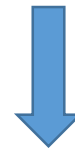
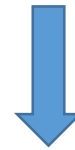


PCM

Pulse Coded Modulation

A joint effort of
Barbara Costantini and Sean Pethybridge

PCM uses
Analog to Digital Conversion (ADC)
3 main functions



PCM uses analog to digital conversion

SAMPLING



QUANTIZATION

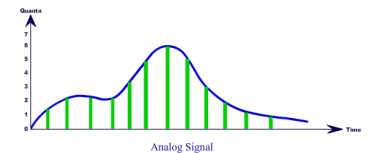
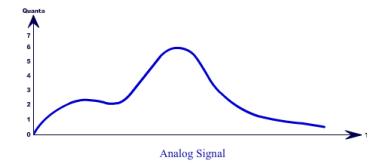


ENCODING

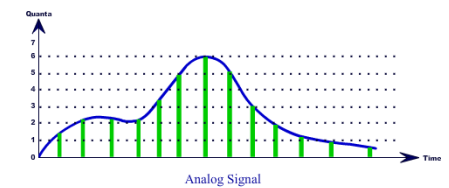
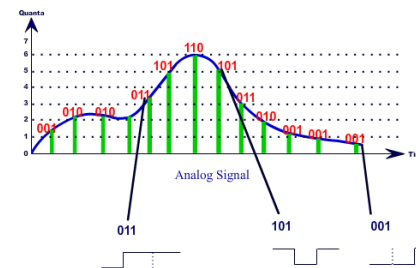


