



# Signal Processing Basics & Filtering

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<https://gitlab.com/surligas/sdr-tutorial>  
<https://gitlab.com/surligas/gr-tutorial>

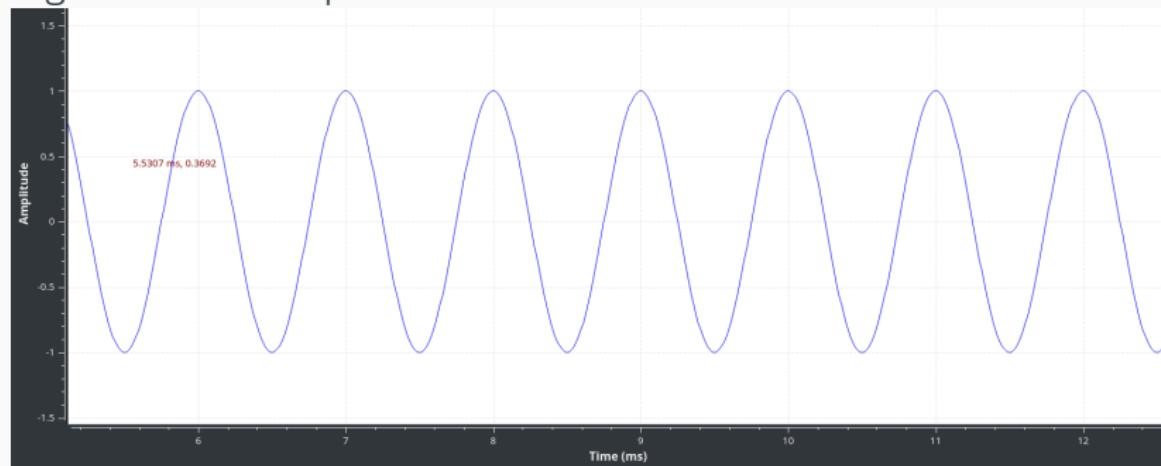
Manolis Surligas - surligas@csd.uoc.gr

# Signal Representation

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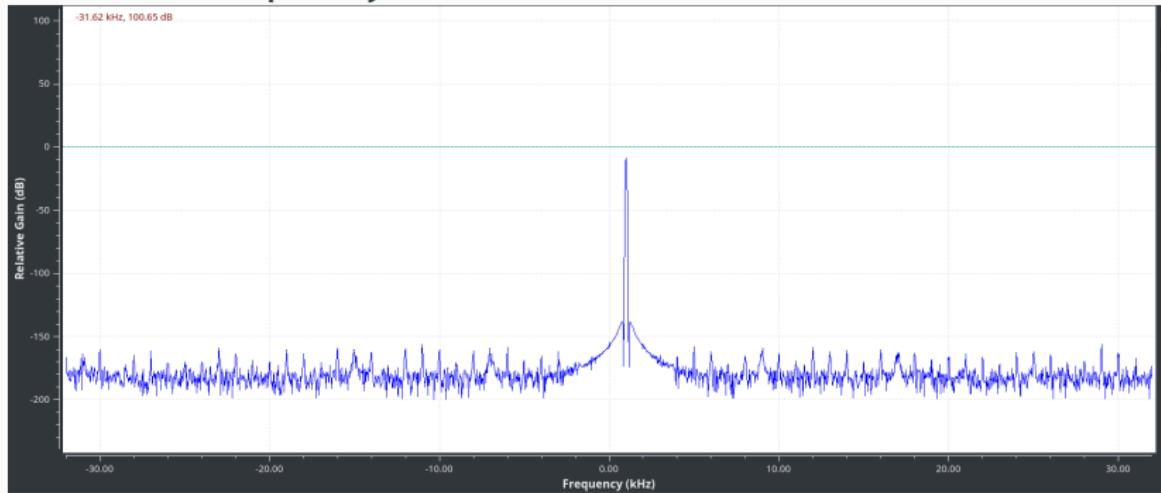
# Signal Representation

