
- The corner frequency (cut-off) fc is "programmable" using an external clock
- Example:
  - MAX7400 8th-order,lowpass, elliptic filter
  - MAX7400 has a transition ratio (fs/fc) of 1.5 and a typical stop band rejection of 82dB

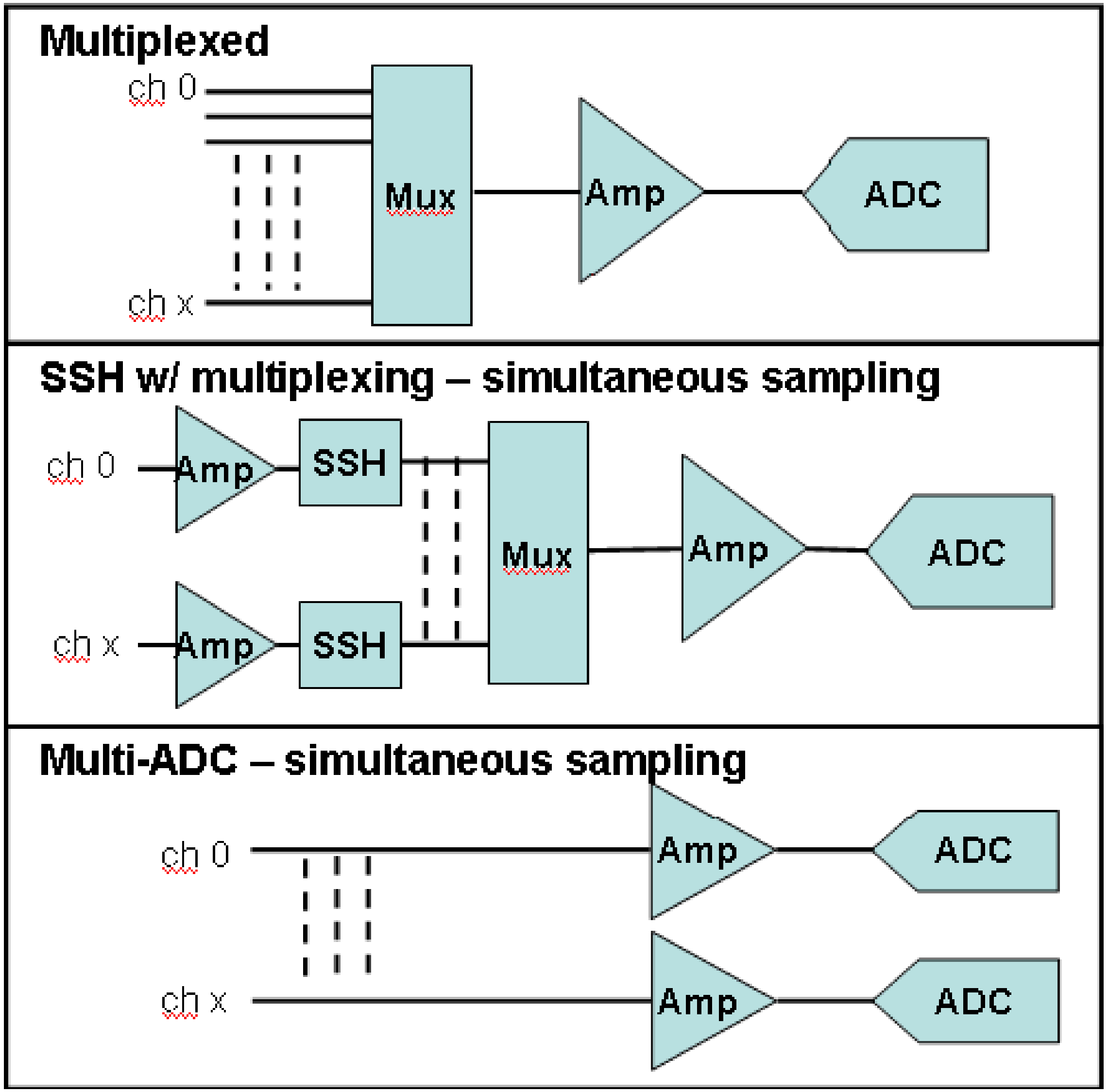


$f_C = f_{CLK} / 100$



ADC architectures

- **Multiplexed** sampling
  - Gives a time delay between channel sampling
- **Simultaneous** sampling
  - One ADC, multiple Sample-and-Hold registers
  - Multiple ADCs
  - Important for phase measurements



