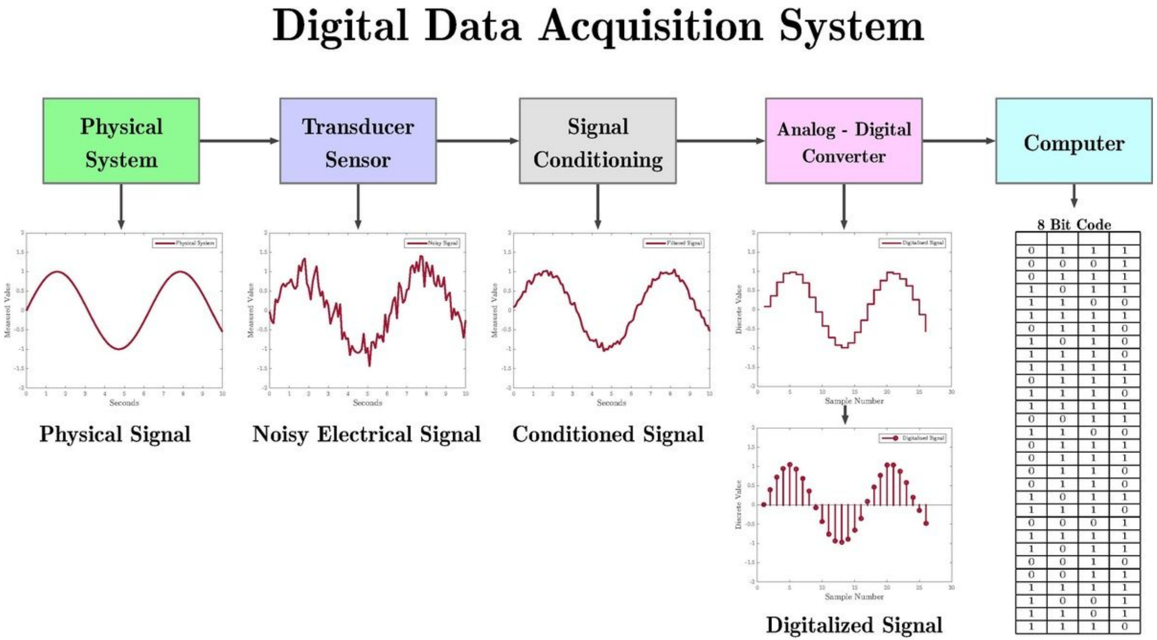


# Making a DAQ System that Makes Sense

John R. Leeman

GEARS 2022

LEEMANGEOPHYSICAL LLC
CONSULTING & INSTRUMENTATION

Image: WikiPedia

# DAQ, ADC, DAC, and more alphabet soup

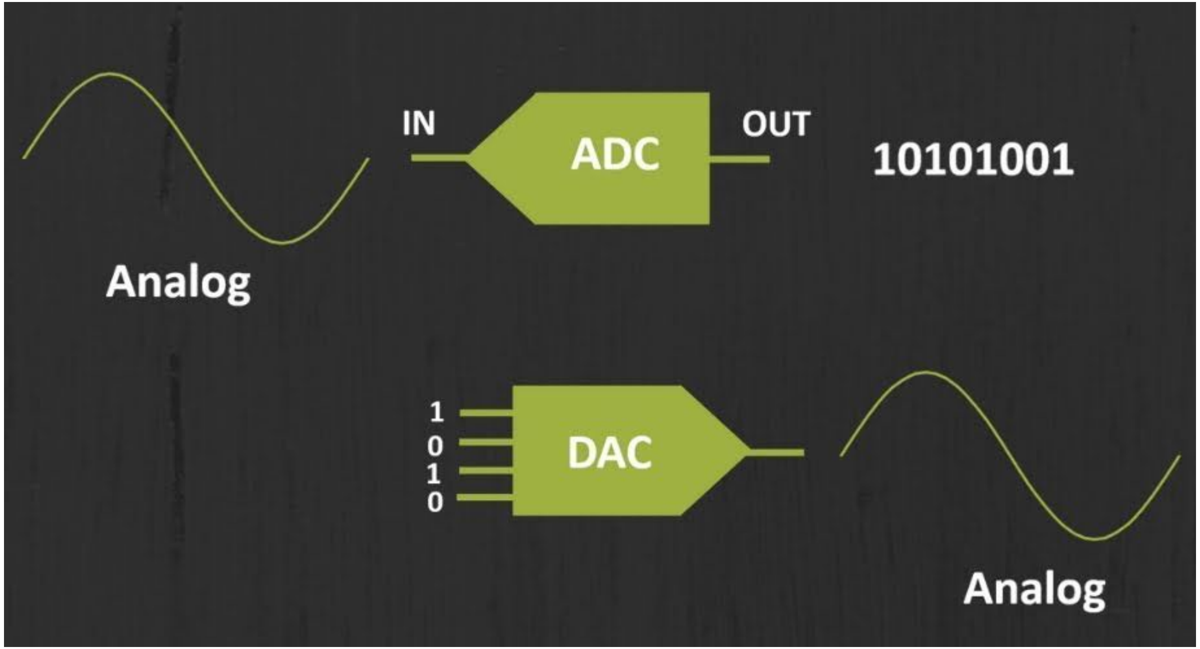


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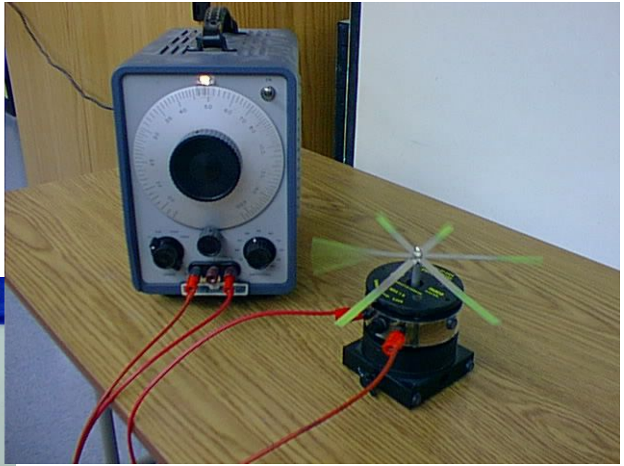
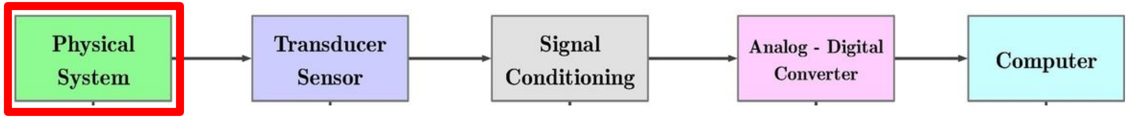
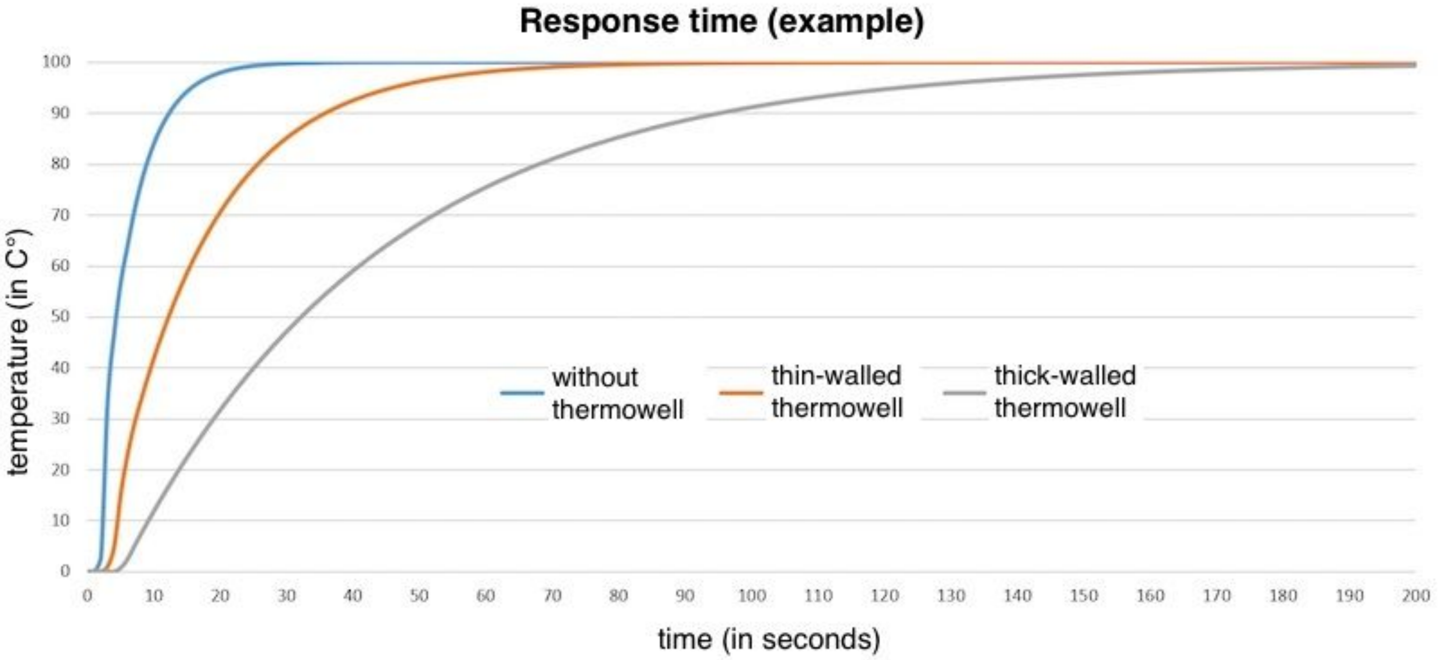
Image: All About Electronics

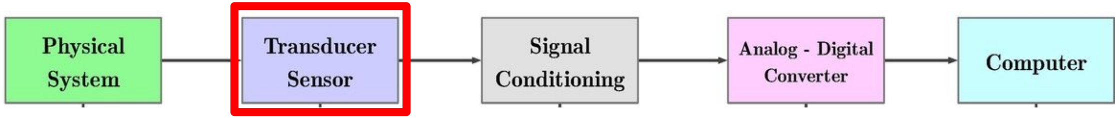
# What you’re measuring is likely a filter in itself



