Data Acquisition I

- ▶ Architecture of DAQ systems
- Signal conditioning
- Aliasing

Architecture of data acquisition systems

Signal conditioning

- **Instrumentation amplifiers**
- ▶ Filters
- ▶ Integrators/differentiators

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 $(R'_k \neq R_k)$, the differential inputs may not have the same gain

$$
V_0 = G(V_2 - V_1)^{\frac{R_1' \times R_1}{2}} G_2 V_2 - G_1 V_1 = G_2 \left(-\frac{V_0}{2} + V_{CM} \right) - G_1 \left(\frac{V_0}{2} + V_{CM} \right) =
$$

= $-V_0 \left(\frac{G_2 + G_1}{2} \right) + V_{CM} (G_2 - G_1) = -V_0 G_0 + V_{CM} G_{CM}$

. We define COMMON-MODE REJECTION RATIO as

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

- 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 (INA2126)
	- Single-ended output
	- High input impedance
	- High CMRR
- \blacktriangleright High input impedance may be achieved by buffering the differential inputs
	- $\,\circ\,$ This solution, however, requires high CMR $\,\vdash\,$ both in the followers and in the final opamp
		- 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 GD and unity GCM
		- Close resistor (R2) matching is NOT critical
	- As a result, the output opamp (U3) does not require exceptional CMRR and resistor matching in U3is not critical
	- Offset trimming can be done at one of the input op-amps

INA... U_i $R_{\rm{2-10k}}$ $\qquad \qquad -$ Ō ₩ 灬 $R_{\scriptscriptstyle 3}$ $190\mathrm{k}$ ξ

$$
A = \left(1 + \frac{2R}{R_a}\right) \frac{R_3}{R_2}
$$

INA126.....

APPLICATIONS

- **. INDUSTRIAL SENSOR AMPLIFIER:** Bridge, RTD, Thermocouple
- **PHYSIOLOGICAL AMPLIFIER:** ECG, EEG, EMG
- **. MULTI-CHANNEL DATA ACQUISITION**
- **O PORTABLE, BATTERY OPERATED SYSTEMS**

PIN CONFIGURATION (Single)

Ve

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 nonlinearity 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
	- n High-pass filter
	- Band-pass filter
	- Band-stop (Notch)

Low- and high-pass filters

Low pass filters

High pass filters \blacksquare

From (Nam96)

From [Ramile]

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

Band-stop

- . High-pass and low-pass in parallel followed by a summer
	- Corner frequency of high-pass must be higher

Types of Filters

normalized frequency (rad/s)

Normalized low-pass Butterworth filter reponse curves 0 $n=1$ Attenuation (dB)
 $\frac{1}{6}$ -20 $= 2$ $n = 32$ $n = 10$ -60 10 0 normalized frequency (rad/s)

Attenuation curves for Butterworth LPF

LC Low Pass Filter network

π

 τ

Note: Values of L_n and C_n are for a 1- Ω load and -3-dB frequency of 1 rad/s and have units of H and F. These values must be scaled down. See text.

Example

Suppose that you want to design a low-pass filter that has a f_{3dB} = 3000 Hz (attenuation is -3 dB at 3000 Hz) and an attenuation of -25 dB at a frequency of 9000 Hz-which will be called the *stop frequency f_s*. Also, let's assume that both the signal-source impedance R_s and the load impedance R_t are equal to 50 Ω . How do you design the filter?

Step 1: Normalization

Step 2&3: Pick Response Curve and determine number of Poles

n=3 from the curve for Butterworth

Step 4: Create Normalized Filter

Step 5: Frequency and Impedance **Scaling**

Active Filter Design

Basic two-pole section

Basic three-pole section

Normalized Curves

TABLE 8.2 Butterworth Normalized Active Low-Pass Filter Values

Exqample

Suppose that you wish to design an active lowpass filter that has a 3-dB point at 100 Hz and at least 60 dB worth of attenuation at 400 Hzwhich we'll call the stop frequency f_s.

Solution

High Pass Filter - Passive -

Suppose that you want to design a high-pass filter that has an $f3dB$ $= 1000$ Hz and an attenuation of at least -45 dB at 300 Hz—which we call the *stop* frequency fs. Assume that the filter is hooked up to a source and load that both have impedances of 50 Ω and that a Butterworth response is desired

How do you design the filter?