Signals can be represented in the **time domain**



# Signal Representation

...or in the frequency domain



# Signal Representation

## Time domain:

- X-axis represents the time
- Y-axis represents the amplitude (e.g Volts)

## Frequency domain:

- X-axis represents the frequency
- Y-axis represents the power
- Commonly power is expressed in logarithmic scale  
 $(10 \times \log_{10}(x))$

# Signal Representation

## Frequency domain:

- For the transition from time to frequency domain, the Discrete Fourier Transform (DFT)  $X_k$  is used

$$\begin{aligned} X_k &= \sum_{n=0}^{N-1} x_n \cdot e^{-\frac{i2\pi}{N} kn} \\ &= \sum_{n=0}^{N-1} x_n \cdot \left[ \cos\left(\frac{2\pi}{N}kn\right) - i \cdot \sin\left(\frac{2\pi}{N}kn\right) \right] \end{aligned}$$

- In simple words, the  $X_k$  transformation, converts the original signal into  $N$  cosines, each one with a frequency  $\frac{2\pi}{N}kn$  and amplitude  $x_n$
- For real signals the imaginary part is 0

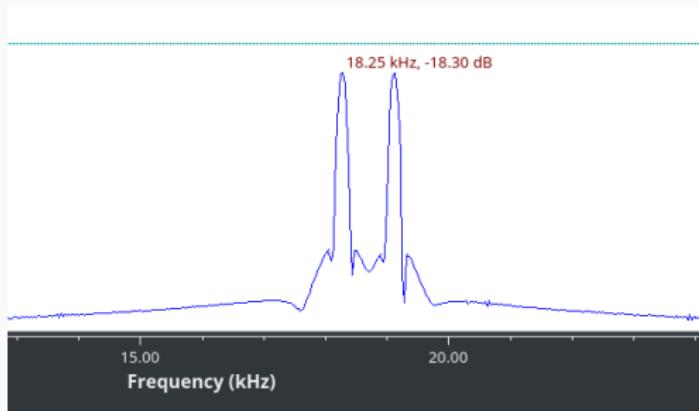
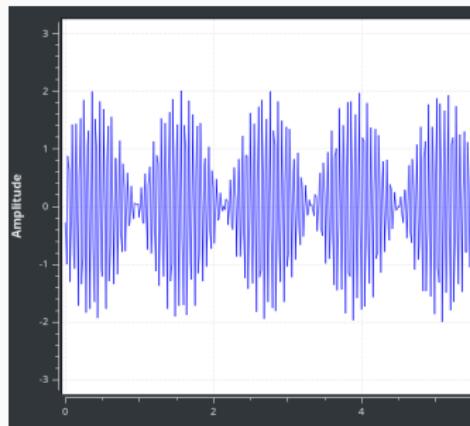
# Signal Representation

- Changing the signal in the time domain affects the frequency domain
- In most cases we use the frequency domain representation
- Different frequency components can be spotted very easily
- Signals can be spotted visually even with significant amount of noise

Use the **simple\_example.grc** flowgraph located at the **examples** directory to play with time and frequency domain representation

