Recap from last time
- Analog circuit design
- Noise
- Microphone interface

Objectives
- Active low pass filter
- Nyquist Theorem and aliasing
- Speaker amplifier

Looking at noise, observe system with known input

**Magnitude**

**Type**

1) DVM AC mode => good for quantitative level
2) Scope
   - Peak-peak => rough measure of quantitative level
   - Shape => type of noise

3) Spectrum analyzer
   - Signal to noise ratio
   - Type
db_{FS} = 20\log_{10}(V/1.5)

Filter types

Analog
  - LPF
  - BPF
  - HPF

Digital
  - Extremely flexible

**Butterworth Filters**

2 pole Butterworth analog filter

1) select the cutoff frequency, \( f_c \)

2) divide the two capacitors by \( 2\pi f_c \) (let \( C_{1A}, C_{2A} \) be the new capacitor values)

\[
C_{1A} = \frac{141.4}{2\pi} \mu F \\
C_{2A} = \frac{70.7}{2\pi} \mu F
\]

3) locate two standard value capacitors (with the 2/1 ratio) with the same order of magnitude as the desired values
   let \( C_{1B}, C_{2B} \) be these standard value capacitors, let \( x \) be this factor

\[
C_{1B} = \frac{C_{1A}}{x} \\
C_{2B} = \frac{C_{2A}}{x}
\]

4) adjust the resistors to maintain the cutoff frequency

\[
R = 10k\Omega \times x
\]

Two pole Butterworth low pass analog filter.

![Two pole Butterworth Low Pass Analog Filter Diagram](image_url)

Capacitor specification

- low leakage (impedance),
- accuracy (tolerance),
- low temperature coefficient
- temperature range,
- voltage range
- frequency range

**Extremely High Quality Capacitors**

- Teflon, polystyrene
- Polypropylene, 1,2,3,5%.

**Medium Quality Capacitors** 5%, 10%, 20% tolerance

- Class 1, C0G ceramic, 5%, 30ppm/°C, ±0.3% over -55 to 125 °C
- Class 2, X7R ceramic, 10%, ±15% over -55 to 125 °C
- Class 3, Z5U ceramic, 20%, 22 to -56% over 10 to 86 °C

**Performance Tip:** If you choose standard value resistors near the desired values, you will save money and the circuit will still be a Butterworth filter. The only difference is that the cutoff frequency will be slightly off from the original specification.

**********Show LPF.XLS**************

Show TI FilterPro

by Jonathan W. Valvano
Quantitative DAS (thermometer in EE445L)
- range ($r_X$)
- resolution ($\Delta x$)
- precision ($n_x$ in alternatives)
- frequencies of interest ($f_{min}$ to $f_{max}$)

Qualitative DAS (sound recording in EE345M)
- "sounds good"
- "looks pretty"
- "feels right"

Other qualitative DAS’s involve the detection of events.
- true positive (TP)
  \[ \text{baby stops breathing and apnea monitor detects it} \]
- false positive (FP)
  \[ \text{baby is breathing OK but apnea monitor alarms} \]
- false negative (FN)
  \[ \text{baby stops breathing but monitor does not alarm} \]

Prevalence = \( \frac{TP + FN}{TP + TN + FP + FN} \)
Sensitivity = \( \frac{TP}{TP + FN} \)
Specificity = \( \frac{TN}{TN + FP} \)
PPV = \( \frac{TP}{TP + FP} \)
NPV = \( \frac{TN}{TN + FN} \)

Using Nyquist Theory to Determine Sampling Rate.
Voltage quantizing
- precision \( n_z = 2^n \)

Time quantizing

by Jonathan W. Valvano
Nyquist theory states that if the signal is sampled at $f_s$, then the digital samples only contain frequency components from 0 to $\frac{1}{2}f_s$.

Conversely, if the analog signal does contain frequency components larger than $\frac{1}{2}f_s$, then there will be an aliasing error. Aliasing is when the digital signal appears to have a different frequency than the original analog signal.

The choice of sampling rate, $f_s$, is determined by the maximum useful frequency contained in the signal.

$$f_s > 2f_{\text{max}}$$

A low pass analog filter may be required to remove frequency components above $0.5f_s$. A digital filter can not be used to remove aliasing.

Analog Filter

Let the gain of the analog filter be $G_3 = |H_3(s)|$. Then the system should pass, with little error as seen by the ADC, for signal frequencies between $f_{\text{min}}$ and $f_{\text{max}}$.

$$\Delta Z$$ properly represented

undetectable

aliased

To prevent aliasing => no measurable signal above $0.5f_s$.

Speaker interface

Need an audio amp to connect DAC output to 32Ω speaker

Be careful to limit voltage and power to speaker

by Jonathan W. Valvano
$0 < V < 5V$
$P < 200 \text{ mW}$

$0.2 \text{ W} > \frac{(V_{O1}-V_{O2})^2}{32\Omega}$ \quad (V_{O1}-V_{O2}) < 2.5V

$0.2 \text{ W} > \frac{(V_{O1}-V_{O2})^2}{8\Omega}$ \quad (V_{O1}-V_{O2}) < 1.2V

Look up maximum current for your board
Measure it with an external +5V supply and a current meter

Choose $R_f, R_i$ so $2*R_f/R_i$ is less than 1

Mount the speaker in a box
http://www.lalena.com/Audio/FAQ/Speaker/

Design choices
ADC bits
Sampling rate
Cheaper cables and electronics cause more noise
Analog filters
Cheaper microphone and speakers introduce more noise

Refer errors to the system specification
Quantitative system errors cause inaccuracy
Qualitative system errors cause it to sound bad