Solution

To convert the low-pass into a high-pass filter, replace the inductors with capacitors that have value of $1/L$, and replace the capacitors with inductors that have values of $1/C$. In other words, do the following:

$$
L_{2(\text{transf})} = 1/C_{2(\text{table})} = 1/1.6180 = 0.6180 \text{ H}
$$
\n
$$
L_{4(\text{transf})} = 1/C_{4(\text{table})} = 1/1.6180 = 0.6180 \text{ H}
$$

$n=5$ from the curves

Start with a " T " low-pass filter...

Transform low-pass filter into a high-pass filter...

 $C_{1(transf)} = 1/L_{1(table)} = 1/0.6180 = 1.6180$ F $C_{3(trank)} = 1/L_{3(table)} = 1/2.0 = 0.5$ F C_{5 (transf)</sub> = $1/L_{5$ (table) = $1/0.6180 = 1.6180$ F

Now Scaling

Next, frequency and impedance scale to get the actual component values:

$$
C_{1(\text{actual})} = \frac{C_{1(\text{trans})}}{2\pi f_{3\text{dB}} R_L} = \frac{1.618 \text{ H}}{2\pi (1000 \text{ Hz})(50 \Omega)} = 5.1 \text{ }\mu\text{F}
$$
\n
$$
L_{2(\text{actual})} = \frac{L_{2(\text{trans})} R_L}{2\pi f_{3\text{dB}}} = \frac{(0.6180 \text{ F})(50 \Omega)}{2\pi (1000 \text{ Hz})} = 4.9 \text{ }\mu\text{F}
$$
\n
$$
C_{3(\text{actual})} = \frac{C_{3(\text{trans})}}{2\pi f_{3\text{dB}} R_L} = \frac{0.5 \text{ H}}{2\pi (1000 \text{ Hz})(50 \Omega)} = 1.6 \text{ }\mu\text{F}
$$
\n
$$
L_{4(\text{actual})} = \frac{L_{4(\text{trans})} R_L}{2\pi f_{3\text{dB}}} = \frac{(0.6180 \text{ F})(50 \Omega)}{2\pi (1000 \text{ Hz})} = 4.9 \text{ }\mu\text{F}
$$
\n
$$
C_{5(\text{actual})} = \frac{C_{5(\text{trans})}}{2\pi f_{3\text{dB}} R_L} = \frac{1.618 \text{ H}}{2\pi (1000 \text{ Hz})(50 \Omega)} = 5.1 \text{ }\mu\text{F}
$$

Impedance and frequency scale high-pass filter to get final circuit

High Pass Filter-Active-

suppose

that you want to design a high-pass filter with a -3-dB frequency of 1000 Hz and 50 dB worth of attenuation at 300 Hz. What do you do?

High-pass frequency response Translation to normalized low-pass response 0dB $0dB$ $V_{\rm out}$ $W_{\rm in}(\rm dB)$ $-3dB$ $V_{\rm out}/V_{\rm in}({\rm dB})$ $-3dB$ $-50dB$ $-50dB$ 25 400 1000 frequency (Hz) frequency (rad/s or 0)

 $n=5$ from the curves

Normalized low-pass filter

Next, the normalized low-pass filter must be converted into a normalized high-pass filter. To make the conversion, exchange resistors for capacitors that have values of $1/R$ F, and exchange capacitors with resistors that have values of 1/C Ω .

Follow.....

Bandpass Filter Design-Wide Band-Passive-

When is the filter wide band??

If f_2/f_1 is greater than 1.5,

Suppose that you want to design a bandpass filter that has -3 -dB points at $f_1 = 1000$ Hz and f_2 = 3000 Hz and at least -45 dB at 300 Hz and more than -25 dB at 9000 Hz. Also, again assume that the source and load impedances are both 50 Ω and a Butterworth design is desired. \mathbf{m} 0.0001 L

Bandpass Filter Design-Wide Band-Active-

Suppose that you want to design a bandpass filter that has -3 -dB points at $f_1 = 1000$ Hz and f_2 = 3000 Hz and at least -30 dB at 300 and 10,000 Hz. What do you do?