# Signal Representation

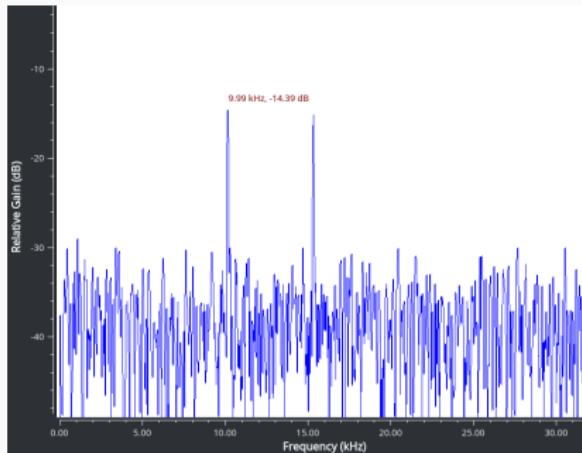
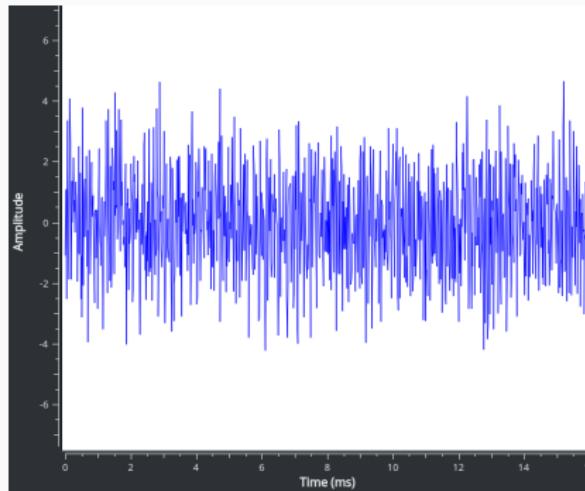
**Example 1:** Finding the frequency components of a mixed signal



The two different sine waves can be spotted quite easily in the frequency domain comparing to the time domain

# Signal Representation

**Example 2:** Finding the frequency components of a noisy mixed signal



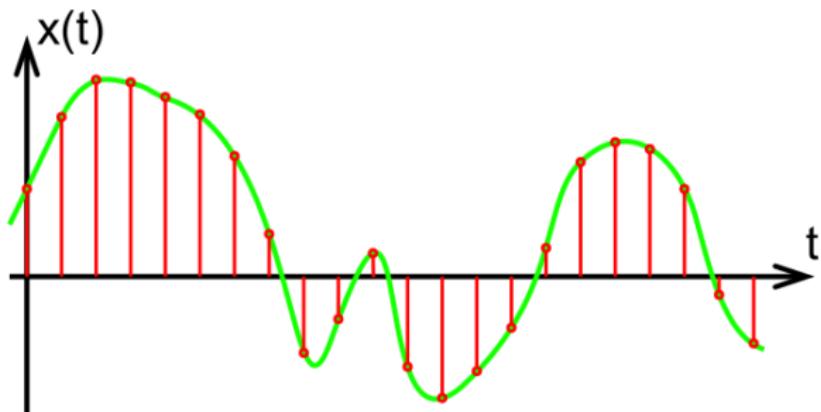
Even with the presence of significant noise, the signal components are distinguishable! (However this is not always the case 😞)

## Sampling & Quantization

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# Sampling Theory

**Sampling** is the process of converting a continuous-time signal into a discrete-time signal



## Nyquist Theorem

A real signal  $x(t)$  band-limited to  $B$  Hz can be reconstructed without loss of information iff the sampling frequency is greater than  $2 \times B$

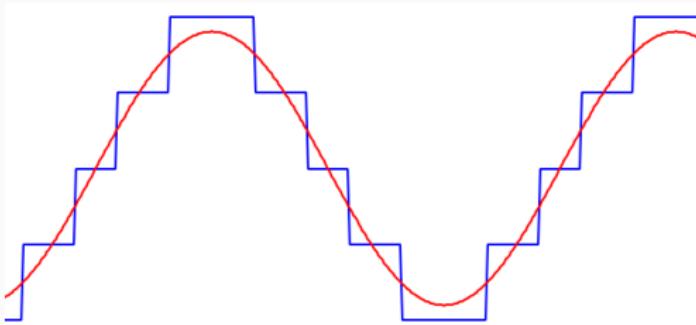
- A band-limited signal  $x(t)$ , is a signal that does not have spectral components above  $B$  Hz
- The minimum sampling frequency  $R = 2 \times B$  is called Nyquist frequency

# Sampling Theory

- In most cases there is a need to change the sampling rate of a signal
  - Hardware capabilities / restrictions
  - Processing maybe easier at specific sampling rates
  - Processing resources
- **Decimation** is the process of reducing the sample rate
- **Interpolation** increases the sampling rate
- Combining decimation and interpolation an arbitrary sampling rate can be achieved
- More about resampling on the next lecture!