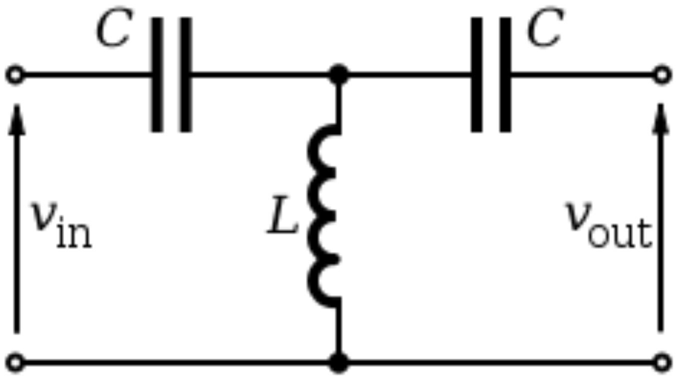
# The transducer also certainly has some response function



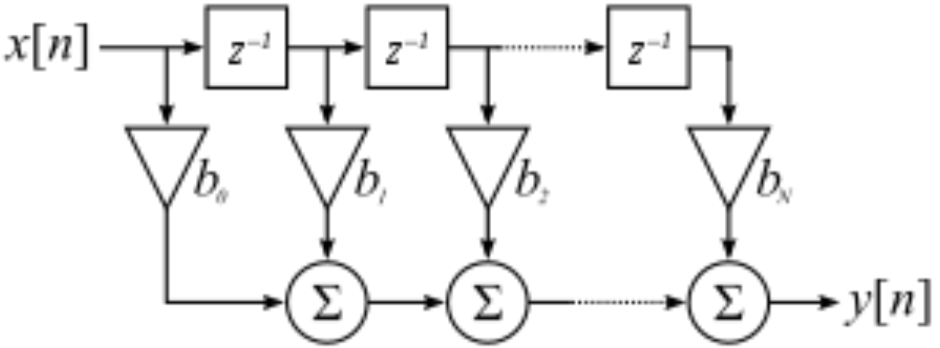
Signal conditioning is amplification and filtering of the signal to get the best SNR possible

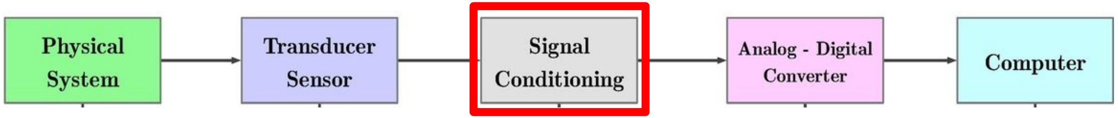
## Analog Filters




## Digital Filters

Image: WikiPedia

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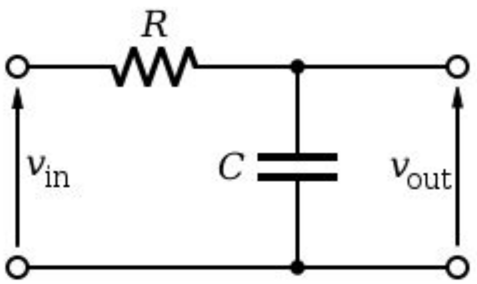
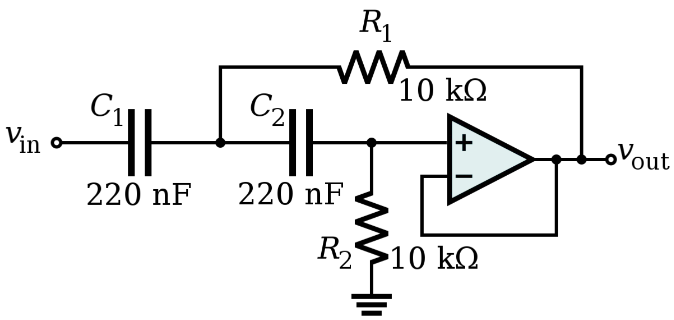
# Filters and classed into two main categories with a third “fake” class

## Passive




## Active




## ASIC

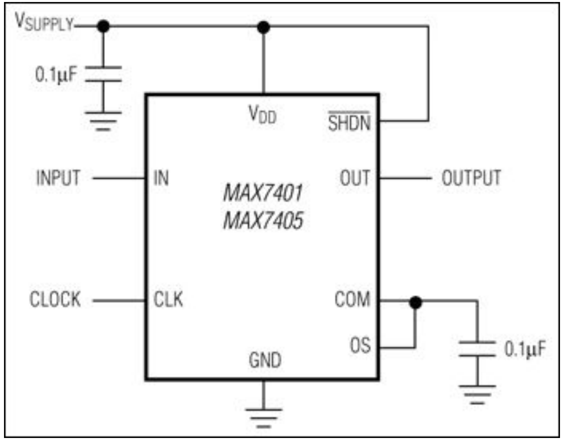
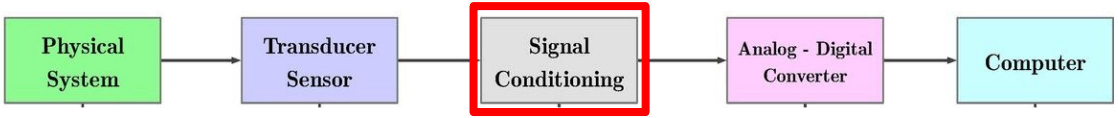



Image: WikiPedia, Maxim Integrated



It is helpful to remember the basic types of filters that we generally employ though

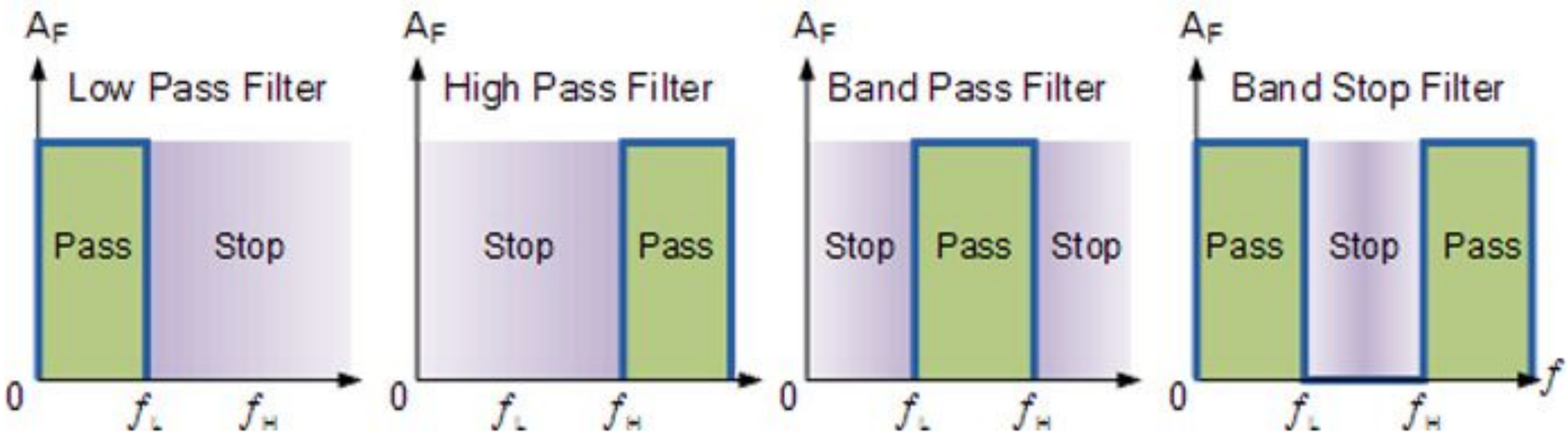



Image: Electronics Tutorials

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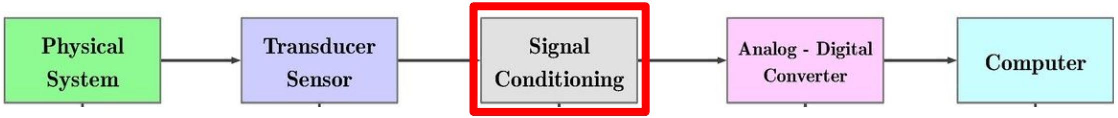
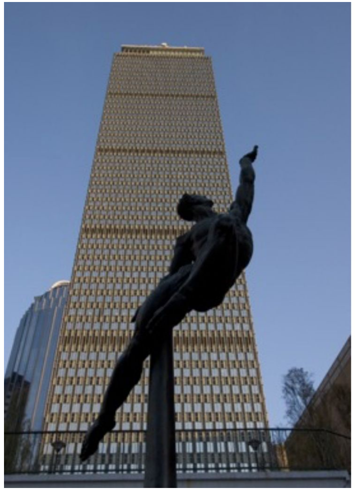