Time Discretization

Amplitude Discretization

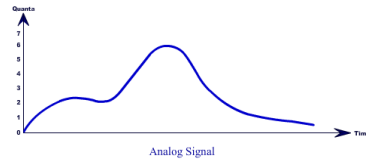


Digital conversion of the quantized values



Match each definition and graph with the corresponding function

PCM uses analog to digital conversion



SAMPLING



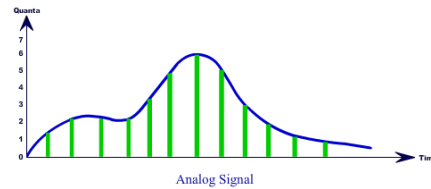
QUANTIZATION



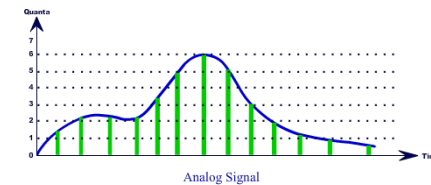
ENCODING



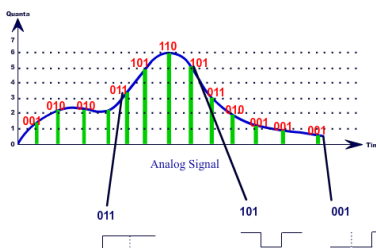
Time Discretization



Amplitude Discretization



Digital conversion of the quantized values



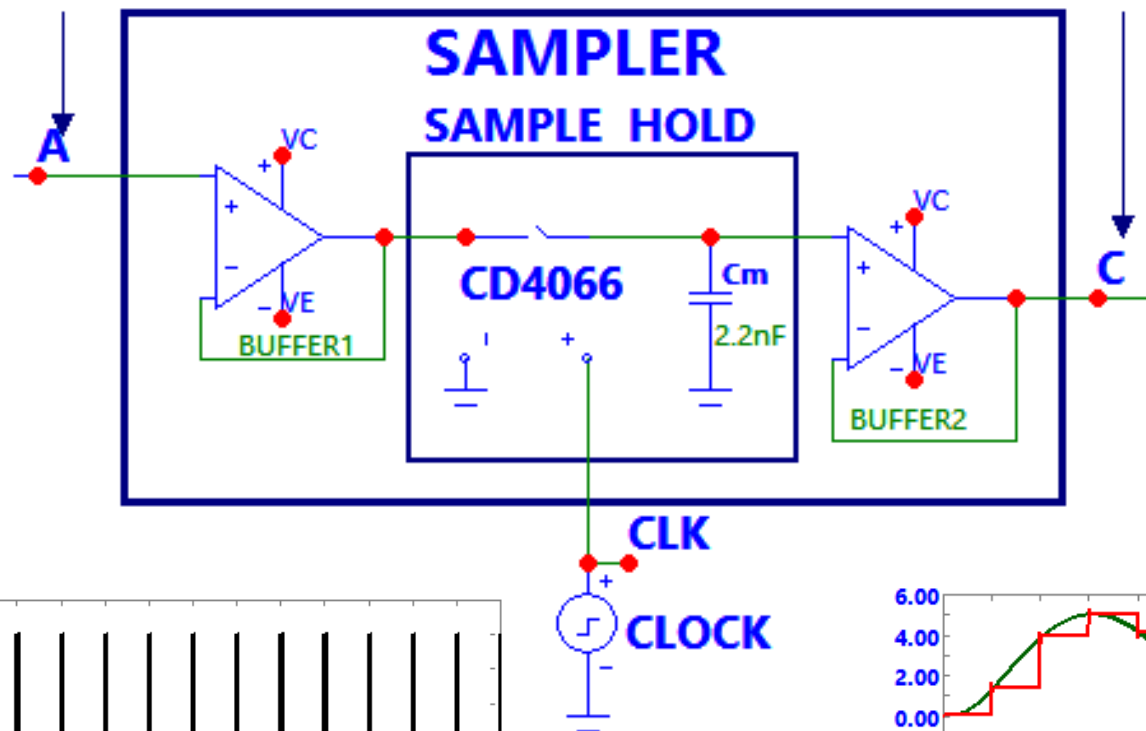
[Take a look at the whole process of digitalization](#)

How can we built a sampler?

ANALOG SIGNAL

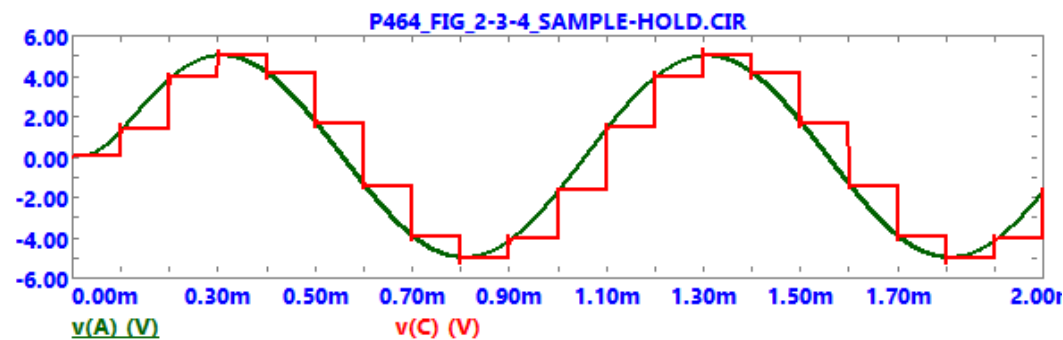
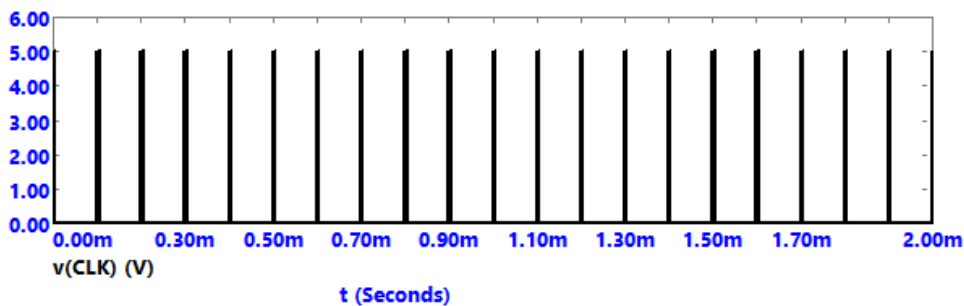
Electronic switch
driven by a clock