# Quantization

**Quantization** converts the signal amplitude into digital form samples



An approximation of the Signal-to-Quantization Noise Ratio (SQNR) for an N-bit ADC/DAC is:

$$\text{SQNR} = 20 \log_{10}(2^N) \approx 6.02 \cdot N \text{ dB}$$

# Filtering

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## Why do we use filters?

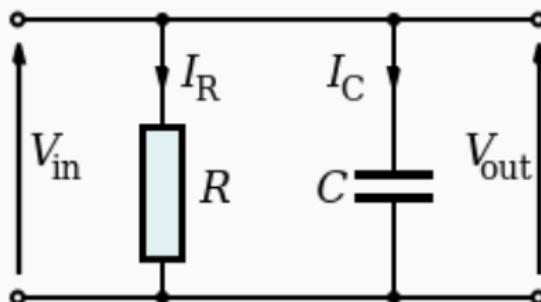
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- Filters are used to separate mixed signals
- Extract a specific band of interest
- Avoid undesired effects (aliasing, imaging e.t.c)
- Restore distorted signals
- Pulse shaping

# Filter types

## Analog Filters:

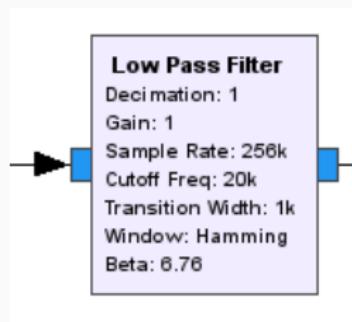
- Implemented using electronic components (resistors, capacitors, coils, transistors)
- Used in analog front-ends
- Standard performance
- Difficult to change their properties
- Application specific



# Filter types

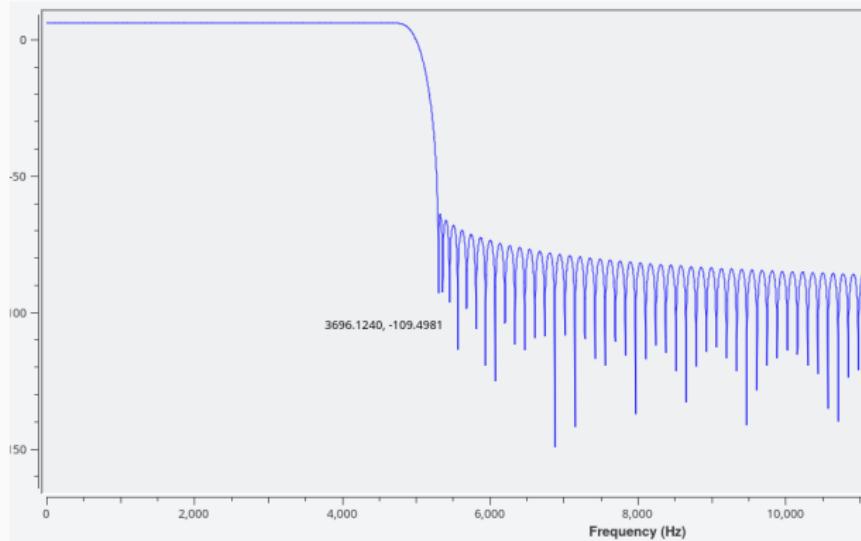
## Digital Filters:

- Implemented in software
- Variable performance depending the processing resources
- Change their properties relatively easily
- Can adapt to any application



