

Scientific computing for geophysical problems

2. Digital signal processing

Léonard Seydoux

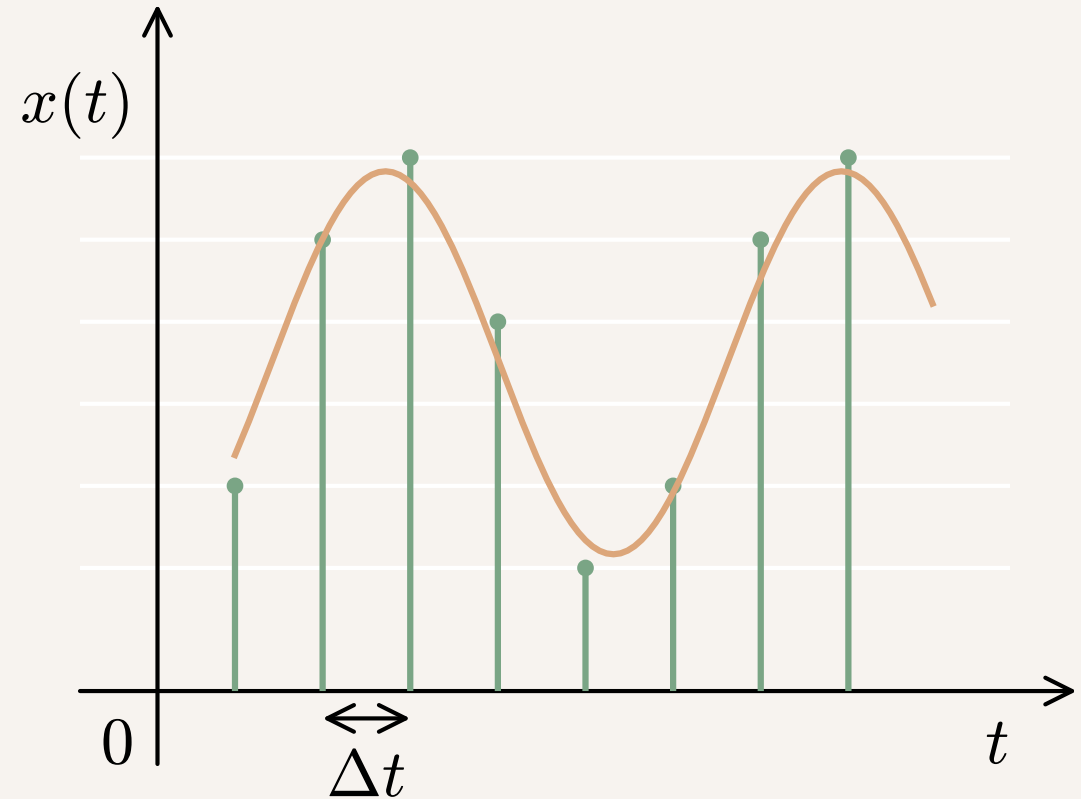
September 2024 at the [institut de physique du globe de Paris](#).



 [leonard-seydoux/scientific-computing-for-geophysical-problems](https://github.com/leonard-seydoux/scientific-computing-for-geophysical-problems)

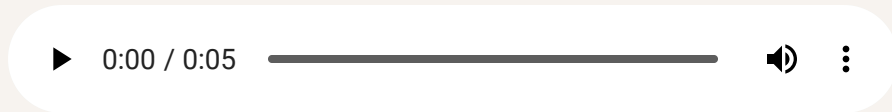
Analog and digital signals

- Analog signals $x(t)$ are continuous in time and amplitude
- Digital signals $x[n]$ are discrete in time and amplitude (sampling rate, vertical resolution)