SAMPLED SIGNAL

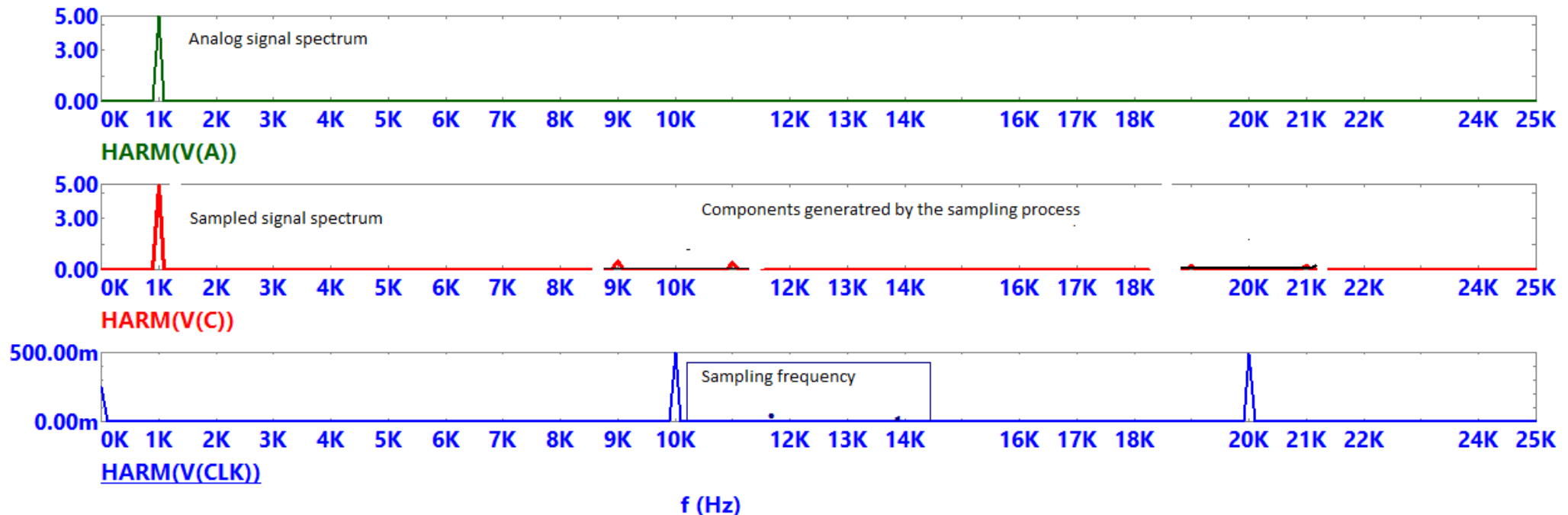


Memory circuit=
capacitor which
maintains the level of
signal constant during
the conversion (hold)

Clock frequency
=sampling rate
=sampling frequency



Now run simulations on Microcap to study the spectra of the signals in the sampling process



Find out the relationship between the frequency of the component generated during the sampling and

- the signal input frequency
- the sampling frequency

The relationship is

$$f_{\text{component}} = f_{\text{sampling}} \pm f_{\text{signal}}$$

Take a look at the spectrum of the sampled signal:

Can you guess what electronic device can be used to recover the original signal at the receiver?

What happens if the frequency of the input signal is increased up to 5.5 kHz?

The issue is:

Which sampling frequency
should be used to
properly recover the signal at the receiver?

This question is answered by the fundamental theorem

in DIGITAL SIGNAL PROCESSING

The NYQUIST-SHANNON sampling THEOREM

Discover more about Claude Shannon [on this video](#)

The NYQUIST-SHANNON sampling THEOREM

states that

Given a signal whose bandwidth is equal to B ,

if the signal is sampled at a sampling rate greater than $1/2B$

$$f_s > 1/2B$$

Then it is always possible to reconstruct the original shape of the signal by interpolating its samples

If the Nyquist-Shannon theorem is not satisfied

i.e. $f_s < 1/2B$



ALIASING

Foldover distortion

ALIASING EFFECT

