Lecture 6: Data Acquisition I

- Architecture of DAQ systems
- Signal conditioning
- Aliasing



Architecture of data acquisition systems





Signal conditioning

- Instrumentation amplifiers
- Filters
- Integrators/differentiators (previous lecture)



Instrumentation amplifiers

Consider the difference amplifier we saw in the previous lecture



We define COMMON-MODE and DIFFERENCE-MODE voltage as





Instrumentation amplifiers

■ As a result of a mismatch in the resistors (R'_k≠ R_k), the differential inputs may not have the same gain

$$V_{0} = G(V_{2} - V_{1})^{R_{k}^{'} \neq R_{k}} G_{2}V_{2} - G_{1}V_{1} = G_{2}\left(-\frac{V_{D}}{2} + V_{CM}\right) - G_{1}\left(\frac{V_{D}}{2} + V_{CM}\right) = -V_{D}\left(\frac{G_{2} + G_{1}}{2}\right) + V_{CM}(G_{2} - G_{1}) = -V_{D}G_{D} + V_{Cm}G_{CM}$$

We define COMMON-MODE REJECTION RATIO as

$$CMRR = 20log_{10} \left(\frac{G_{D}}{G_{CM}} \right) = 20log_{10} \left(\frac{G_{2} + G_{1}}{2(G_{2} - G_{1})} \right)$$

- CMRR is, in practice, a function of frequency, and its magnitude decreases with increasing frequency
- An additional shortcoming of the difference amplifier is its LOW INPUT IMPEDANCE



Instrumentation amplifiers

- The term INSTRUMENTATION AMPLIFIER is used to denote a difference amplifier with
 - High gain (recall INA2126 in Lab I)
 - Single-ended output
 - High input impedance
 - High CMRR
- High input impedance may be achieved by buffering the differential inputs



- This solution, however, requires high CMRR both in the followers and in the final op-amp
 - Otherwise, since the input buffers have unity gain, all the CM rejection must come in the output op-amp, requiring precise resistor matching



Common mode rejection ratio

A better solutions is the "standard" instrumentation amplifier shown below

- Input stage provides high $\rm G_{\rm D}$ and unity $\rm G_{\rm CM}$
 - Close resistor (R₂) matching is NOT critical
- As a result, the output op-amp (U₃) does not require exceptional CMRR and resistor matching in U₃ is not critical
- Offset trimming can be done at one of the input op-amps





Filters

- **Filters are used to remove** *unwanted* **bandwidths from a signal**
- Filter classification according to implementation
 - Active filters include RC networks and op-amps
 - Suitable for low frequency, small signal
 - Active filters are preferred since avoid the bulk and non-linearity of inductors and can have gains greater than 0dB
 - However, active filters require a power supply
 - Passive filters consist of RCL networks
 - Simple, more suitable for frequencies above audio range, where active filters are limited by the op-map bandwidth
 - Digital filters
 - DSP is beyond the scope of this course



Filters

Filter classification according to frequency response

- Low-pass filter
- High-pass filter
- Band-pass filter
- Band-stop (Notch)





Low- and high-pass filters



High pass filters



From [Ram96]

From [Ram96]



Band-pass and band-stop filters

Band-pass

- High-pass and low pass in series
 - High-pass should usually precede
 - Corner frequency of low-pass must then be higher
 - If these are passive filters they should be buffered in between





State-variable filters

Also known as a Universal Active Filter

- Consists of one amplifier and two integrators
- High-pass, low-pass and band-pass in the same IC
- Example below: Burr Brown UAF42





Anti-aliasing

The sampling theorem

- A continuous signal can be represented completely by, and reconstructed from, a set of instantaneous measurements or samples of its voltage which are made at equally-spaced times. The interval T(=1/F_S) between such samples must be less than one-half the period of the highest-frequency component F_{MAX} in the signal
- In other words: you must sample at least twice the rate of the maximum frequency in your signal to prevent aliasing (F_S≥2F_{MAX})
- The sampling rate $F_s=2F_{MAX}$ is called the Nyquist rate





Anti-aliasing

The effects of aliasing can also be observed on the frequency spectrum of the signal

Alias frequency $\hat{E} - \min kE - E$

- In the figures below
 - F_1 appears correctly since $F_1 \le F_S/2$
 - F_2 , F_3 and F_4 have aliases at 30, 40 and 10Hz, respectively
 - You can compute these aliased frequencies by $\underline{folding}$ the spectrum around $\mathrm{F_S}/\mathrm{2}$ or with the expression

FIGURE 117.5 Spectral of signal with multiple frequencies.

FIGURE 117.6 Spectral of signal with multiple frequencies after sampled at fs = 100 Hz.



Anti-aliasing filters

- An anti-aliasing filter is a low-pass filter designed to filter out frequencies higher than the sampling frequency
 - An anti-aliasing filter should have
 - Steep cut-off and
 - Flat response in the frequency band



• Typical filters are:

- **Butterworth**: flattest response in the frequency band but phase shifts well below the break frequency
- **Bessel**: phase shift proportional to frequency, so the signal is not distorted by the filter
 - Recommended for anti-aliasing if it is important to preserve the waveform
- **Chebyshev**: steepest cut-off but it has ripples in the band-pass



References

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