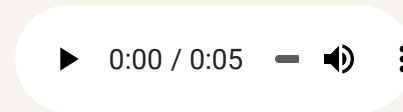
Digital signal processing in music

24 bits



correspond to 16,777,216
levels and 1.6MB

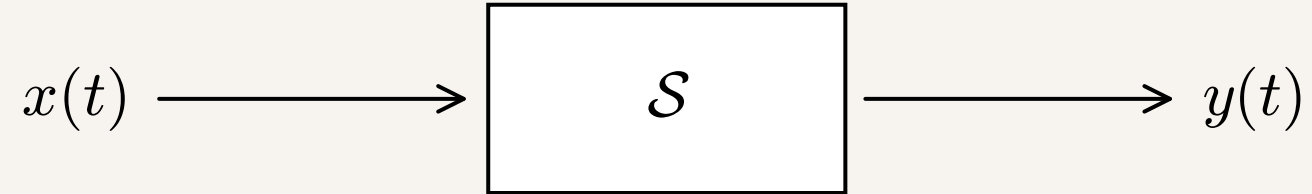
8 bits



correspond to 256 levels,
but only 95kB

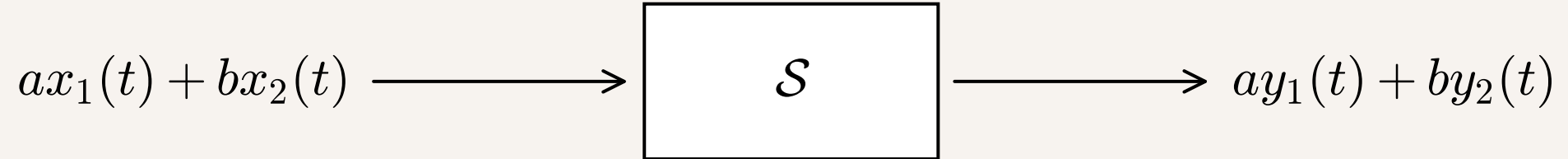
Signal processing as a system

A system \mathcal{S} maps an input signal $x(t)$ to an output signal $y(t)$



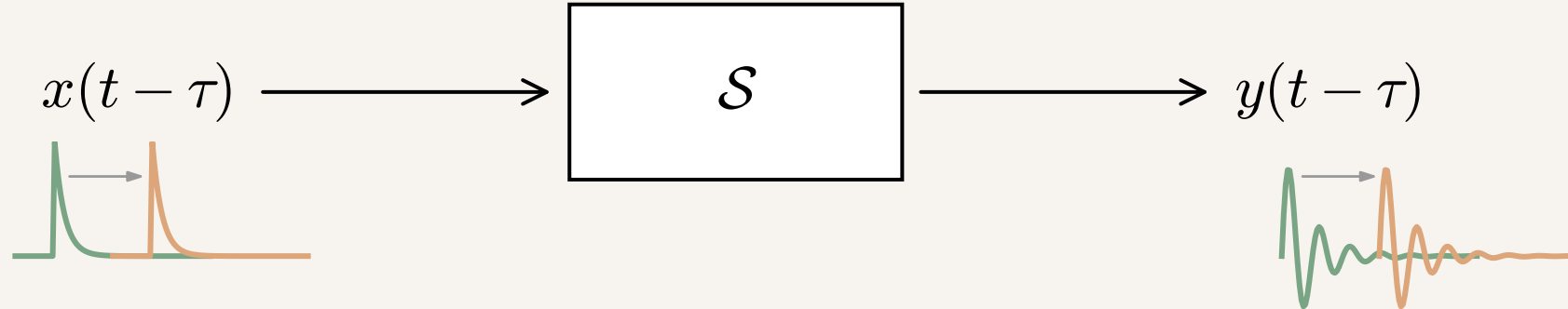
Linear systems

The system \mathcal{S} is **linear** if it satisfies the following property



Time-invariant systems

The system \mathcal{S} is **time-invariant** if it satisfies the following property



Linear and time-invariant systems

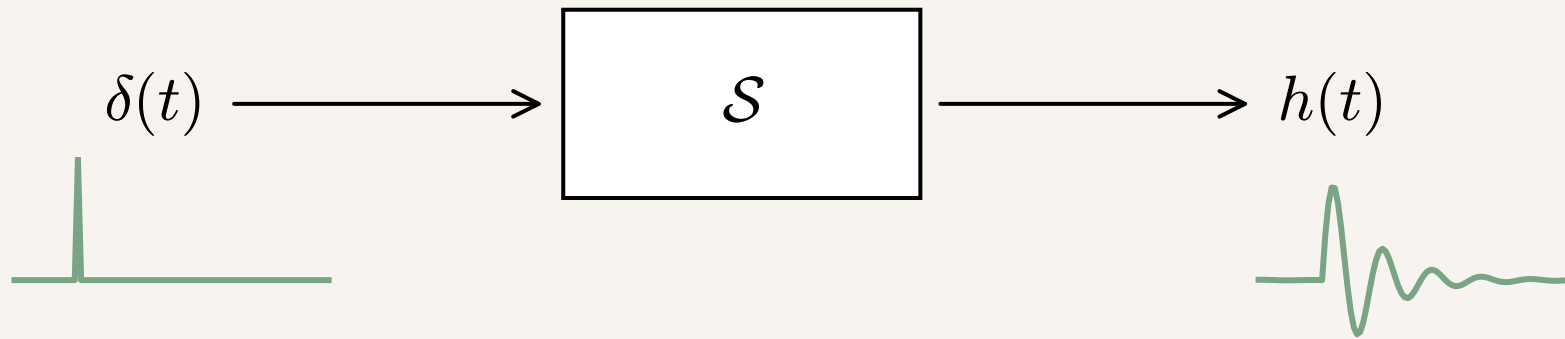
\mathcal{S} is linear and time-invariant (LTI) if it satisfies both properties:

- Linearity: $\mathcal{S}\{ax_1(t) + bx_2(t)\} = ay_1(t) + by_2(t)$
- Time-invariance: $\mathcal{S}x(t - \tau) = y(t - \tau)$

Linear and time-invariant systems are fully modeled by the **impulse response**

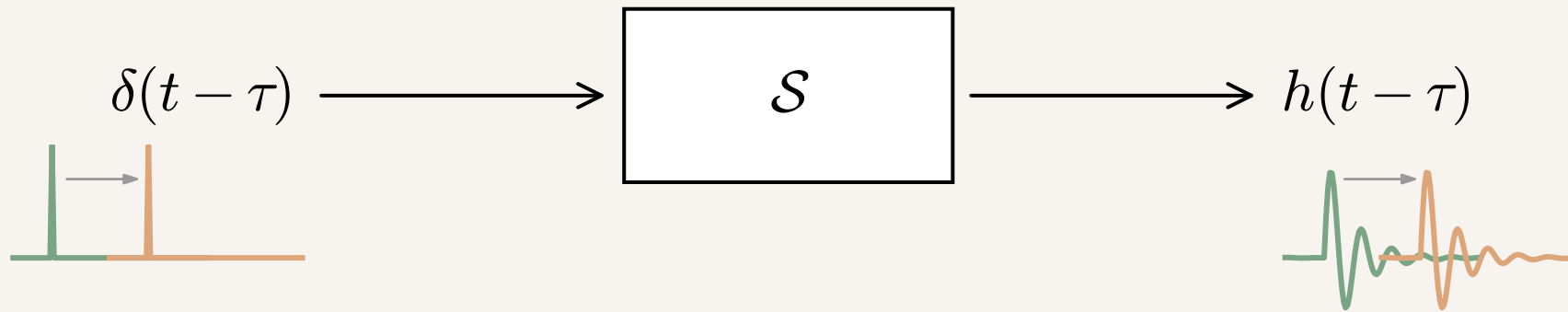
Impulse response

The impulse response $h(t)$ of a system \mathcal{S} is the output of the system when the input is the Dirac delta function $\delta(t)$



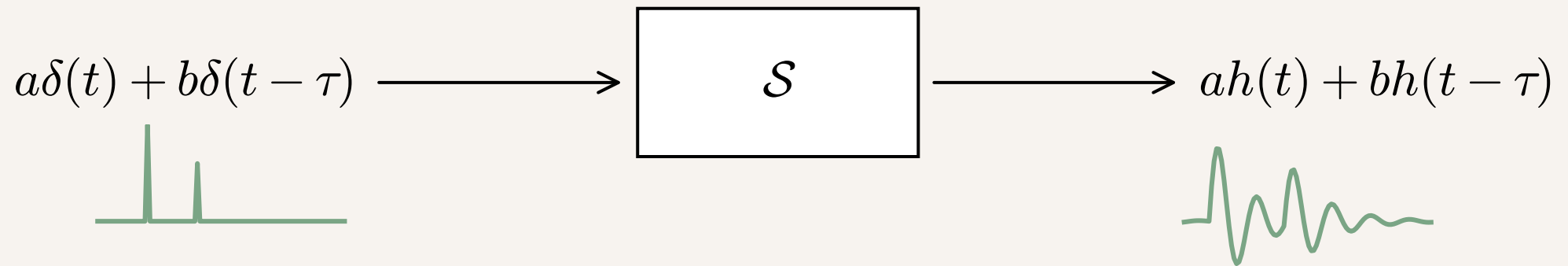
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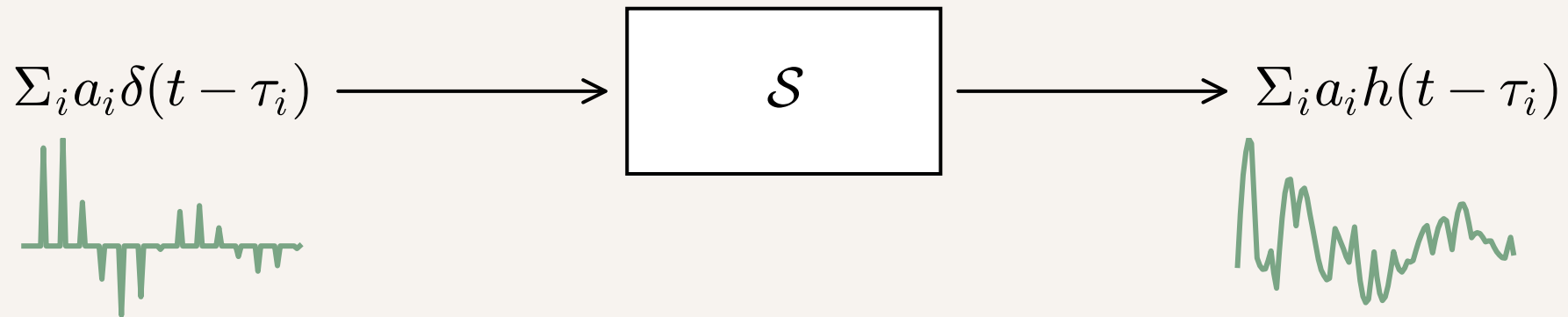
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Convolution

