

Wavelet Transform and its Applications

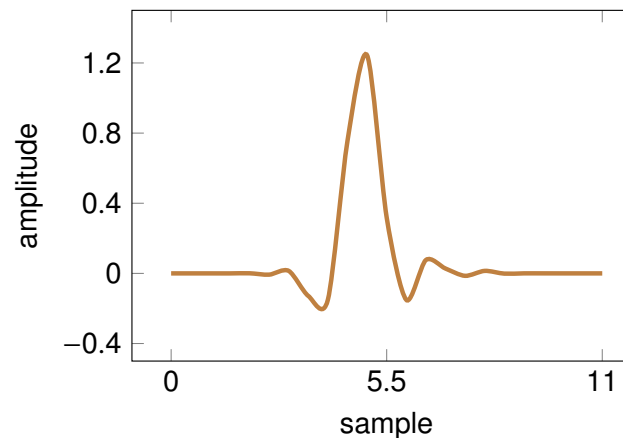
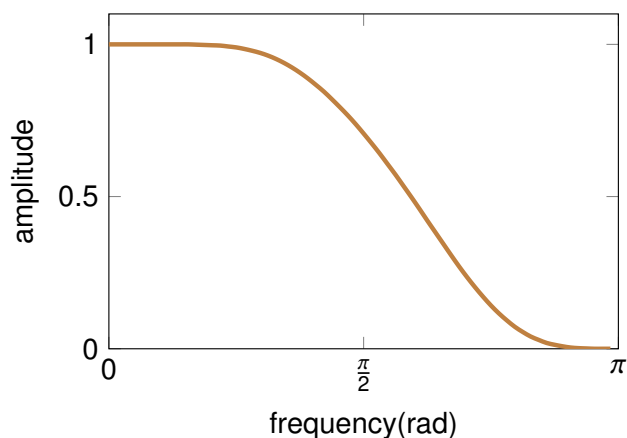
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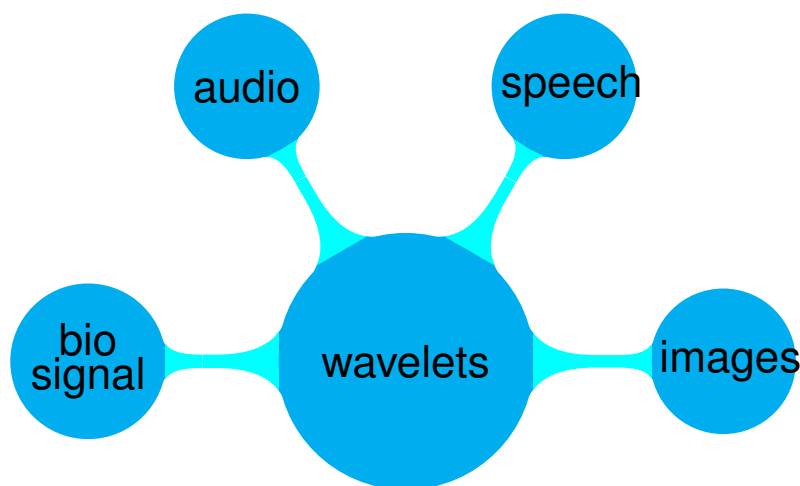
2024



Wavelet Transform and its Applications

- - - Introduction - - -

- ▶ **Introducing ourselves in class... time to socialize!** 😊 🧐 😊
- ▶ **Comments on syllabus, bibliography, schedule, grades and related details**
- ▶ **Course overview**