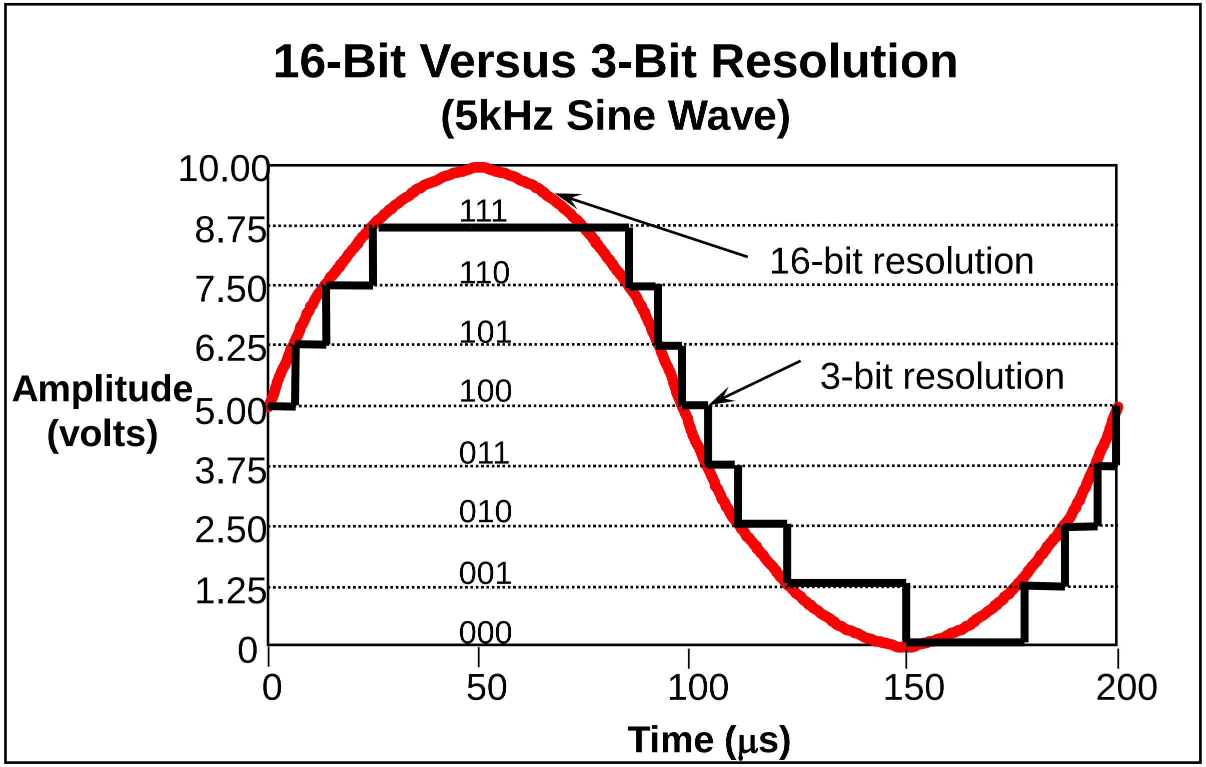
ADC resolution

- The number of bits used to represent an analog signal determines the resolution of the ADC
- Larger resolution = more precise representation of your signal
- The resolution determine the smallest detectable change in the input signal, referred to as code width or LSB (least significant bit)