# The most common filter you'll apply is the anti-aliasing filter



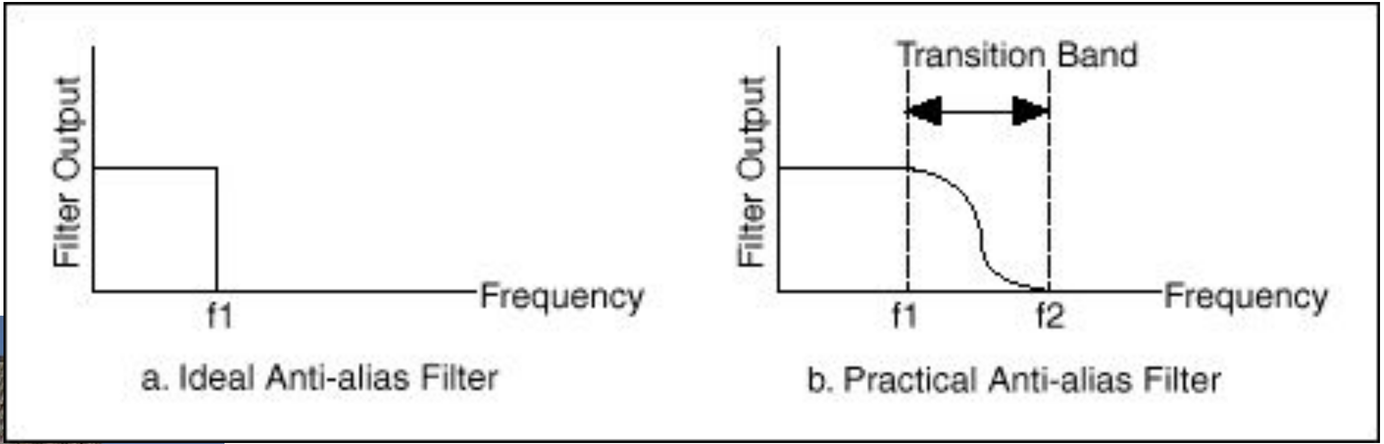
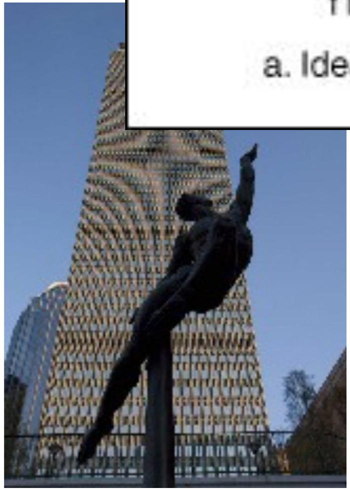
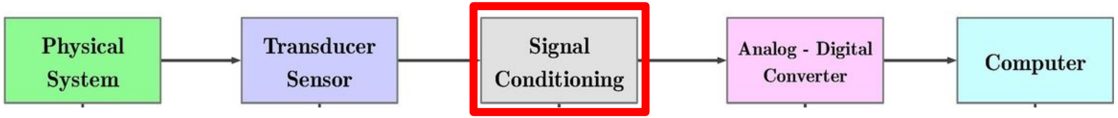



Image: National Instruments, mrpc





# Analog to Digital Converters (ADCs) take a analog (continuous) signal and create a digital representation of it

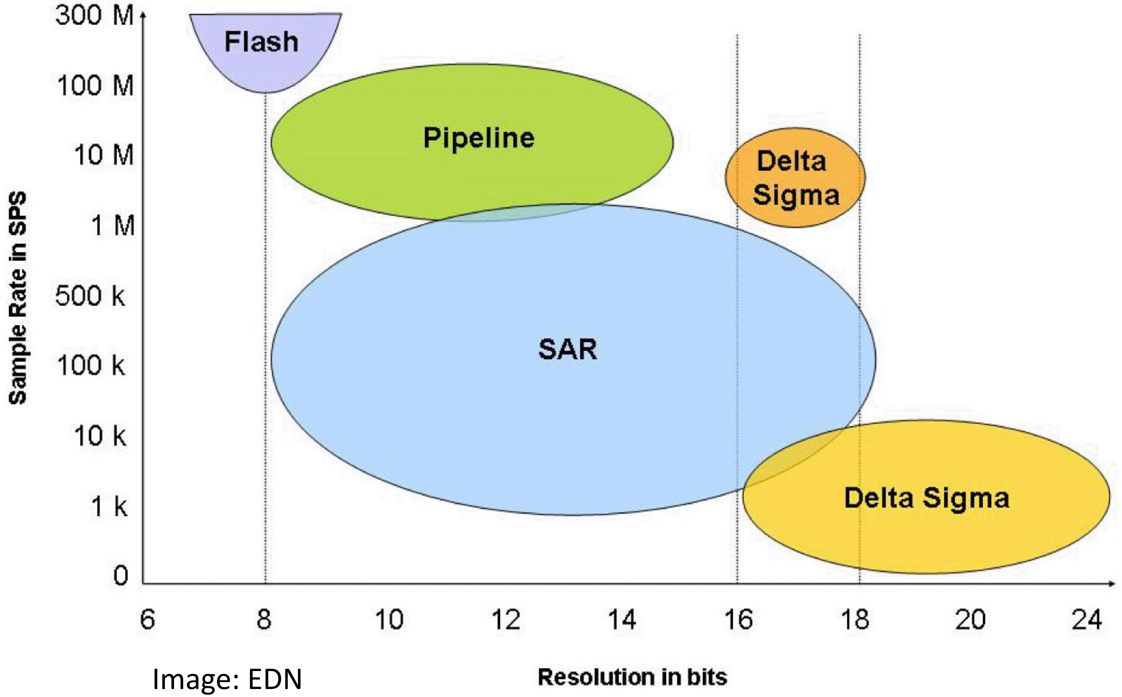



Image: EDN

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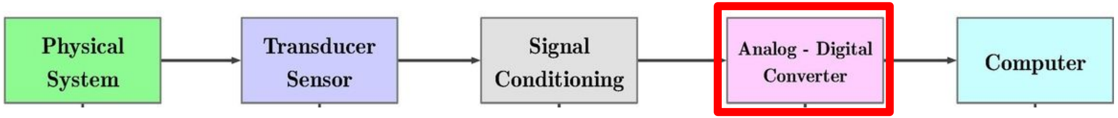
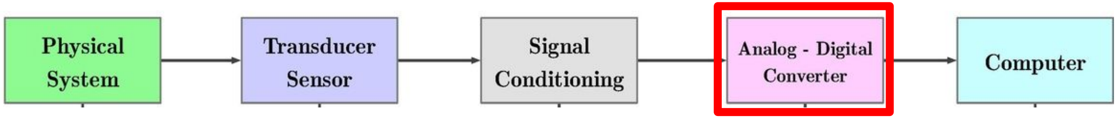


# Let’s look at some of the effects of ADC bit depth and sampling rate

$$\text{volts per bit} = \frac{\text{range}}{2^n - 1}$$

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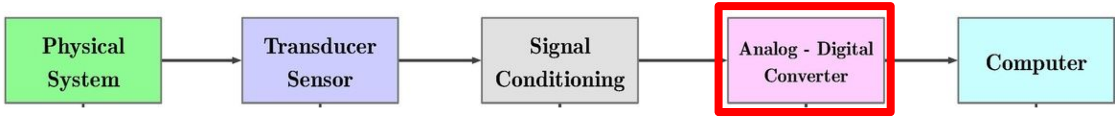
# We often look at the SNR and ENOB

$$\mathrm{SNR}_{\mathrm{qe}} = (6.02N_{\mathrm{bits}}) + 1.76\mathrm{dB}$$

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

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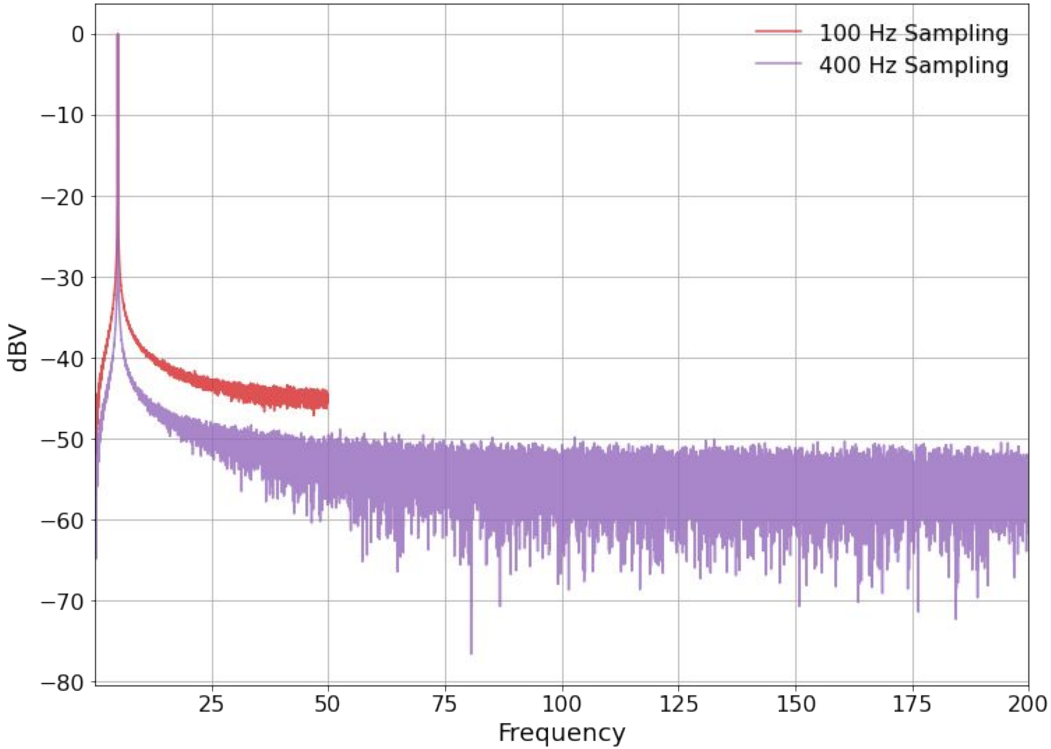
# Oversampling is an effective strategy to increase the effective number of bits

