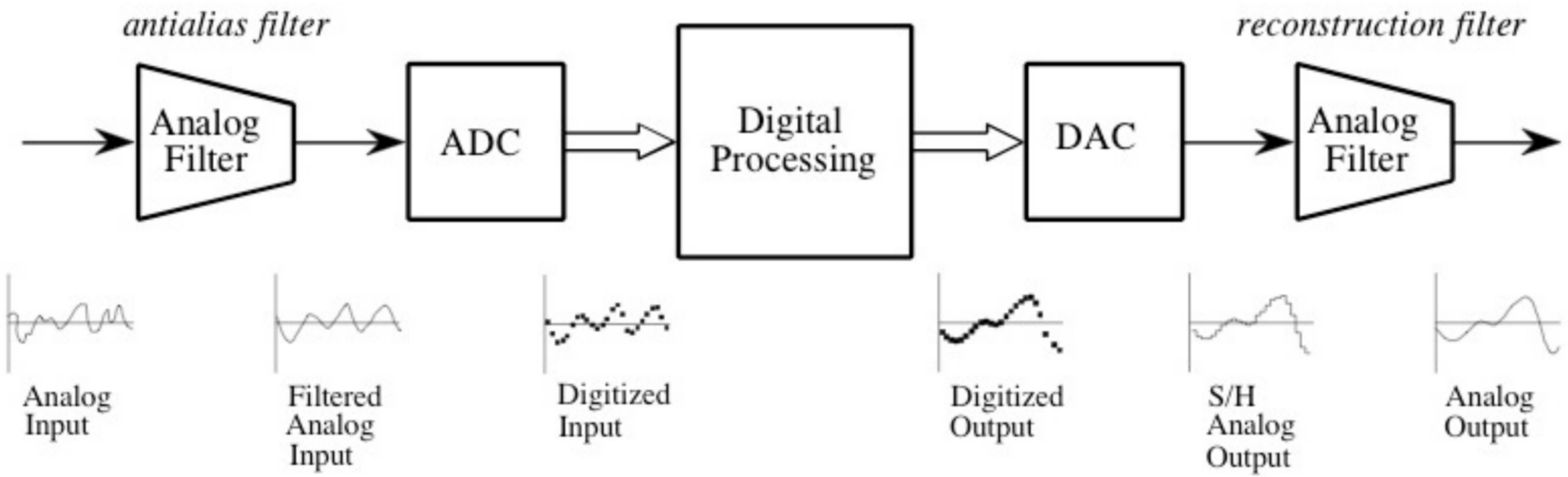
DAC

https://guitar.ucsd.edu/mauricio/courses/mae143a-W2011/lectures/8sampling.pdf

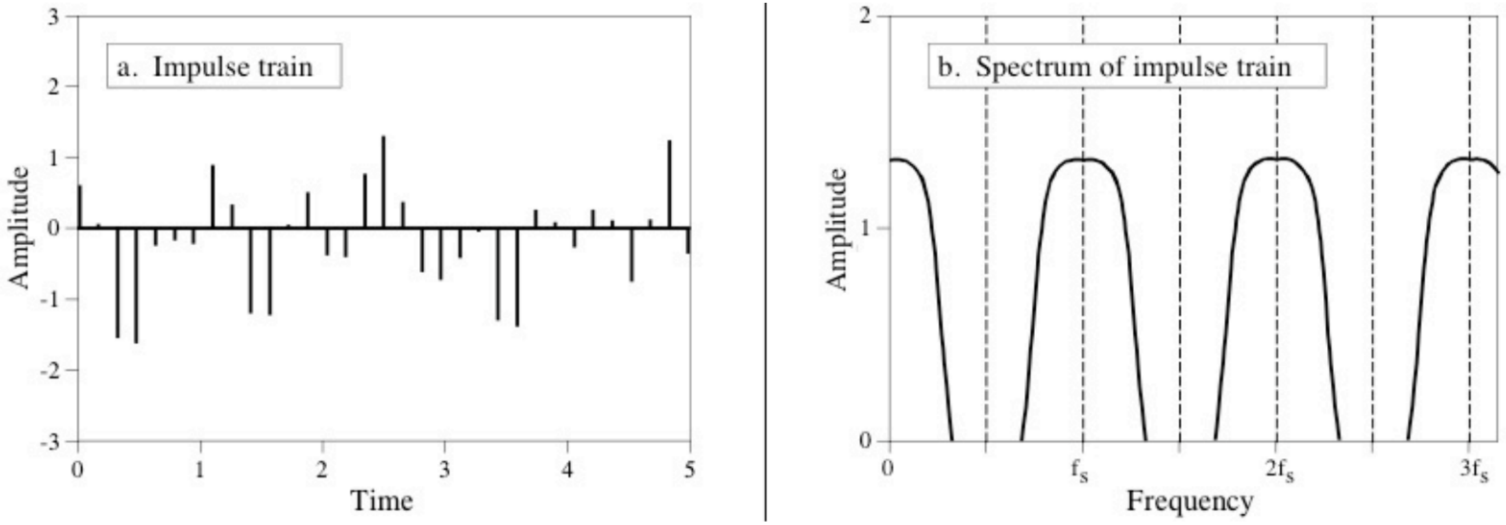


Analog electronic filters are used to comply with the sampling theorem. The filter placed before ADC is an antialias filter. It removes frequencies higher than half the sampling rate. The filter placed after the DAC is a reconstruction filter. It may include a correction for the zeroth-order hold.

Digital-to-analog conversion

The DAC reverses the ADC process:

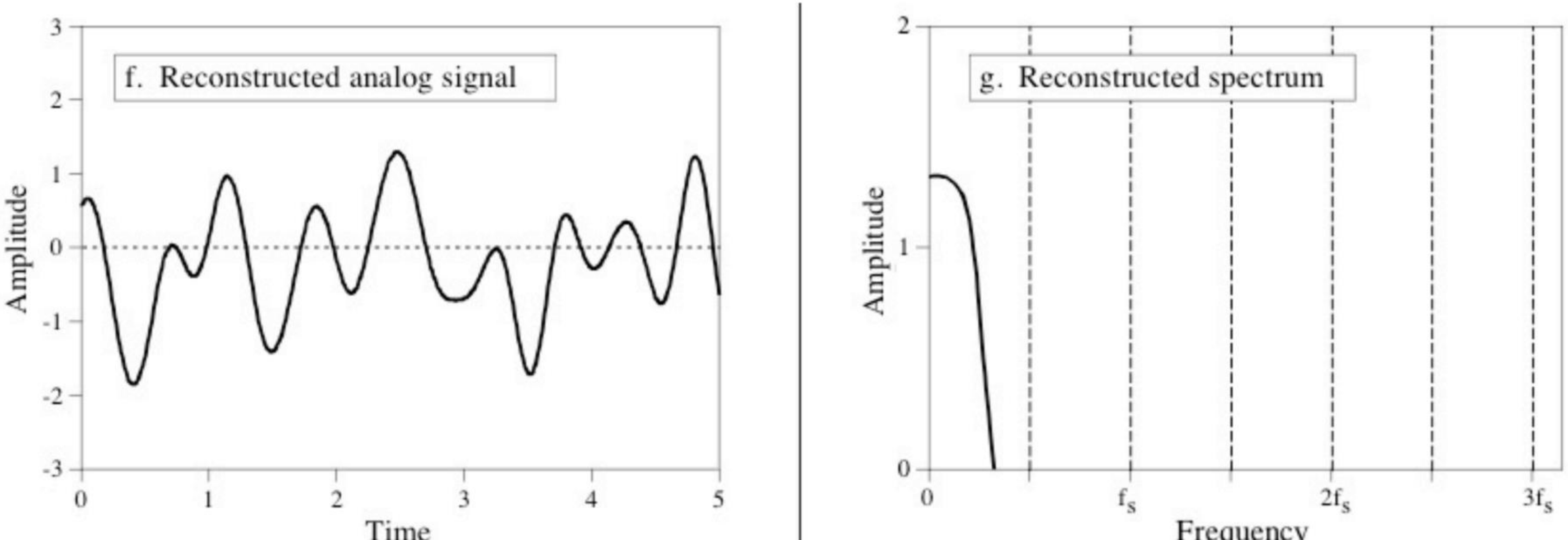
(1) It decodes the signal making a conversion from a bit sequence to an impulse train:



Digital-to-analog conversion

(2) The signal is reconstructed with an electronic low-pass filter to remove the frequencies about ½ the sampling rate

After filtering the impulse train with a such a low pass filter, we would obtain:



Practical operation uses only a finite number of samples. Many techniques can be used to approximately reconstruct the signal.

One such technique is a **zero-order holder**.