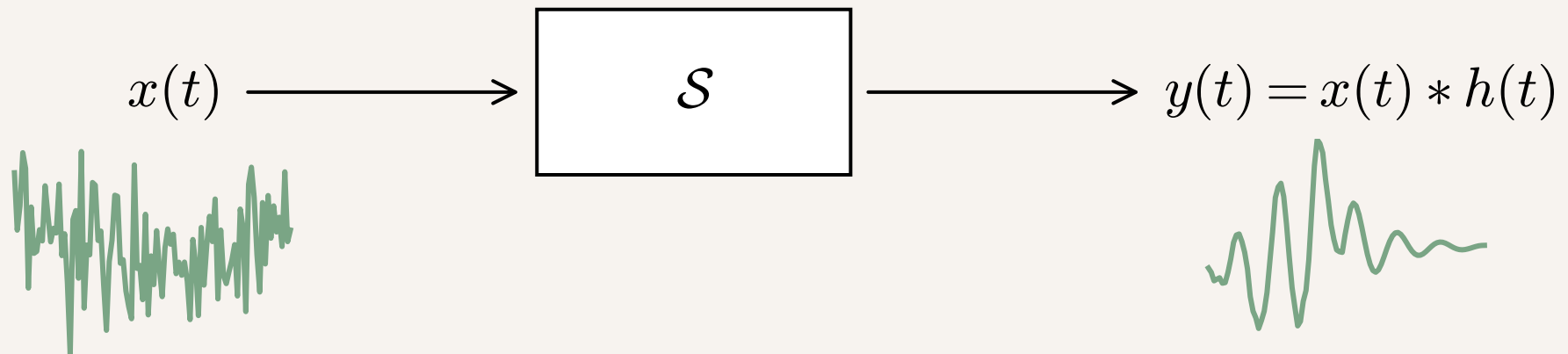
The impulse response $h(t)$ of a system \mathcal{S} is the output of the system when the input is the Dirac delta function $\delta(t)$



Convolution

The output signal $y(t)$ of a system \mathcal{S} is the convolution of the input signal $x(t)$ with the impulse response $h(t)$ such as

$$y(t) = \int_{-\infty}^{+\infty} x(\tau)h(t - \tau)d\tau$$



Example: adding reverb to a piano record

The impulse response
 $i(t)$ of a large hall

▶ 0:00 / 0:07 — 🔊 ⋮

The original piano
record $x(t)$ in a dry room

▶ 0:00 / 0:05 — 🔊 ⋮

Convolution theorem to get the piano record
 $y(t)$ as if it was played in the hall

$$y(t) = \frac{1}{T} \int_0^T x(\tau) i(t - \tau) d\tau$$

The piano record $y(t)$ as if
it was played in the hall

▶ 0:00 / 0:13 — 🔊 ⋮

Fourier transform (continuous)