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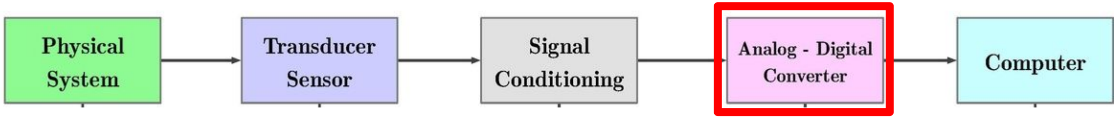
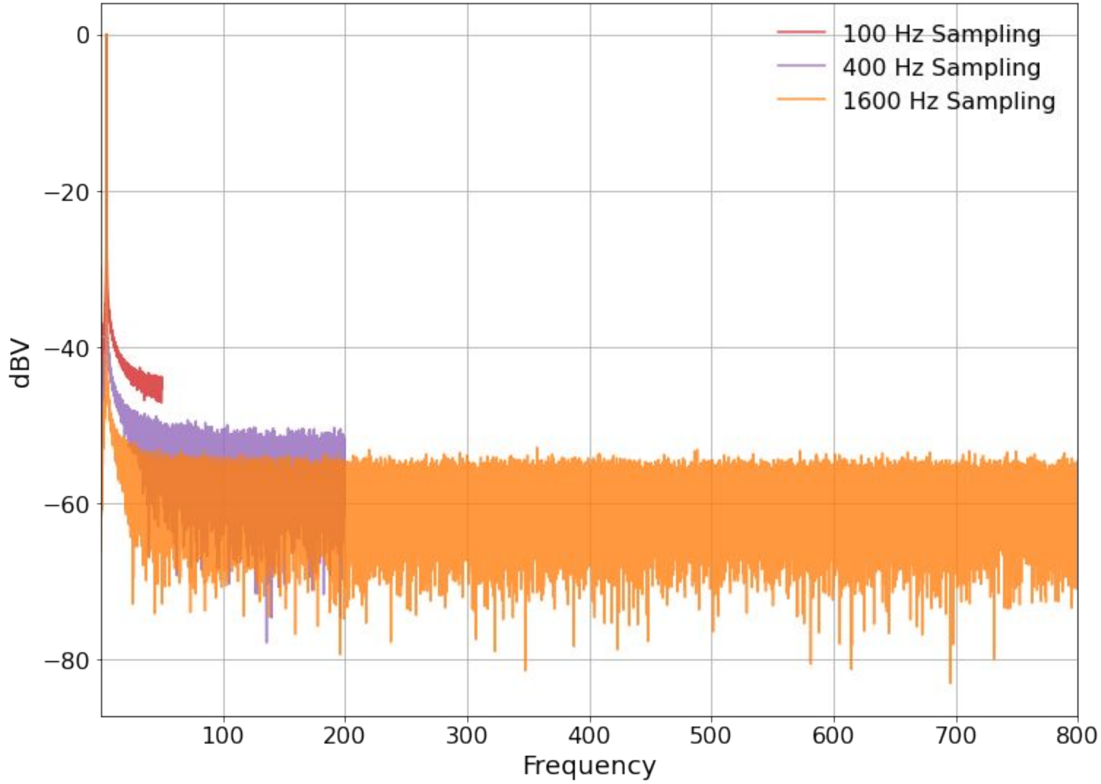
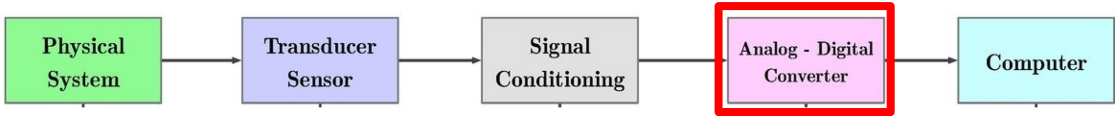


# Oversampling is an effective strategy to increase the effective number of bits

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# Noise shaping is another way many ADCs reduce the noise you see in your signal

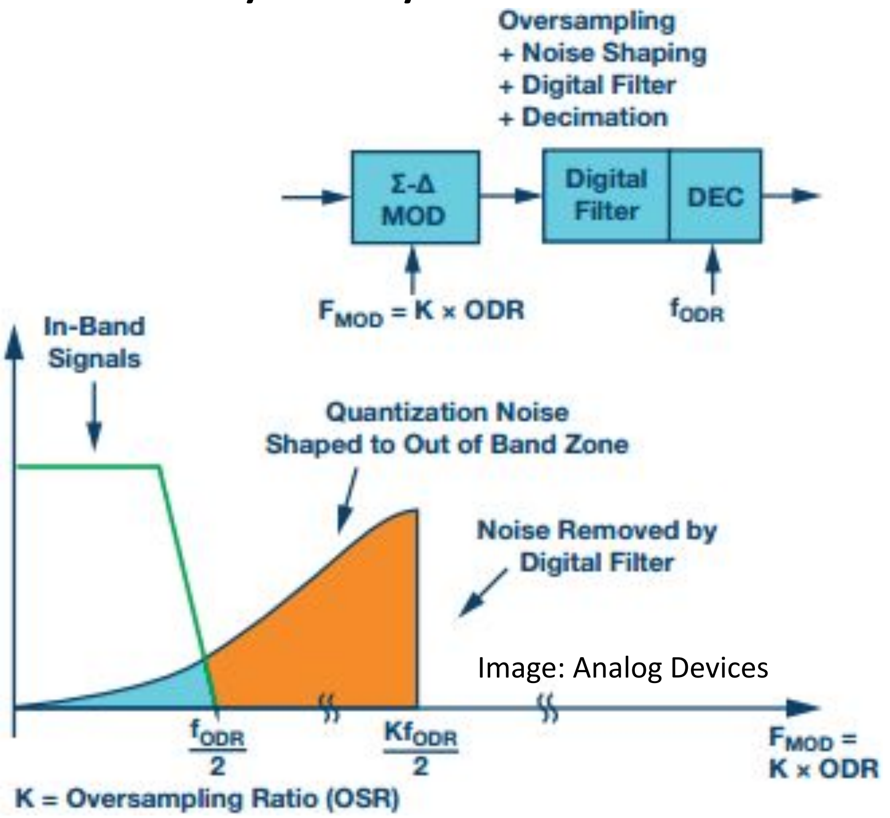
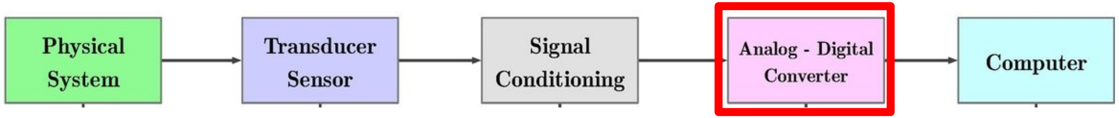



Image: Analog Devices

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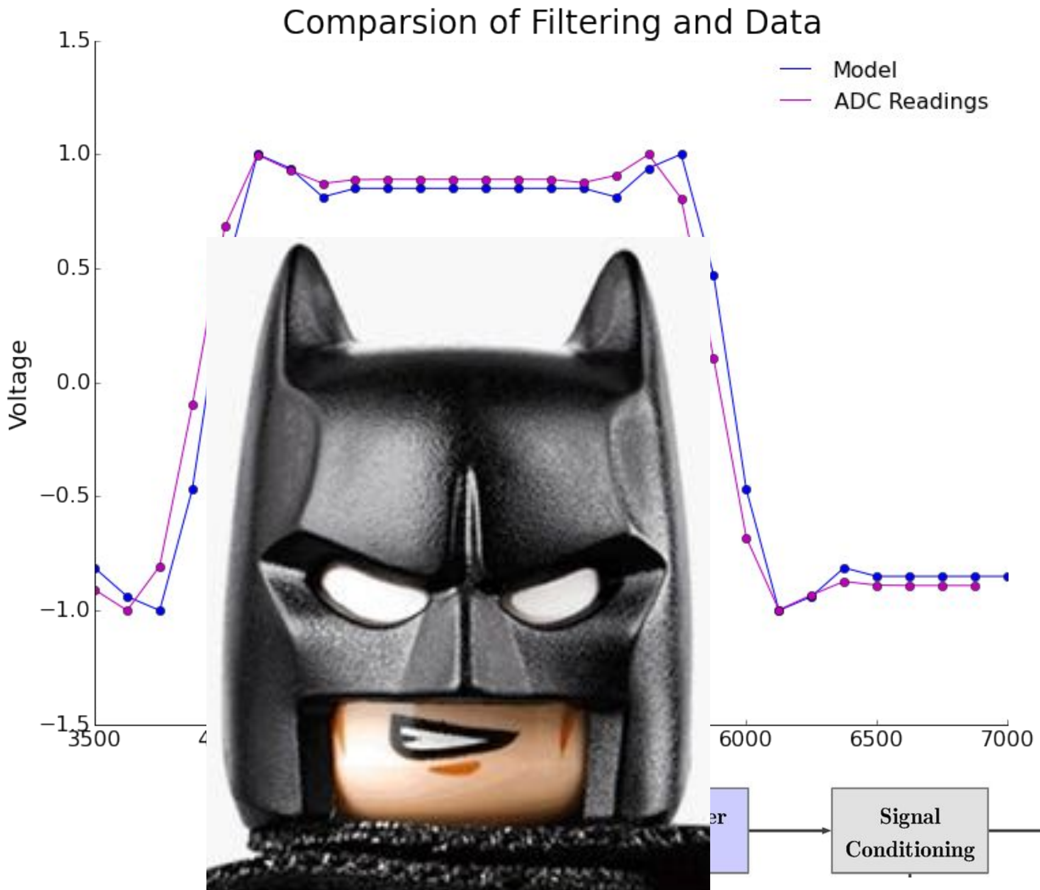
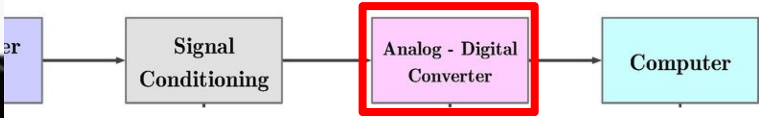
# Be careful of filters in the ADC introducing artifacts though!

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Luckily by the time we digitize we're mostly done with filtering our physical values, but of course we can still drop packets, etc.