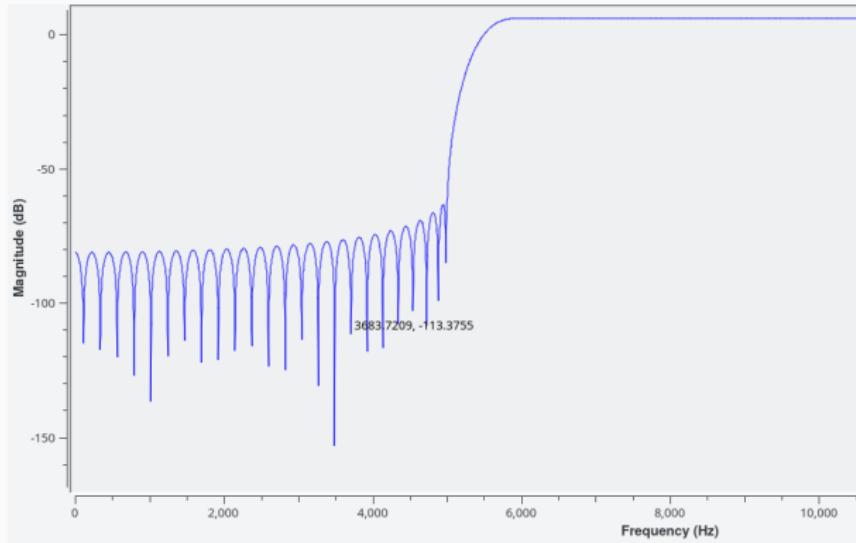
# Filter types

- Low Pass Filters
  - Frequencies above a threshold do not pass the filter



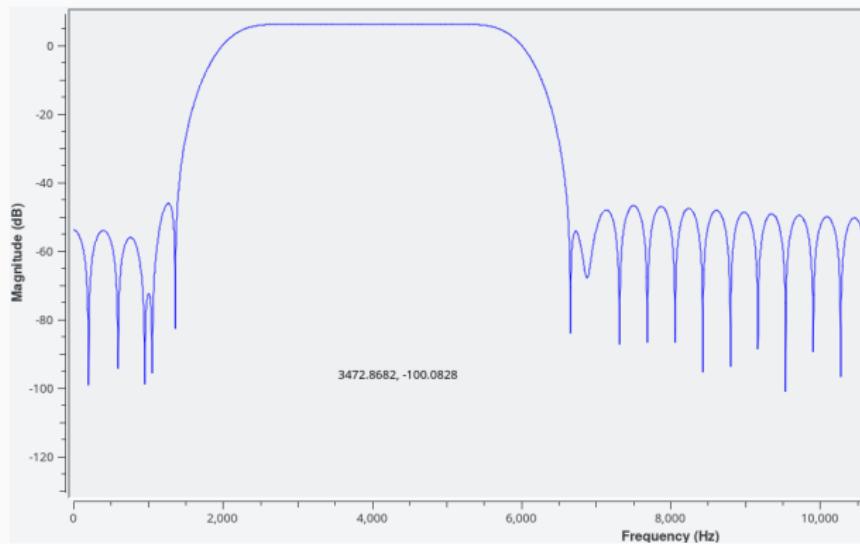
# Filter types

- High Pass Filters
  - Frequencies below a threshold do not pass the filter



# Filter types

- Band Pass Filters
  - Frequencies of a specific band pass the filter



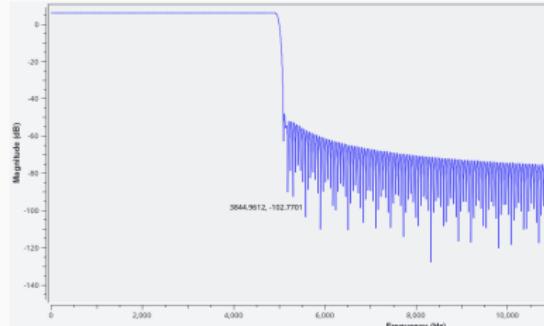
## Filter parameters

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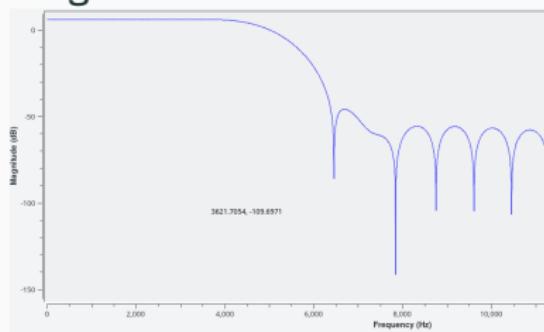
- **Cutoff frequency:** The frequency above (low pass filter) or below (high pass filter) which, the filter starts to attenuate the signal
- **Transition width:** Control how steep is the attenuation of the signal above (low pass filter) or below (high pass filter) the cutoff frequency