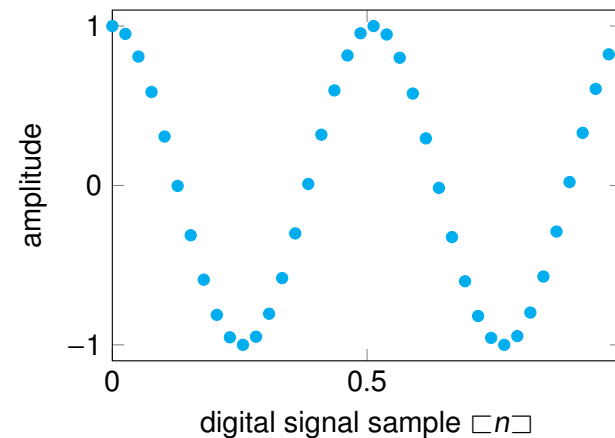
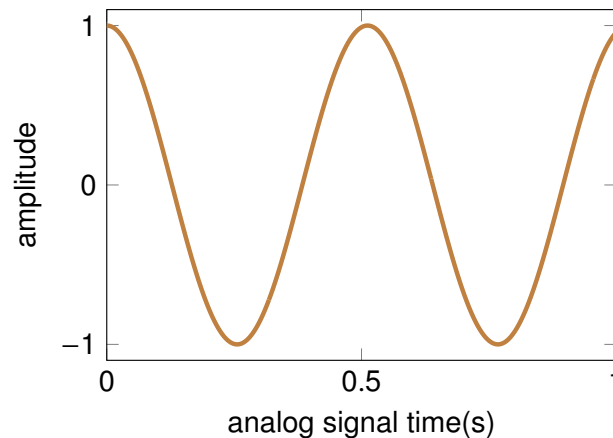


- ▶ **Classroom Activity:** you are now requested to comment on the impression you are having on this course introduction. Do you feel encouraged? Are you afraid of anything? What will be the main obstacle for you, in your opinion?

Wavelet Transform and its Applications

- - - Signal Processing Basics - - -

- ▶ **Signals**
- ▶ **Analog Signal** versus **Digital Signal**



- ▶ Usually, for a better visualization, digital signals are depicted as continuous curves
- ▶ **Computers** basically handle **Digital Signals**:
 - ▶ Sampling Theorem (Nyquist's criterion): analog signal is sampled at S_r samples per second
 - ▶ Quantization

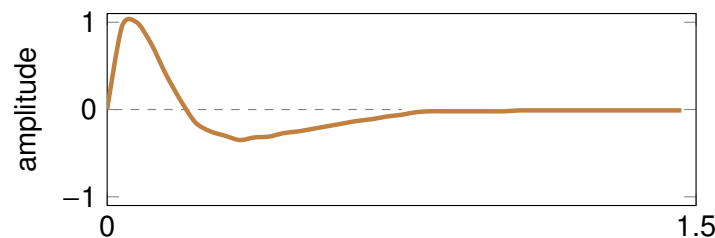
Wavelet Transform and its Applications

- - - Signal Processing Basics - - -

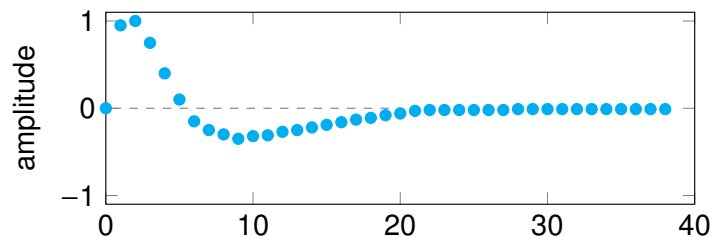
► Time Domain versus Frequency Domain

Fourier Analysis: any arbitrary function can be expressed as a sum of sinusoidal, i.e., “pure”, functions