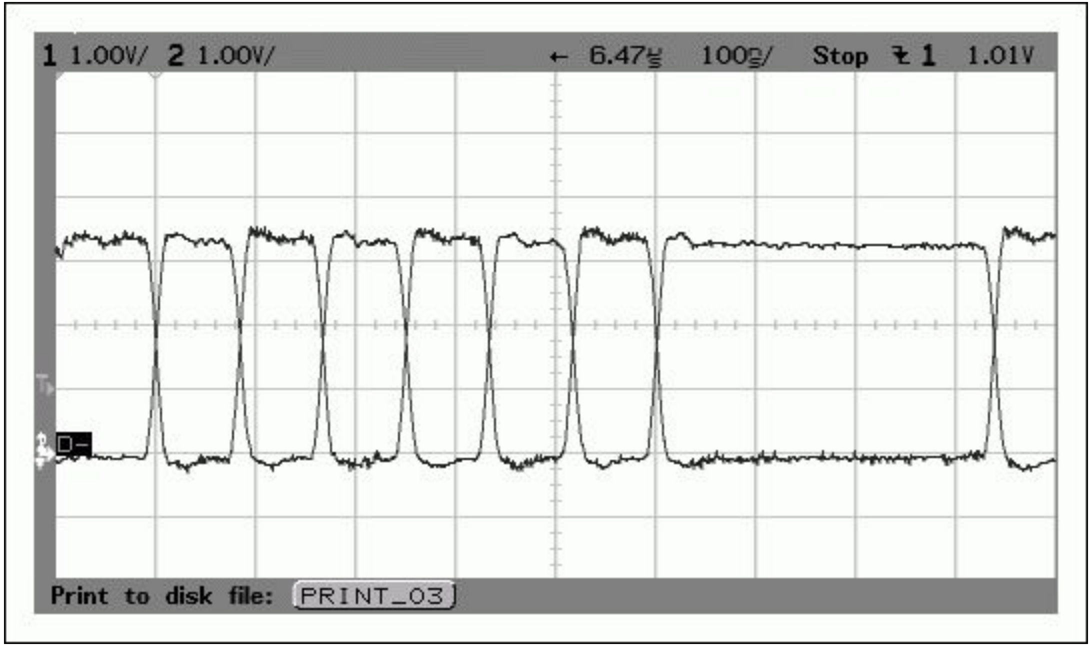


Image: Maxim Integrated

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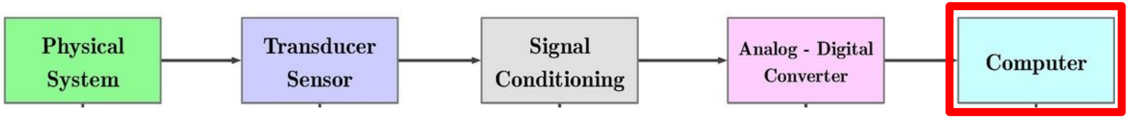


IT'S DEMO TIME!

# DACs go the other way, creating a voltage for a digital input value

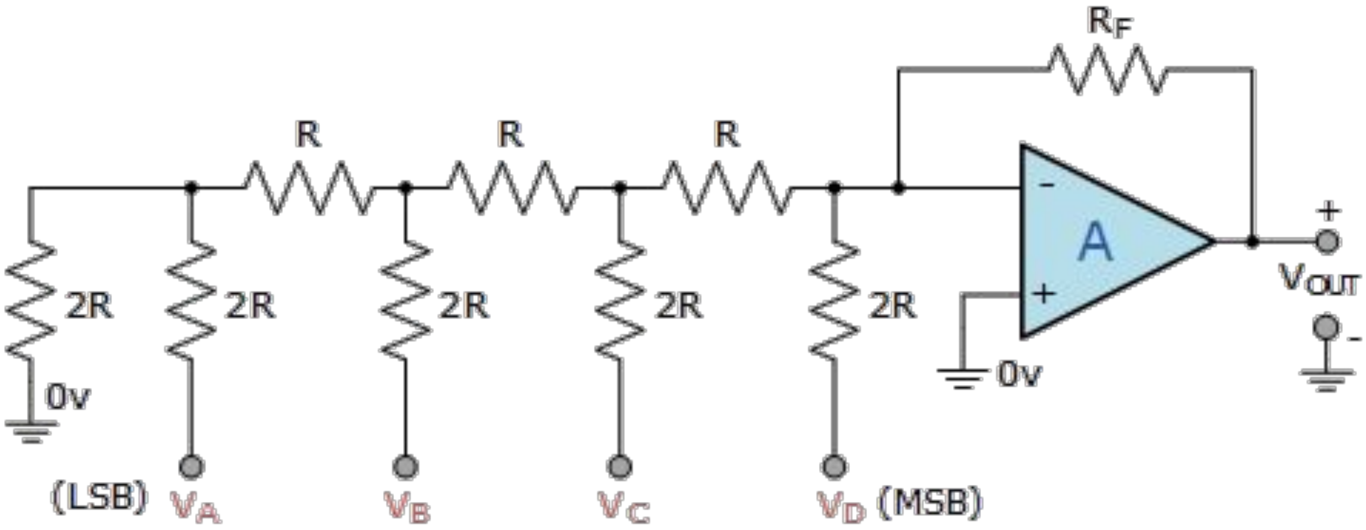


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Image: Electronics Tutorials

# Pulse width modulation

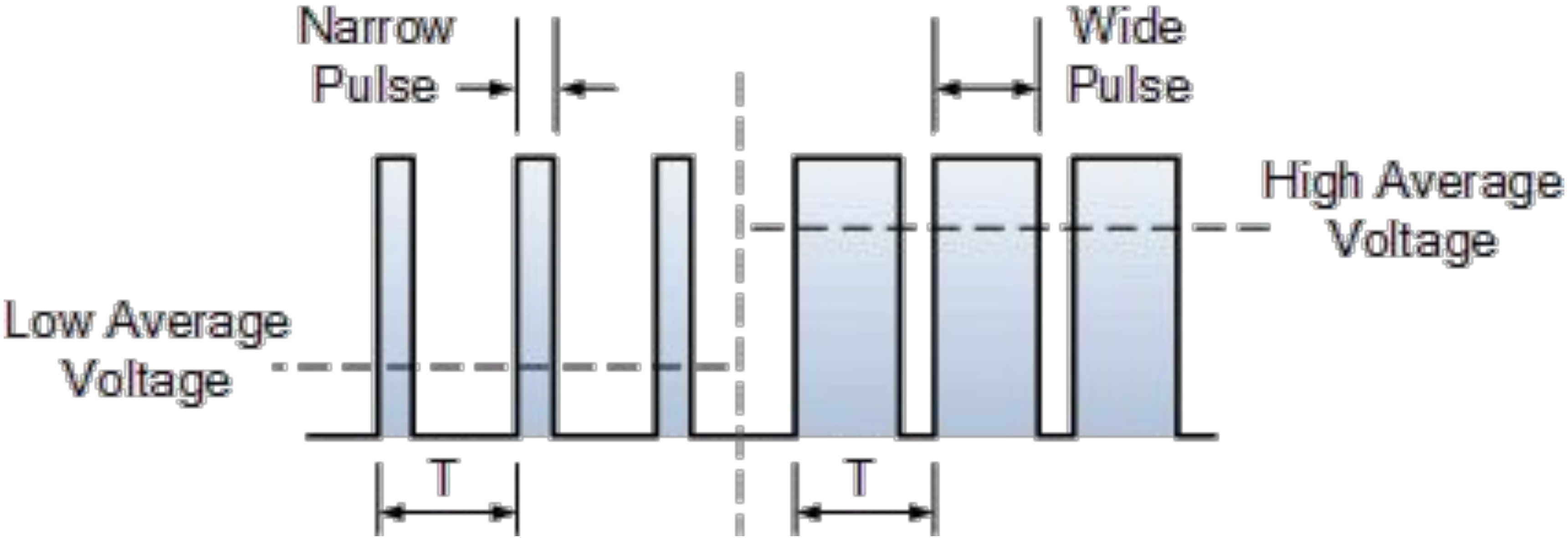


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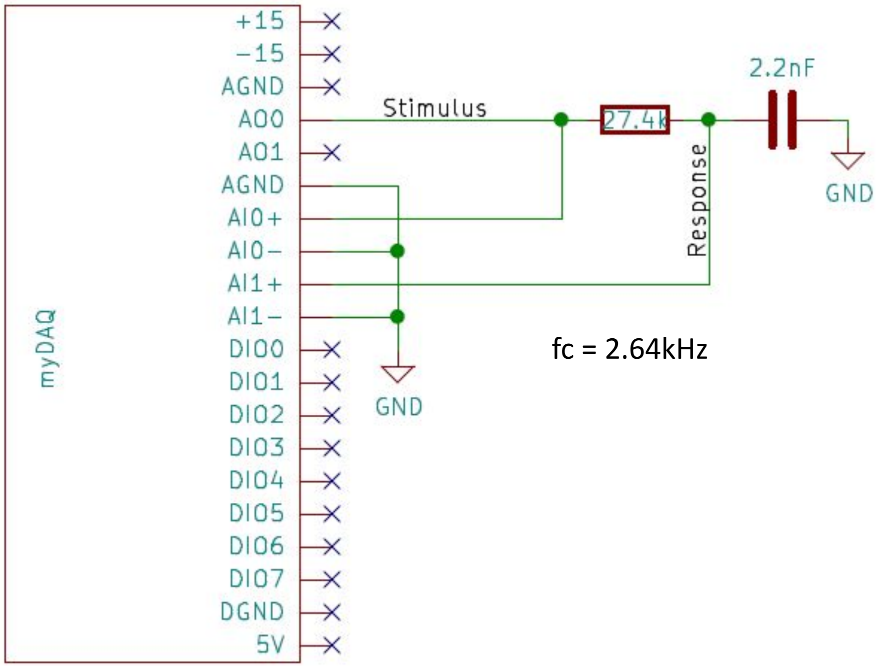
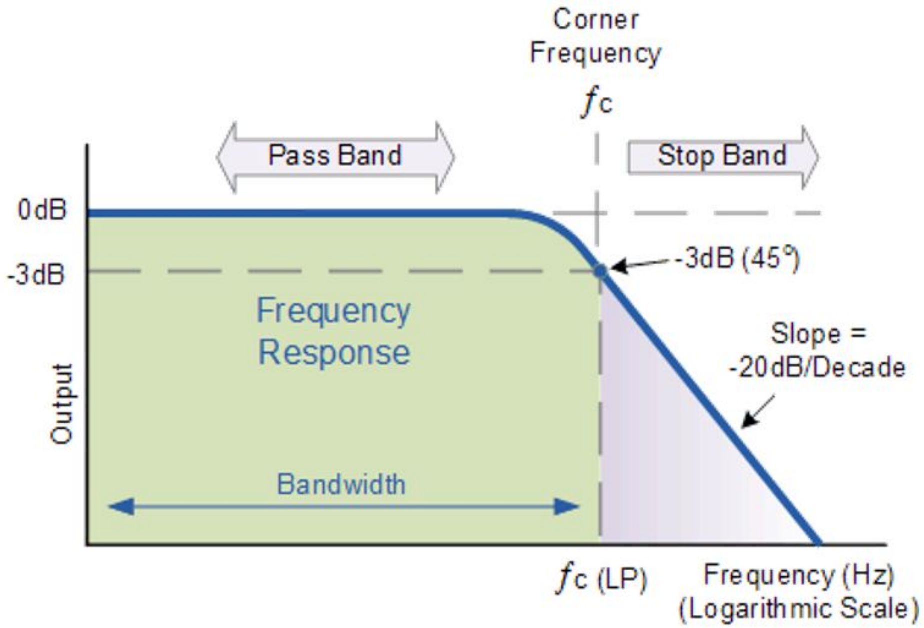
Image: Electronics Tutorials


