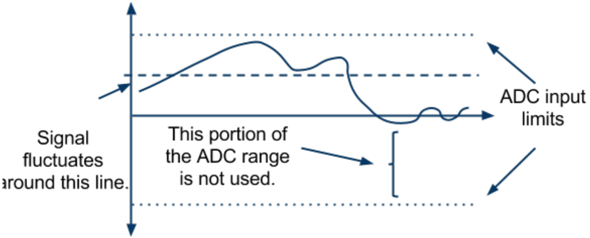
CS122A: Intermediate Embedded and Real Time Operating Systems

Jeffrey McDaniel

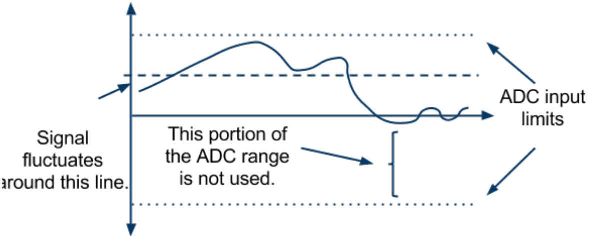
University of California, Riverside

Bias Correction



- Many signals are centered around 0

Bias Correction



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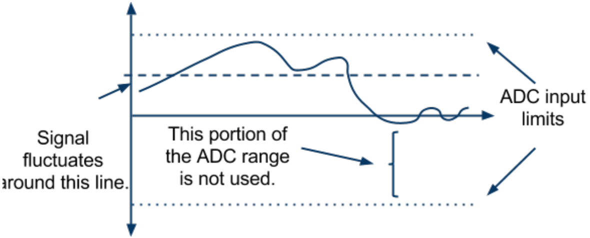

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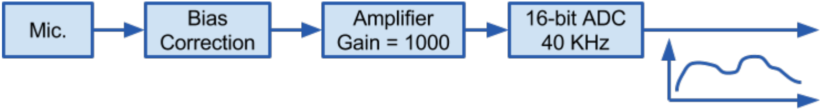

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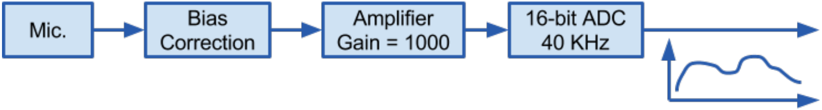

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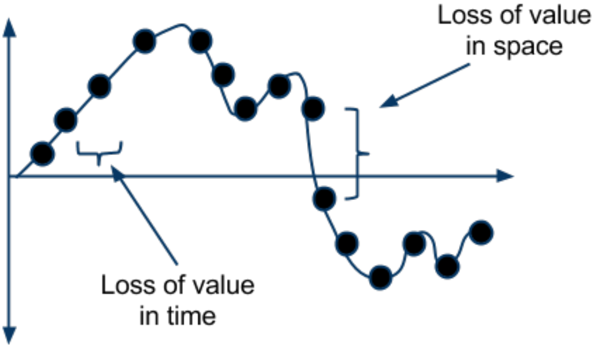
- Sometime signals fluctuate around a non-zero value
- These signals are said to have **bias**
- This bias needs to be corrected or it will lead to a waste during ADC
- A bias correction circuit maintains the bias, and outputs the signal minus the bias

Quantization



- We have been using **signal conditioning** to fully utilize our ADC input range

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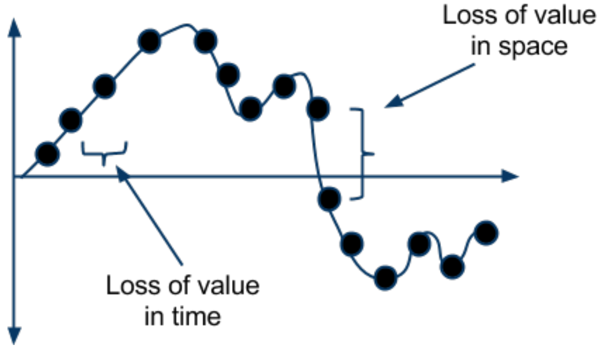