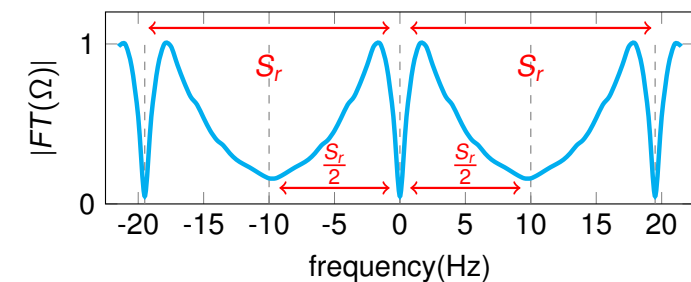
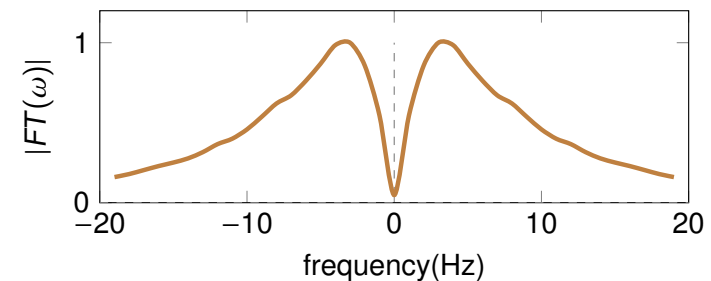
Frequency Domain: Fourier analysis provides different results depending on the signal type, i.e., continuous- or discrete-time



real continuous-time signal $x(t)$ where time is denoted in seconds



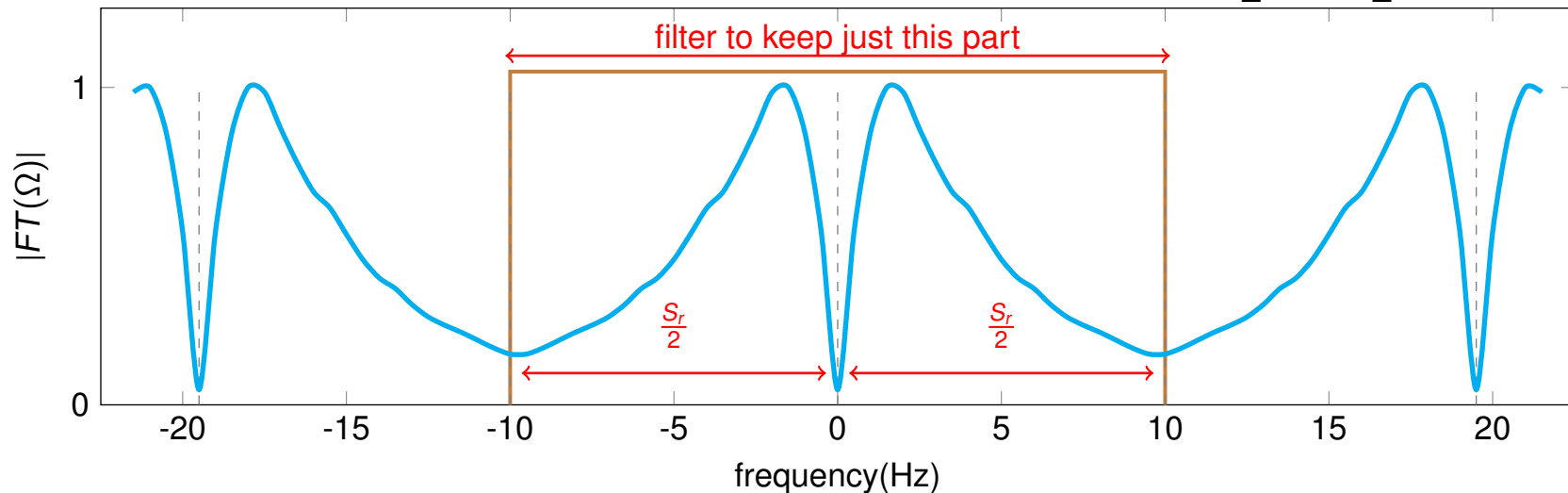
real discrete-time signal $x[n]$ sampled at S_r samples per second



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- ▶ to reconstruct the analog signal from its digital version, we have to filter the latter, keeping only the frequencies from $-\frac{S_r}{2}$ to $\frac{S_r}{2}$



- ▶ to reconstruct the original signal from its sampled version, S_r **should be at least twice the highest signal frequency component**
- ▶ **Short-Test 1::** comment on sampling and quantization. Why is it of fundamental importance in signal processing?