DEMO TIME!
38
...WHAT COULD POSSIBLY GO WRONG
makeameme.org

# Let’s look at the simple RC low-pass filter




fc = 2.64kHz




$$fc = \frac{1}{2\pi RC}\ Hz$$

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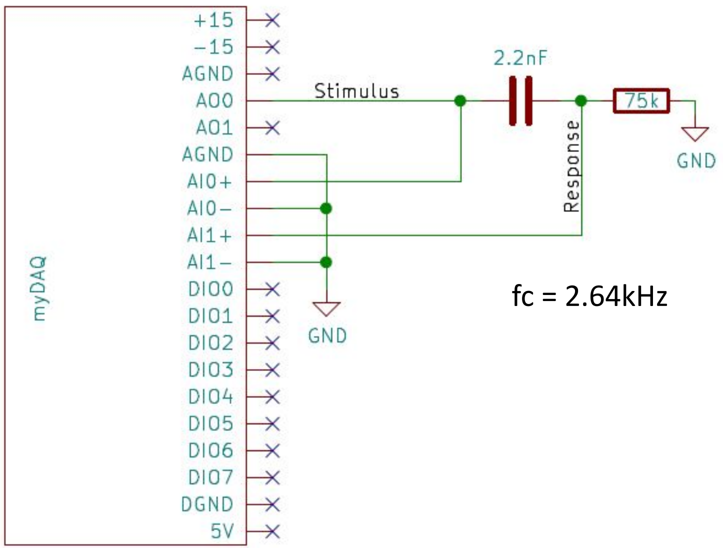
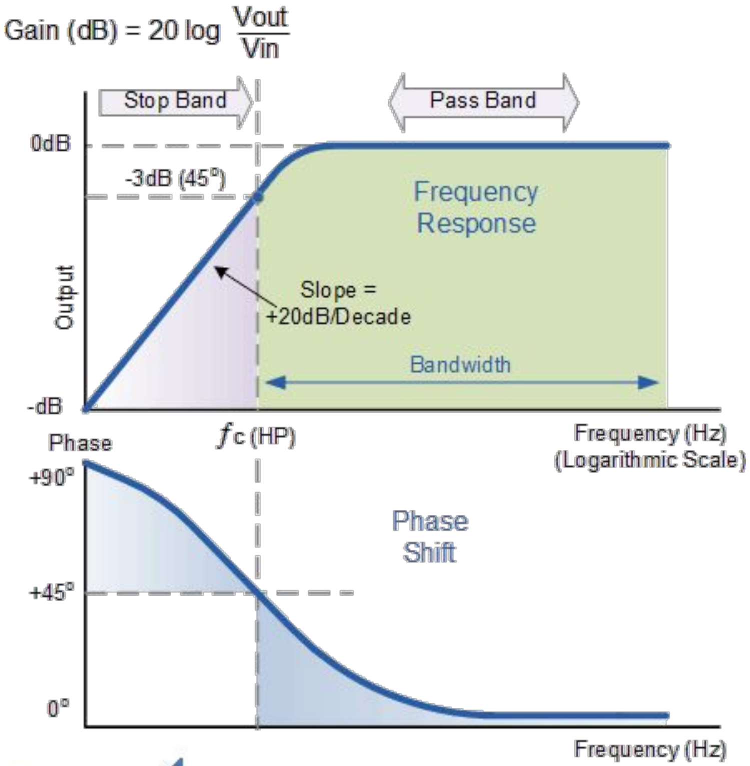
# Now the same thing, but a high-pass

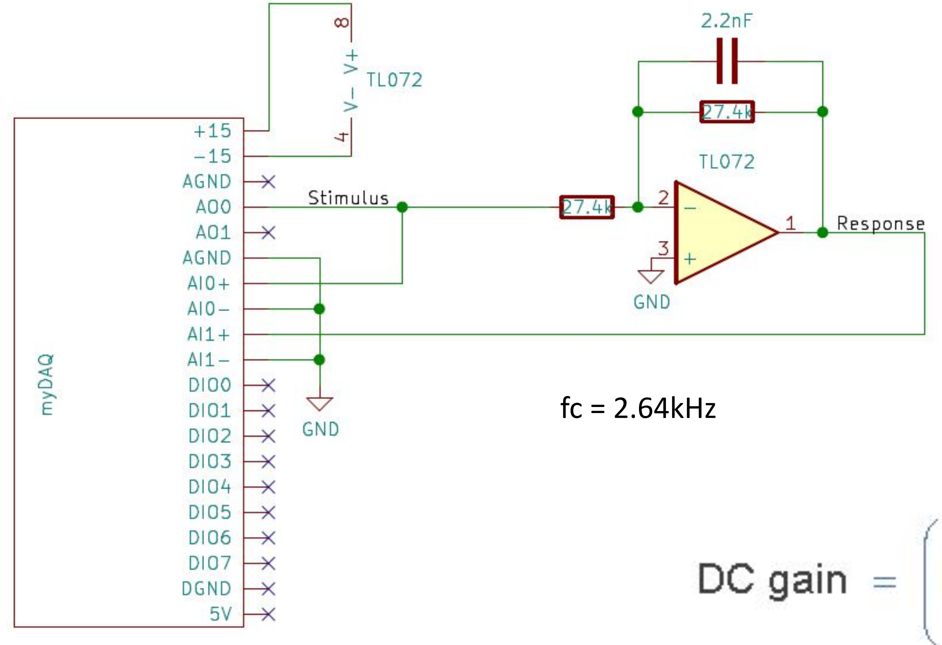



fc = 2.64kHz

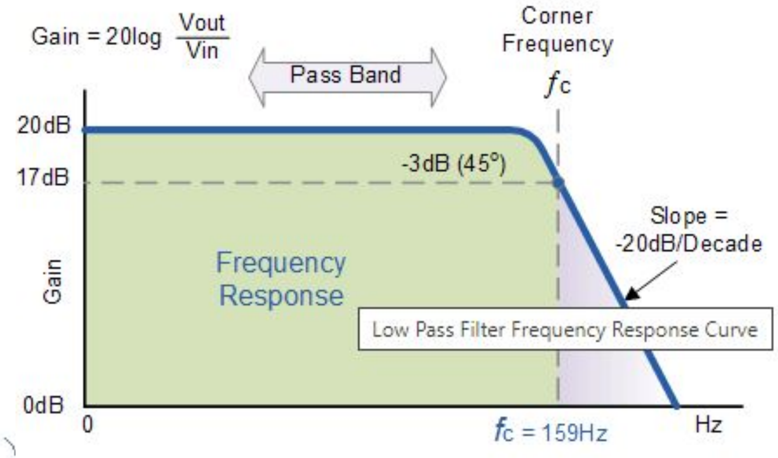



$$fc = \frac{1}{2\pi RC} Hz$$

# Active filters filter, buffer, and amplify all at the same time




fc = 2.64kHz




$$\text{DC gain} = \left(1 + \frac{R_2}{R_1}\right)$$

$$fc = \frac{1}{2\pi RC}\,Hz$$

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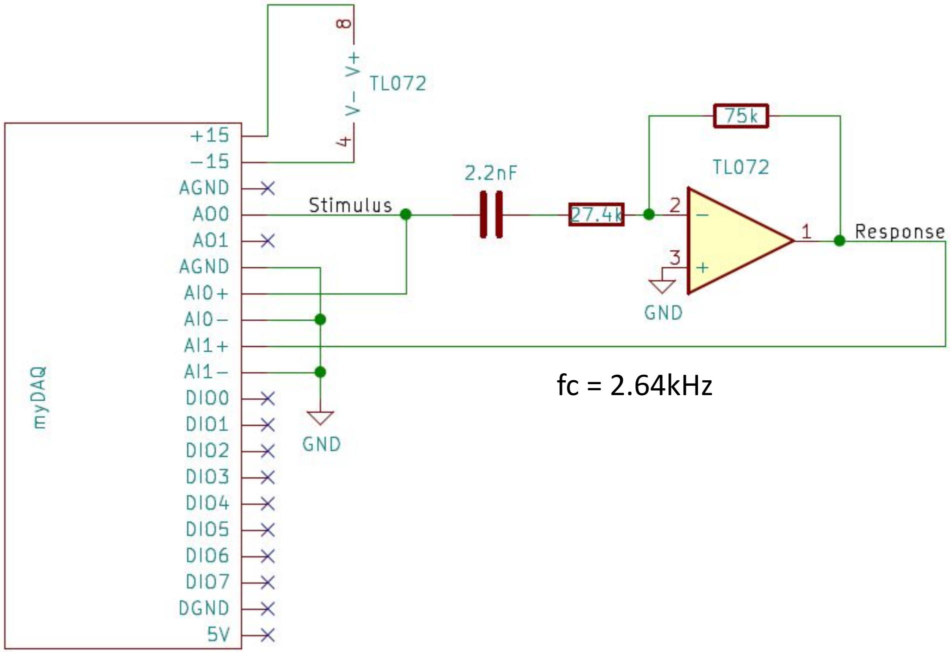
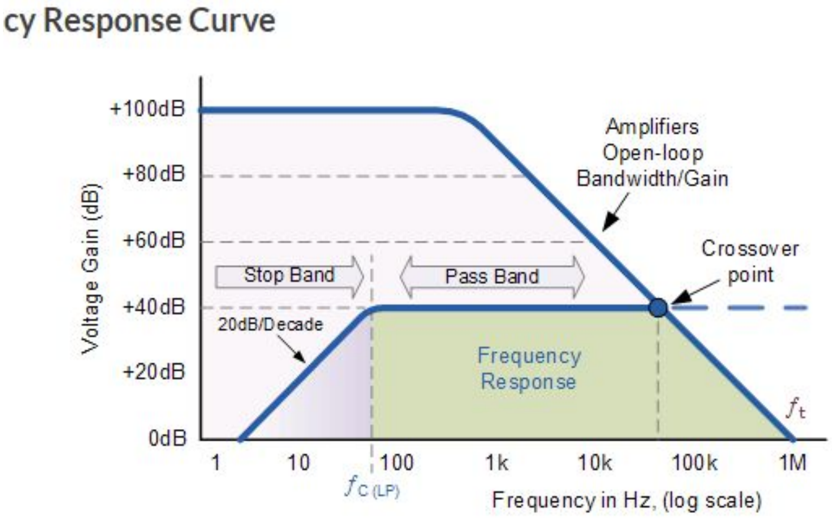
# Active filters filter, buffer, and amplify all at the same time