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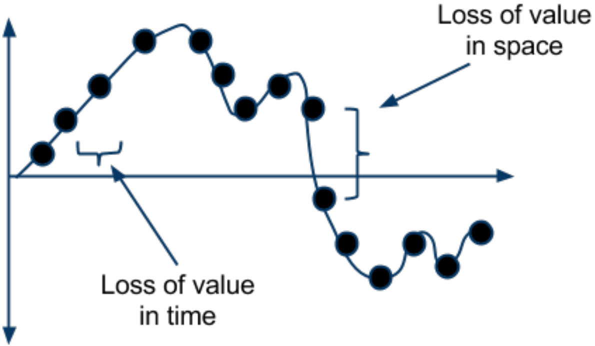

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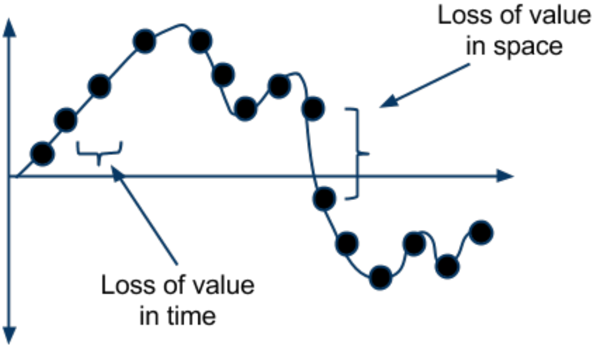

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- How precise does ADC need to be though?
- We have been pulling discrete data out of a continuous data set
- Information is lost based on how often we sample (**sampling rate**) and how much information we use

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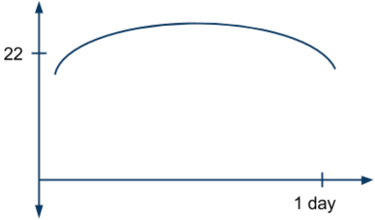
- **Resolution**, or bit width of the ADC, governs how much data is lost in space
- **Sampling rate** governs how much information is lost in time, higher sampling rate less lost information

Resolution

- The conversion range shows the min and max voltage values that can be represented
- The gap shows how precisely the voltage can be represented
- The **dynamic range** of a device is 2^N (where N is the bit width) shows the range precision capabilities of the ADC
- Dynamic range is presented in the decibel logarithmic scale

$$
\begin{aligned}
dB &= 10 * log(range^2/gap^2) \\
&= 10 * log(range/gap)^2 \\
&= 20 * log(range/gap) \\
&= 20 * log(2)^N \\
&= N * 20 * log(2) \\
&= N * 6.02
\end{aligned}
$$

Sampling Rate



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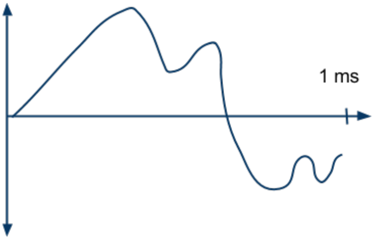

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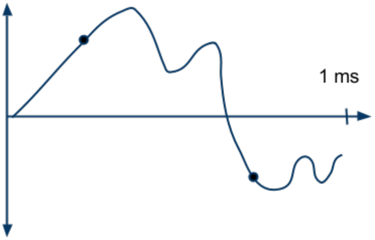

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- If the signal has a very quick rate of change you'll need to sample faster
- If the sampling is too slow you can't reconstruct the signal