$$code\ width = \frac{device\ range}{2^{resolution}}$$

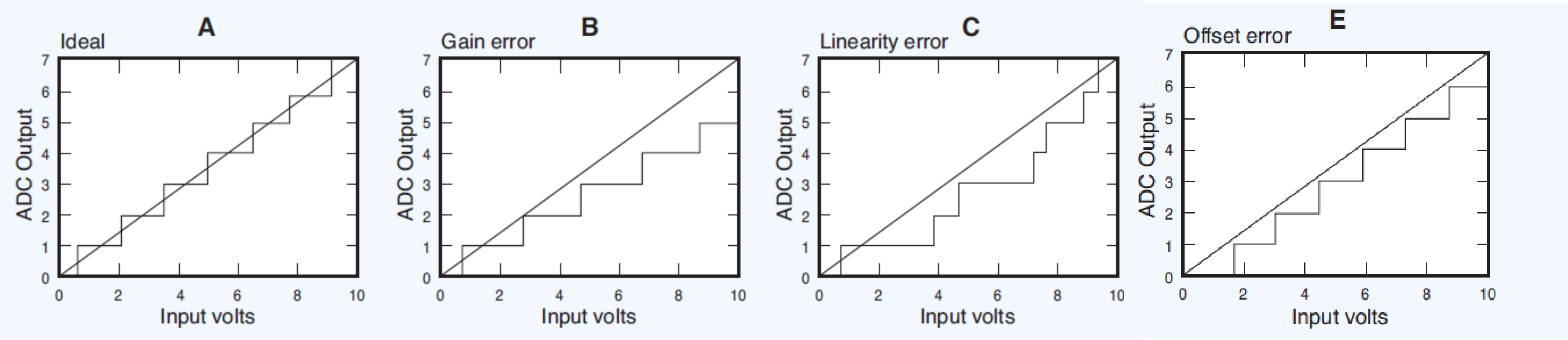
Example:

$$\frac{device\ range}{2^{resolution}} = \frac{10}{2^{16}} = .15\ mV$$



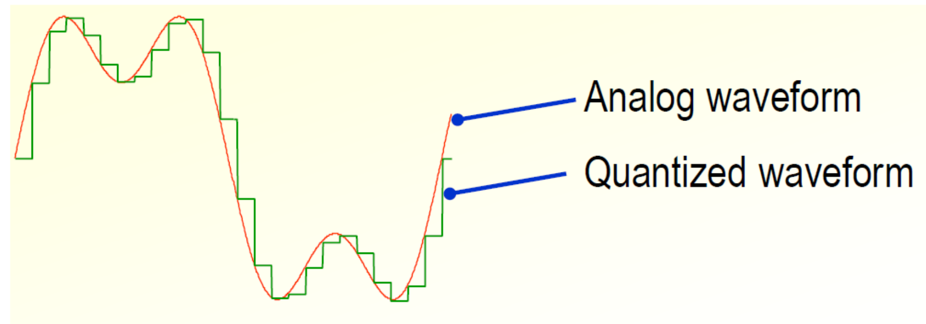
ADC accuracy

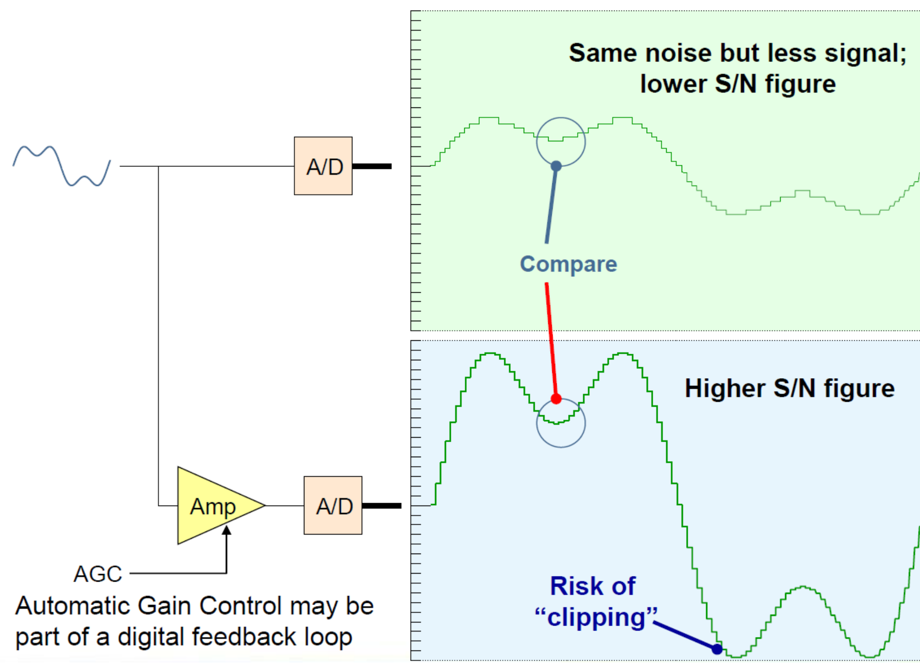
- Common ADC errors:
  - Noise
  - Linearity error
  - Gain error
  - Offset error
  - Quantization (resolution error)
    - Less than LSB/2