The Fourier transform of a signal $x(t)$ is defined as

$$\hat{x}(f) = \int_{-\infty}^{+\infty} x(t) e^{-i\omega t} dt$$

- The Fourier transform is linear: $\mathcal{F}\{ax_1(t) + bx_2(t)\} = a\hat{x}_1(\omega) + b\hat{x}_2(\omega)$
- Convolution in time is multiplication in frequency: $\mathcal{F}\{x(t) * h(t)\} = \hat{x}(\omega)\hat{h}(\omega)$

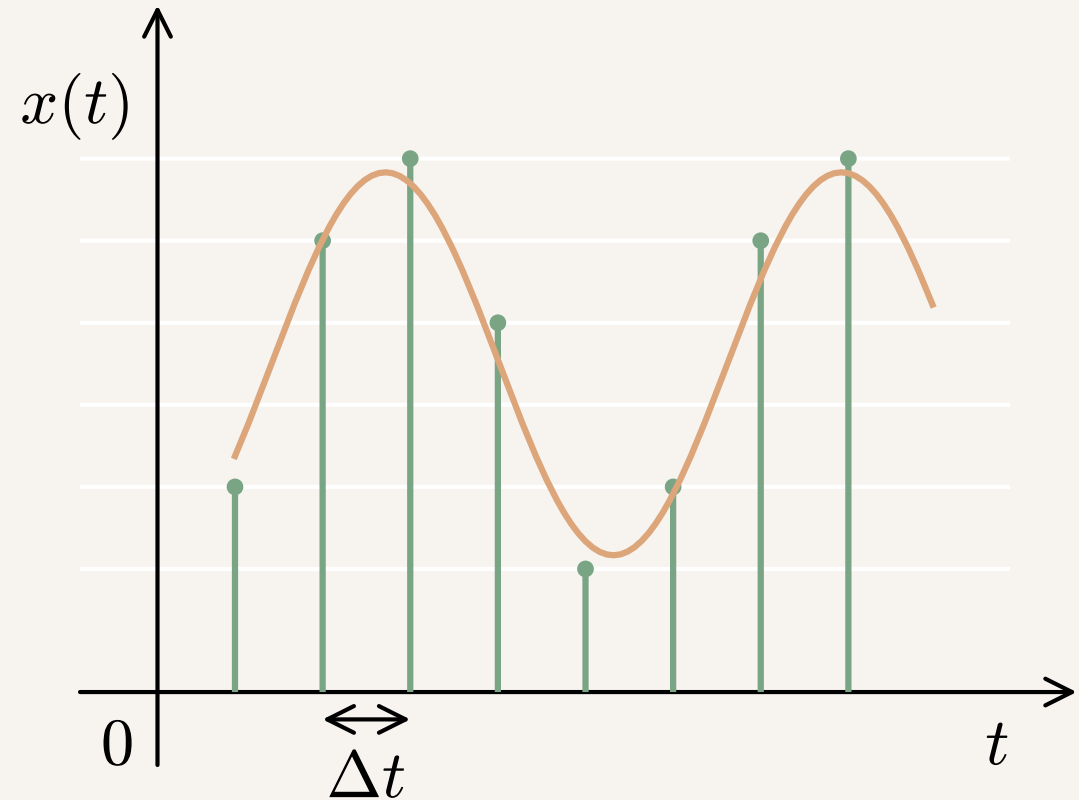
Sampling

The sampling of a continuous signal $x(t)$ at a every period Δt is defined as

$$x[n] = x(t)\mathbb{W}_{\Delta t}(t)$$

where the sampling rate is defined as

$$f_s = \Delta t^{-1}.$$



Discrete-time Fourier transform

The Discrete-time Fourier transform (DTFT) of a discrete signal $x[n]$ is defined as

$$\hat{x}(\omega) = \mathcal{F}x[n] = \sum_{n=-\infty}^{+\infty} x[n]e^{-i\omega n}$$

- The DTFT is linear: $\mathcal{F}\{ax_1 + bx_2\}(\omega) = a\hat{x}_1(\omega) + b\hat{x}_2(\omega)$
- Convolution in time is multiplication in frequency: $\mathcal{F}\{x * h\}(\omega) = \hat{x}(\omega)\hat{h}(\omega)$

Sampling and aliasing

In the spectral domain, sampling duplicates the spectrum at every multiple of the sampling rate ω_s .

$$\hat{x}(\omega) = \sum_{n=-\infty}^{+\infty} x(\omega - n\omega_s)$$

Sampling theorem: a signal can be reconstructed if $\omega_s \geq 2\omega_{max}$