Sampling Rate

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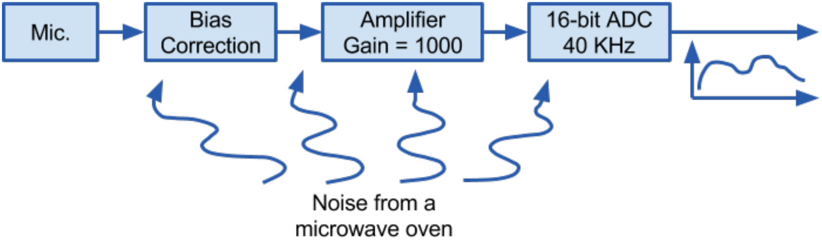
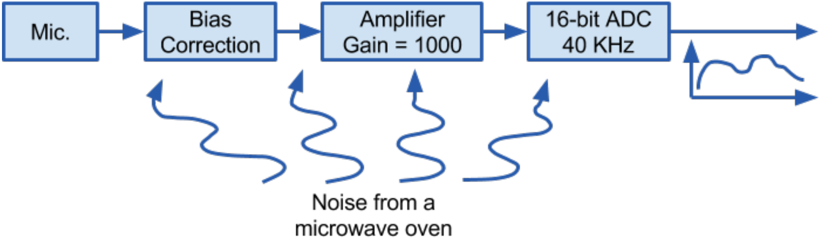
  - The sampling rate but be at least $2 * f$ Hz
- This rate is also known as the **Nyquist** rate

Aliasing



- All systems are subject to **noise**

Aliasing



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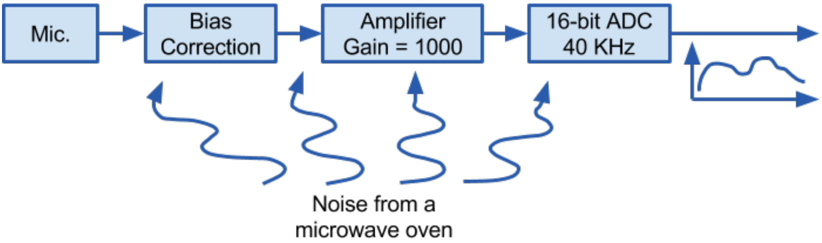

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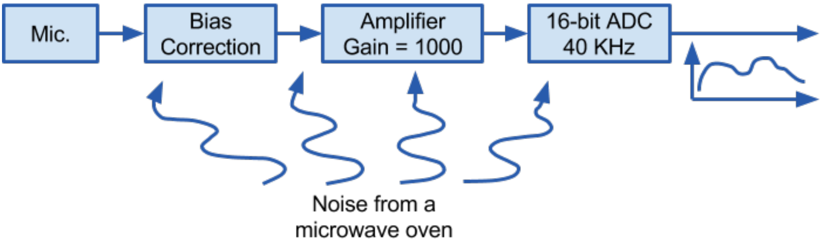

- All systems are subject to **noise**
- Noise is any unwanted signal from the environment that makes it's way into the signal path
- The **signal-to-noise ration (SNR)** is the amount of signal present vs. noise
- **Aliasing** is the presence of a false or unexpected signal at the time of sampling

Low Pass Filter

- A **Low Pass Filter**, or **anti-aliasing filter**, can be used to remove the aliasing

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- This specific frequency is called the **center frequency**

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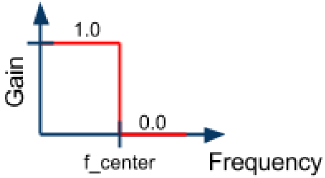

- A perfect low pass filter would have a gain of 1 for everything below the center frequency (f_{center}) and 0 for everything above

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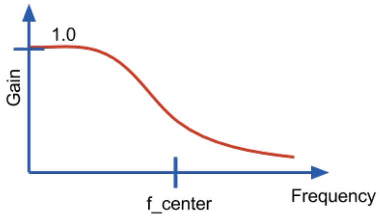
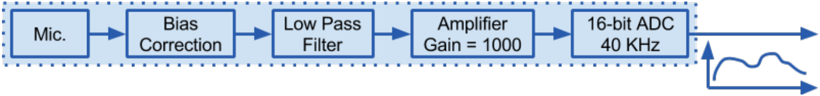