Digital signals: Bits, dynamic range, and SNR

- SNR = signal to noise ratio
- The number of bits used determines the maximum possible signal-to-noise ratio
- Using the entire ADC range (using an amplifier) increases the SNR
- The minimum possible noise level is the error caused by the quantization of the signal, referred to as **quantization noise.**





ADC oversampling

- Oversampling means to sample faster than the Nyquist rate $f_{nyquist}$, which is given by $f_{nyquist} = 2 * \Delta f$, where $\Delta f = f_{max} - f_{min}$

- The SNR of an ideal N-bit ADC (due to quantization effects) is:

$$SNR(dB) = 6.02*N + 1.76$$

ADC oversampling II

- If the sampling rate f_s is increased above $f_{nyquist}$, we get the following SNR: $SNR(dB) = 6.02*N + 1.76 + 10*\log_{10}(OSR)$, where $OSR = f_s/f_{nyquist}$
- Oversampling makes it possible to use <u>a simple RC anti-aliasing filter</u> before the ADC
- After A/D conversion, perform digital low-pass filtering and then down sampling to $f_{nyquist}$
- Effective resolution with oversampling $N_{eff} = N + 1/2 *\log_2 (f_s/f_{nyquist})$, where N is the resolution of an ideal N-bit ADC at the Nyquist rate
  - If $OSR = f_s/f_{nyquist} = 1024$, an 8-bit ADC gets and effective resolution equal to that of a 13-bit ACD at the Nyquist rate

Trigger (from hardware or software)

- A trigger is a signal that causes a device to perform an action, such as starting a data acquisition. You can program your DAQ device to react on triggers such as:
  - a software command (software trigger)
  - a condition on an external digital signal
  - a condition on an external analog signal
    - E.g. level triggering

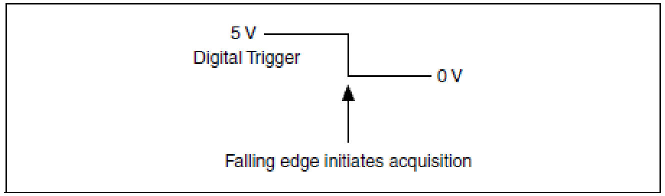


Figure 13-1. Falling-Edge Trigger

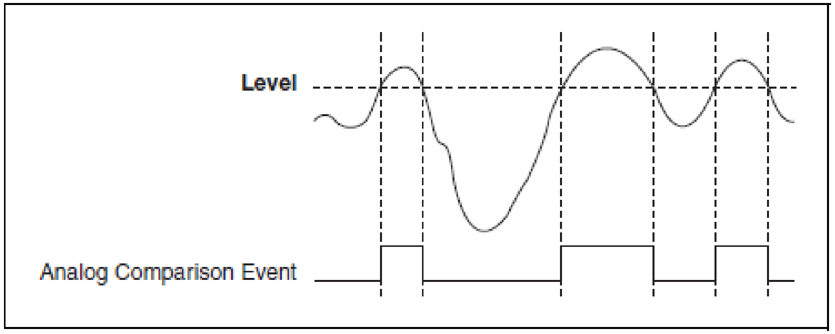


Figure 13-4. Above-Level Analog Triggering Mode

Important trigger types

- **Start trigger**
  - start data acquisition when an external digital signal have e.g. a rising edge.

- **Pre-trigger**
  - Uses a data buffer (circular buffer)
    - Can include a specified number of samples before the trigger event.
    - Useful for e.g. high speed imaging.