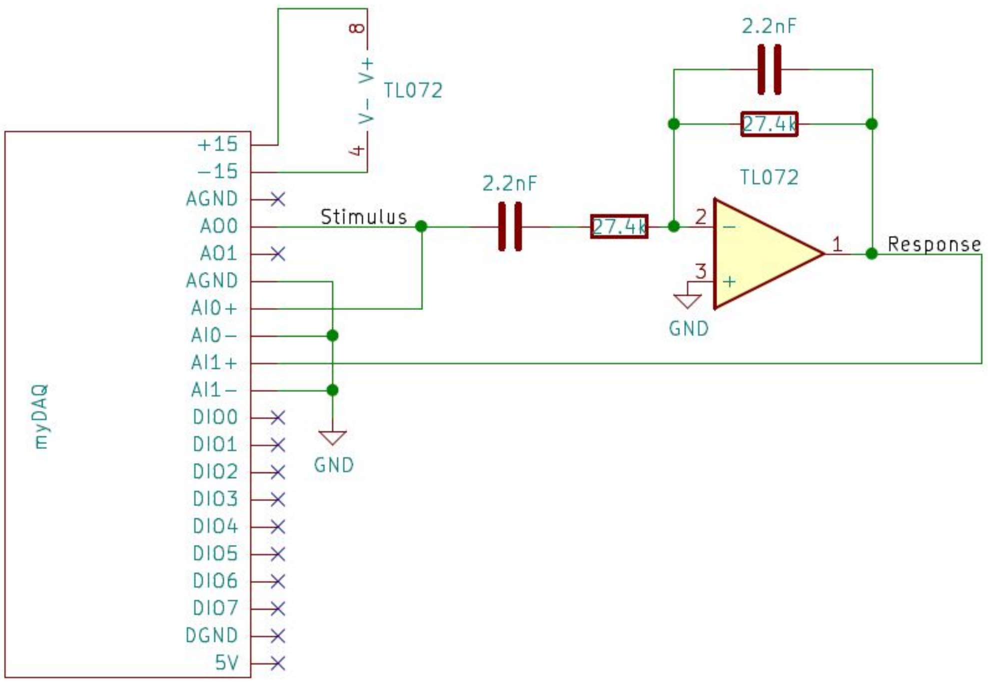

fc = 2.64kHz

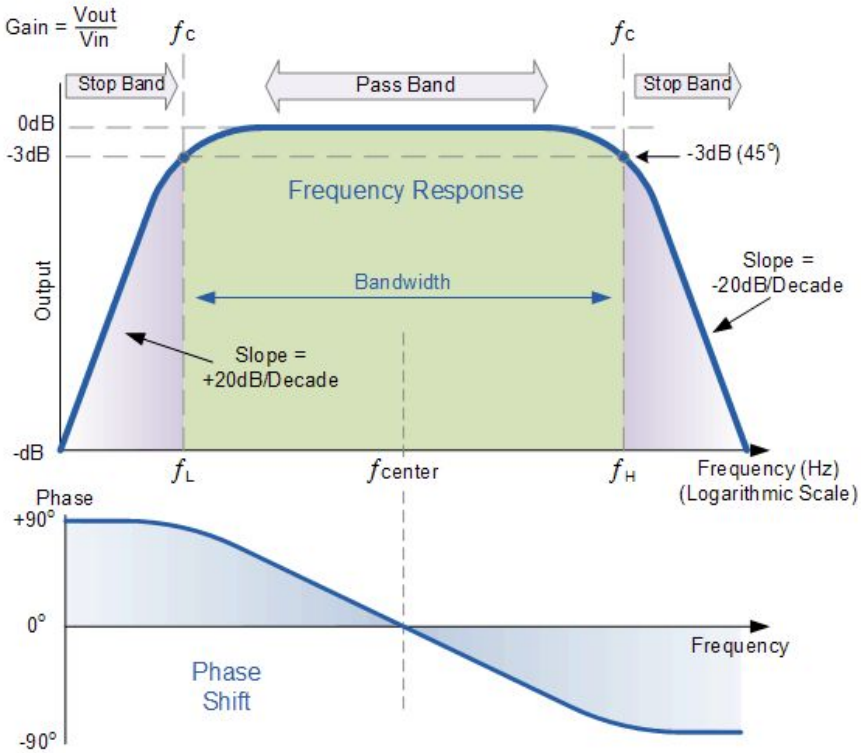



$$A_v = -\frac{R_2}{R_1}$$

$$f_c = \frac{1}{2\pi R_1 C}$$

# Combined we can make a bandpass filter

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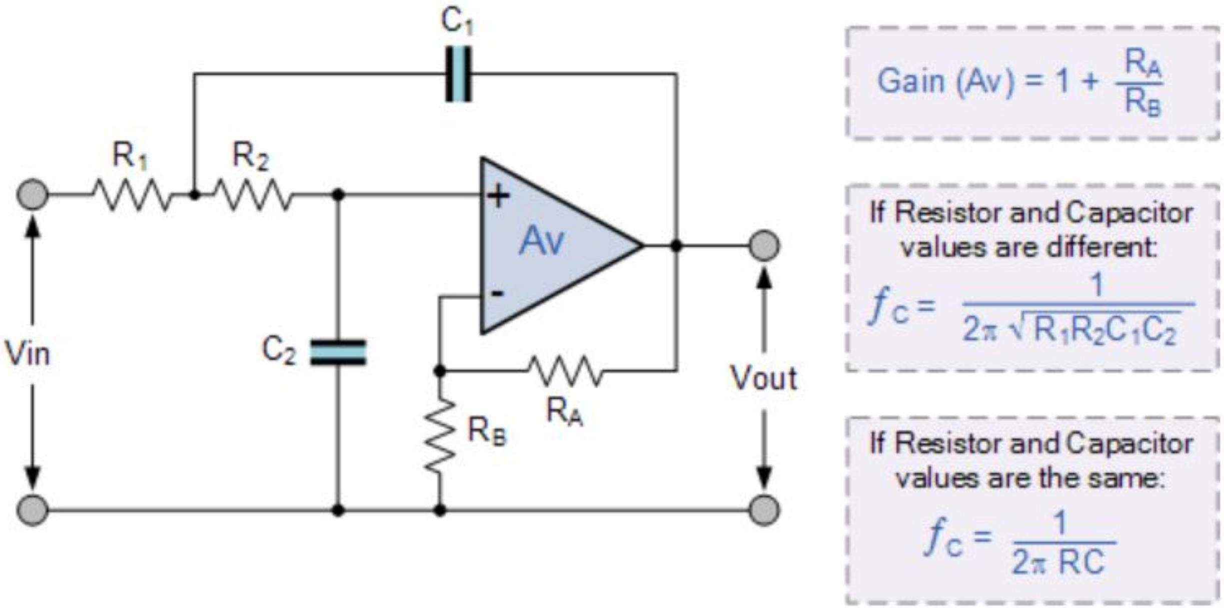
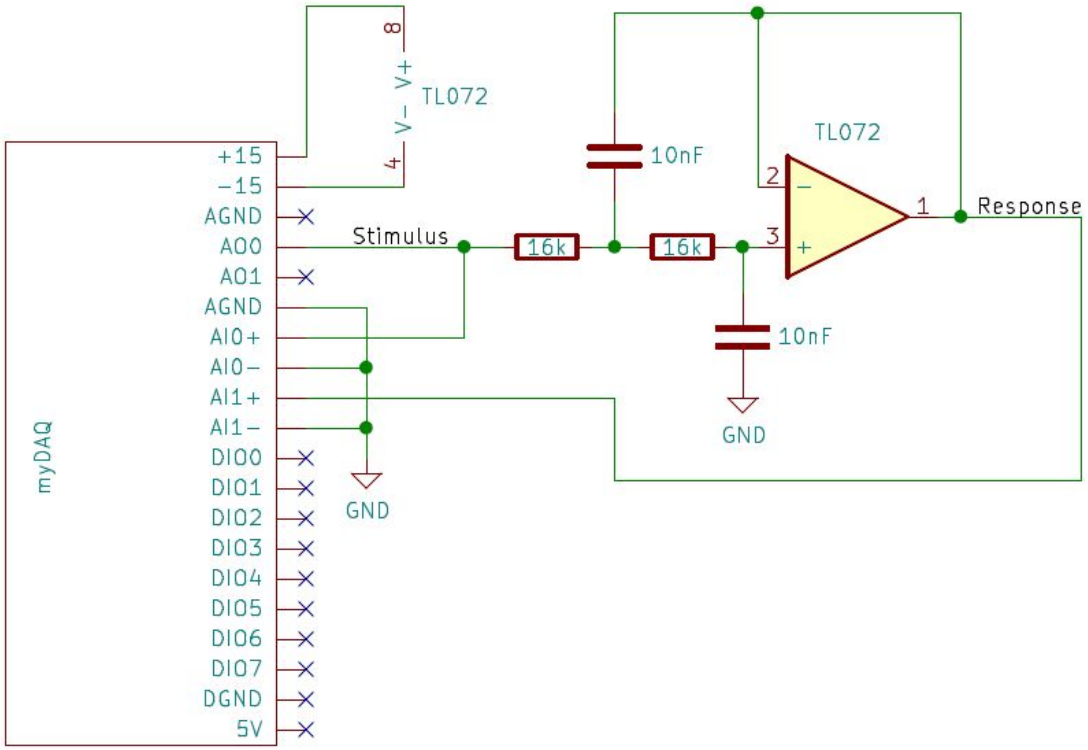
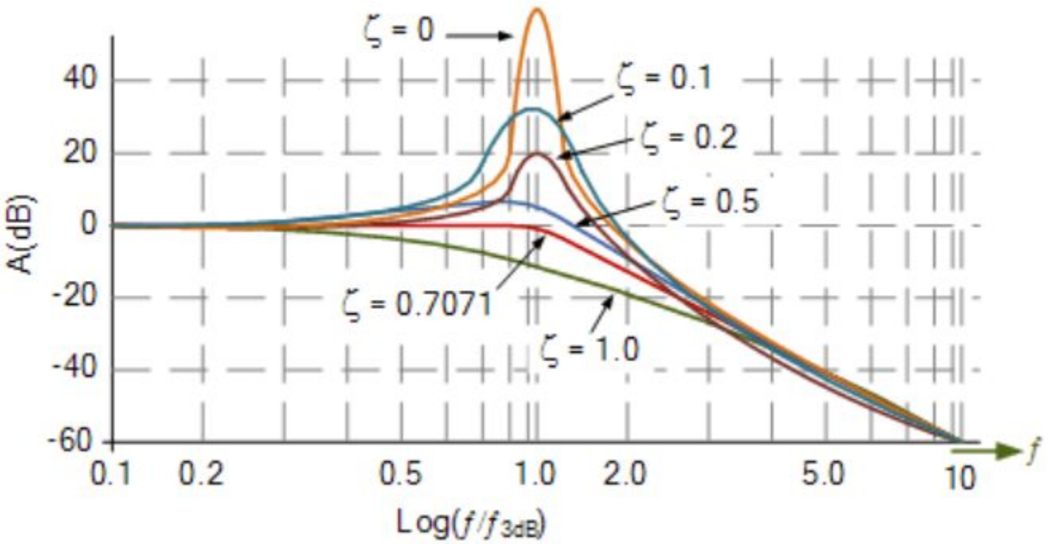
# We can continue the complication with higher order filters