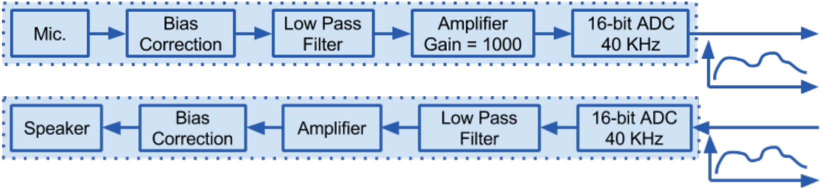
- A perfect low pass filter would have a gain of 1 for everything below the center frequency (f_{center}) and 0 for everything above
- A more realistic filter would lower the gain as it approaches f_{center}

Complete DSP circuit



- The complete DSP circuit reading a signal from the mic to the chip

Complete DSP circuit



- The complete DSP circuit reading a signal from the mic to the chip
- In the complete playback path Filtering is done before amplification, and the bias correction is done last before the signal reaches the actuator

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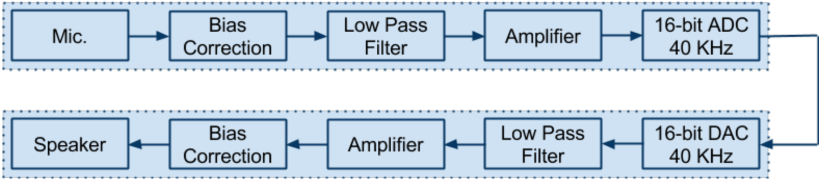

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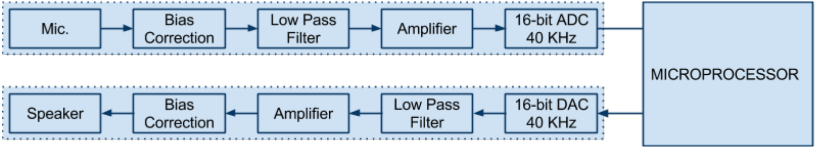
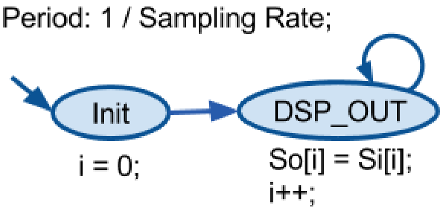


- The complete DSP circuit reading a signal from the mic to the chip
- In the complete playback path Filtering is done before amplification, and the bias correction is done last before the signal reaches the actuator
- The circuits can be connected to create a complete circuit analog input to analog output
- The signal can be sent through a microprocessor for additional processing

Digital Processing



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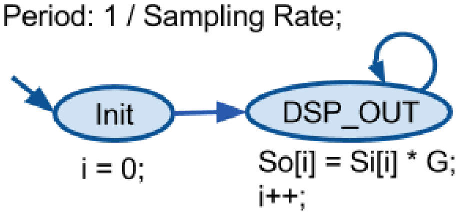

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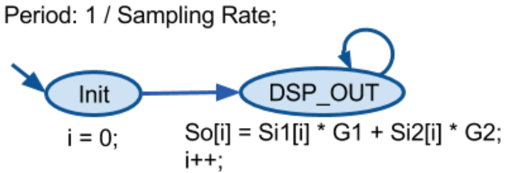

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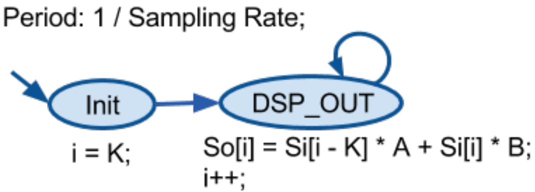

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- We can **scale** the output
- Or mix two outputs
- Even echo previous inputs