 $\frac{f_2}{f_1} = \frac{3000 \text{ Hz}}{1000 \text{ Hz}} = 3$

 -3 dB at 3000 Hz Low-pass: -30 dB at 10,000 Hz -3 dB at 1000 Hz High-pass: -30 dB at 300 Hz

The steepness factor for the low-pass filter is

$$
A_s = \frac{f_s}{f_{3dB}} = \frac{10,000 \text{ Hz}}{3000 \text{ Hz}} = 3.3
$$

while the steepness factor for the high-pass filter is

Follow...

Narrow Bandwidth BPF-Passive (optional)

Suppose that you want to design a bandpass filter with -3 -dB points at $f_1 = 900$ Hz and $f_2 = 1100$ Hz and at least -20 dB worth of attenuation at 800 and 1200 Hz. Assume that both the source and load impedances are 50 Ω and that a Butterworth design is desired.

Since $f_2/f_1 = 1.2$, which is less than 1.5, a narrow-band filter is needed.

geometric center frequency $f_0 = \sqrt{f_1 f_2} = \sqrt{(900 \text{ Hz})(1100)} = 995 \text{ Hz}$

Next, compute the two pair of geometrically related stop-band frequencies by using

 $f_a f_b = f_0^2$ $f_a = 800 \text{ Hz}$ $f_b = \frac{f_0^2}{f_a} = \frac{(995 \text{ Hz})^2}{800 \text{ Hz}} = 1237 \text{ Hz}$ $f_b - f_a = 437 \text{ Hz}$ $f_b = 1200 \text{ Hz}$ $f_a = \frac{f_0^2}{f_b} = \frac{(995 \text{ Hz})^2}{1200 \text{ Hz}} = 825 \text{ Hz}$ $f_b - f_a = 375 \text{ Hz}$ Choose the pair having the least separation, which represents more severe requirements-375 Hz

Follow.....

Low-pass bandpass relationship

Normalized low-pass response

Follow......

Impedance and frequency scaled low-pass filter

The important part comes now. Each circuit branch of the lowpass filter must be resonated to f_0 by adding a series capacitor to each inductor and a parallel inductor to each capacitor. The LC resonant equation is used to determine the additional component values:

$$
L_{\text{(parallel with }C1)} = \frac{1}{(2\pi f_0)^2 C_{1(\text{actual})}} = \frac{1}{(2\pi \cdot 995 \text{ Hz})^2 (15.92 \text{ }\mu\text{F})} = 1.61 \text{ mH}
$$
\n
$$
L_{\text{(parallel with }C3)} = \frac{1}{(2\pi f_0)^2 C_{3(\text{actual})}} = \frac{1}{(2\pi \cdot 995 \text{ Hz})^2 (15.92 \text{ }\mu\text{F})} = 1.61 \text{ mH}
$$
\n
$$
C_{\text{(series with }L2)} = \frac{1}{(2\pi f_0)^2 L_{2(\text{actual})}} = \frac{1}{(2\pi \cdot 995 \text{ Hz})^2 (79.6 \text{ mH})} = 0.32 \text{ }\mu\text{F}
$$

Final bandpass filter

BPF-Narrow Band-Active Design

Suppose that you want to design a bandpass filter that has a center frequency $f_0 =$ 2000 Hz and a -3-dB bandwidth $\Delta f_{BW} = f_2 - f_1 = 40$ Hz. How do you design the filter?

No Cascading Since $f_2/f_1 = 2040$ Hz $/1960$ Hz $= 1.04$,

Choose convenient value of C, let it

 0.01

 $Q = \frac{f_0}{f_0 - f_1} = \frac{2000 \text{ Hz}}{40 \text{ Hz}} = 50$ Narrow-band filter circuit $R_1 = \frac{Q}{2\pi f_0 C}$ $R_2 = \frac{R_1}{2C^2 - 1}$ $R_3 = 2R_1$ O $V_{\rm out}$

 $R_1 = \frac{50}{2\pi(2000 \text{ Hz})(0.01 \text{ }\mu\text{F})} = 79.6 \text{ k}\Omega$ $R_2 = \frac{79.6 \text{ k}\Omega}{2(50)^2 - 1} = 400 \Omega$ $R_3 = 2(79.6 \text{ k}\Omega) = 159 \text{ k}\Omega$

Final Design ...