# A solid and easy to implement filter is the Sallen-Key



Generally keep gains below about 3 as resonance can be an issue. Higher gains decrease the damping factor.



# Chip scale filters are really the way to go for sharp and well defined responses, but they are expensive

*19-4788; Rev 0; 10/98*

**MAXIM**

## *8th-Order, Lowpass, Bessel, Switched-Capacitor Filters*

### *General Description*

The MAX7401/MAX7405 8th-order, lowpass, Bessel, switched-capacitor filters (SCFs) operate from a single +5V (MAX7401) or +3V (MAX7405) supply. These devices draw only 2mA of supply current and allow corner frequencies from 1Hz to 5kHz, making them ideal for low-power post-DAC filtering and anti-aliasing applications. They feature a shutdown mode, which reduces the supply current to 0.2µA.

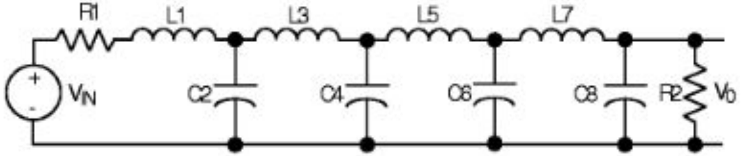
Two clocking options are available on these devices: self-clocking (through the use of an external capacitor) or external clocking for tighter corner-frequency control. An offset adjust pin allows for adjustment of the DC output level.

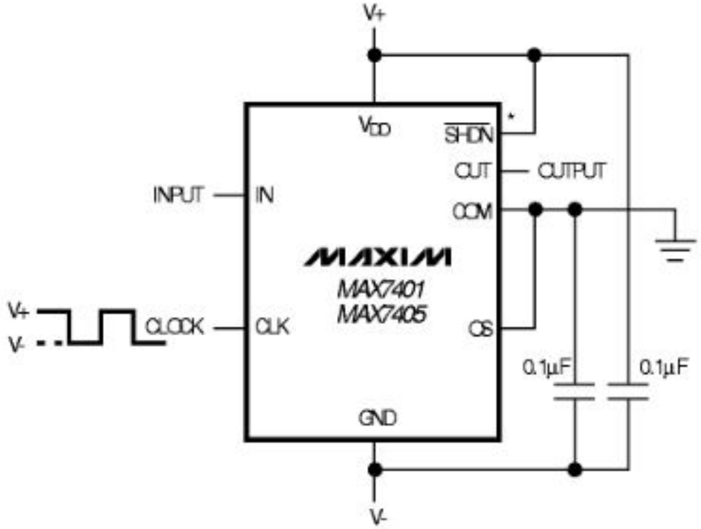
The MAX7401/MAX7405 Bessel filters provide low overshoot and fast settling. Their fixed response simplifies the design task to selecting a clock frequency.

### *Features*

- **8th-Order, Lowpass Bessel Filters**
- **Low Noise and Distortion: -82dB THD + Noise**
- **Clock-Tunable Corner Frequency (1Hz to 5kHz)**
- **100:1 Clock-to-Corner Ratio**
- **Single-Supply Operation**
  - **+5V (MAX7401)**
  - **+3V (MAX7405)**
- **Low Power**
  - **2mA (Operating Mode)**
  - **0.2µA (Shutdown Mode)**
- **Available in 8-Pin SO/DIP Packages**
- **Low Output Offset: ±5mV**

**MAX7401/MAX7405**

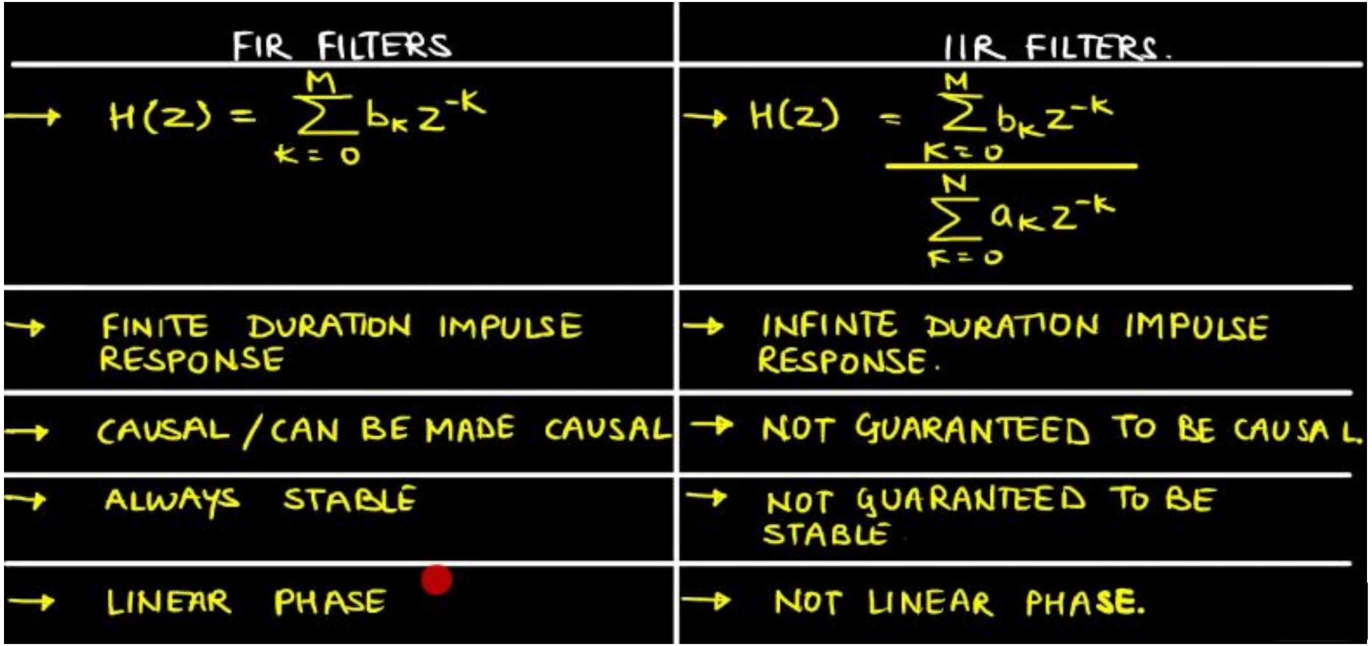


# A few other practical notes on filter topologies

- Butterworth - Amplitude accuracy is very flat and well controlled
- Chebyshev - Very steep rolloff, but more ripple in the passband
- Bessel - Uniform time delay for constant group delay (i.e. linear phase response with frequency) and great transient of pulse input response. Have ripple in the passband and slow initial roll-off.

# Digital filtering is a different class, but FIR/IIR filters are out there



| FIR FILTERS | IIR FILTERS. |
|---|---|
| → $H(z) = \sum_{k=0}^{M} b_k z^{-k}$ | → $H(z) = \dfrac{\sum_{k=0}^{M} b_k z^{-k}}{\sum_{k=0}^{N} a_k z^{-k}}$ |
| → FINITE DURATION IMPULSE RESPONSE | → INFINTE DURATION IMPULSE RESPONSE. |
| → CAUSAL / CAN BE MADE CAUSAL | → NOT GUARANTEED TO BE CAUSAL. |
| → ALWAYS STABLE | → NOT GUARANTEED TO BE STABLE |
| → LINEAR PHASE | → NOT LINEAR PHASE. |

Checkout http://t-filter.engineerjs.com/

# There’s even an easy Arduino FIR library!