# Filter parameters

- Small transition width:



- Large transition width:



## Filtering and CPU resources

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- Transition width specifies the number of taps of the filter
- Steep filters have more taps than relaxed ones
- Designing a filter is always a balance between RF performance and CPU utilization
- Sampling rate plays a key role too

## Example

- Assume a sampling rate of 1 MSPS
- We are interested in a range of only 10 kHz
- The requirements demand a very steep filter and we have limited CPU resources (e.g RPi3)
- How???

## Example

- Assume a sampling rate of 1 MSPS
- We are interested in a range of only 10 kHz
- The requirements demand a very steep filter and we have limited CPU resources (e.g RPi3)
- How???
- Use a relaxed first stage relaxed filter
- Decimate the signal by a factor of e.g 50
- Apply a second stage steep filtering

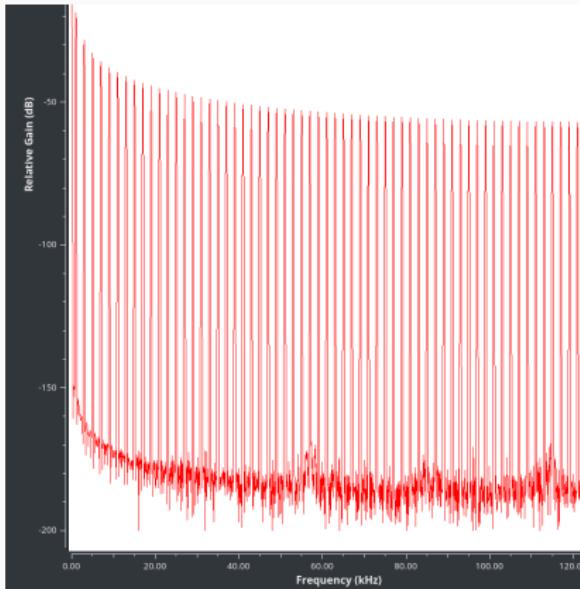
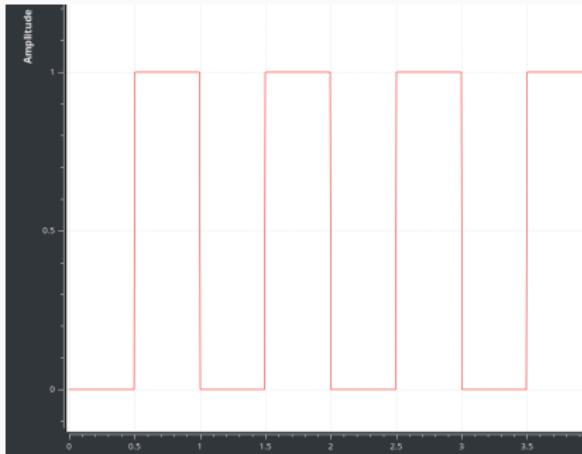
## Pulse Shaping Filtering

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- Filters can be used also for pulse shaping
- The goal is to minimize spurious signals and increase the spectral efficiency
- Spurious signals can be generated by extreme transitions of the source signal
- A simple square wave generated from a bit stream has an infinite bandwidth → non practical to transmit

# Pulse Shaping Filtering

- A simple square wave generated from a bit stream has an infinite bandwidth → non practical to transmit
- A square wave is an infinite sum of sine waves



# Pulse Shaping Filtering

- Commonly for shaping we use the Root Raised Cosine filter (RRC)
- Smooths the extreme transitions thus reducing the spurious