Final filter circuit R_3
159K $0.01 \mu F$ \rightsquigarrow $rac{R_1}{79.6}$ K $0.01 \mu F$ $V_{\rm in}$ O-**WV** $\circ V_{\text{out}}$ $rac{R_2}{400\Omega}$ $\boldsymbol{+}$ ħ

Passive Notch Filter (optional)

EXAMPLE

Suppose that you want to design a notch filter with -3 -dB points at $f_1 = 800$ Hz and $f_2 = 1200$ Hz and at least -20 dB at 900 and 1100 Hz. Let's assume that both the source and load impedances are 600 Ω and that a Butterworth design is desired.

First, you find the geometric center frequency:

 $f_0 = \sqrt{f_1 f_2} = \sqrt{(800 \text{ Hz})(1200 \text{ Hz})} = 980 \text{ Hz}$

Next, compute the two pairs of geometrically related stop-band frequencies:

$$
f_a = 900 \text{ Hz} \qquad f_b = \frac{f_0^2}{f_a} = \frac{(980 \text{ Hz})^2}{900 \text{ Hz}} = 1067 \text{ Hz}
$$
\n
$$
f_b - f_a = 1067 \text{ Hz} - 900 \text{ Hz} = 167 \text{ Hz}
$$
\n
$$
f_b = 1100 \text{ Hz} \qquad f_a = \frac{f_0^2}{f_b} = \frac{(980 \text{ Hz})^2}{1100 \text{ Hz}} = 873 \text{ Hz}
$$
\n
$$
f_b - f_a = 1100 \text{ Hz} - 873 \text{ Hz} = 227 \text{ Hz}
$$

Choose the pair of frequencies that gives the more severe requirement-227 Hz.

Follow....

Follow...

Actual high-pass filter

Final bandpass filter

Active Notch Filter - Wide-Band

Basic wide-band notch filter

For example, if you need a notch filter to have -3-dB points at 500 and 5000 Hz and at least -15 dB at 1000 and 2500 Hz,

Narrow-Band Notch Filter

Improved notch filter

Suppose that you want to make a "notch" at f_0 = 2000 Hz and desire a -3-dB bandwidth of Δf_{BW} = 100 Hz. To get this desired response, do the following: First determine the Q:

 $Q = \frac{\text{``notch'' frequency}}{-3 \text{-dB bandwidth}} = \frac{f_0}{\Delta f_{BW}} = \frac{2000 \text{ Hz}}{100 \text{ Hz}} = 20$

$$
R_1 = \frac{1}{2\pi f_0 C} \qquad \text{and} \qquad K = \frac{4Q - 1}{4Q}
$$

Now arbitrarily choose R and C; say, let $R = 10$ k and $C = 0.01 \mu F$. Next, solve for R_1 and K:

$$
R_1 = \frac{1}{2\pi f_0 C} = \frac{1}{2\pi (2000 \text{ Hz})(0.01 \text{ }\mu\text{F})} = 7961 \text{ }\Omega
$$

$$
K = \frac{4Q - 1}{4Q} = \frac{4(20) - 1}{4(20)} = 0.9875
$$

Anti-aliasing

- \triangleright 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 $T(=1/fs)$ between such samples must be less than one-half the period of the highest-frequency component fmax in the signal
	- In other words: you must sample at least twice the rate of the maximum frequency in your signal to prevent aliasing $(Fs \geq 2FMAX)$

◦ The sampling rate FS=2FMAX is called the Nyquist rate

Anti-aliasing

- The effects of aliasing can also be observed on the frequency spectrum of the signal
- . In the figures below
	- F, appears correctly since $F_1 \leq F_2/2$
	- \bullet F₂, F₃ and F₄ have aliases at 30, 40 and 10Hz, respectively
	- You can compute these aliased frequencies by folding the spectrum around $F₈/2$ or with the expression

FIGURE 117.5 Spectral of signal with multiple frequencies.

FIGURE 117.6 Spectral of signal with multiple frequencles after sampled at $f s = 100$ 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

