Sampling and aliasing

Kjartan Halvorsen

2021-06-30

Computer-controlled system
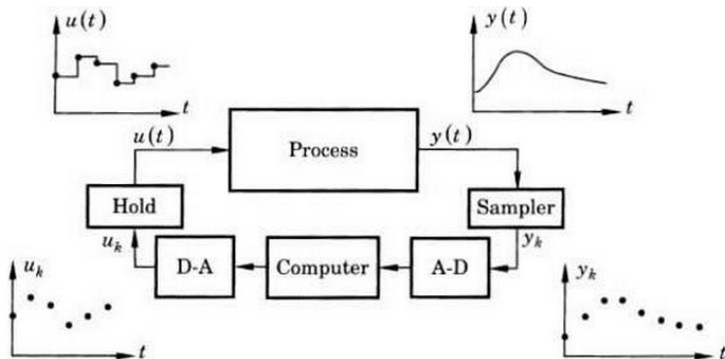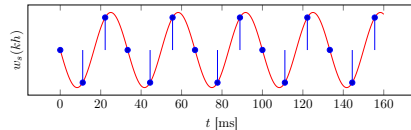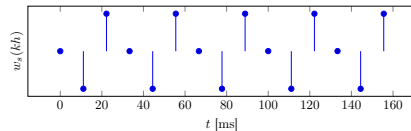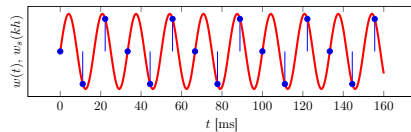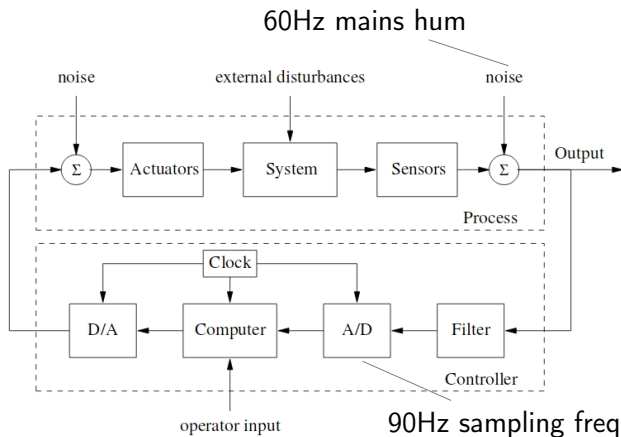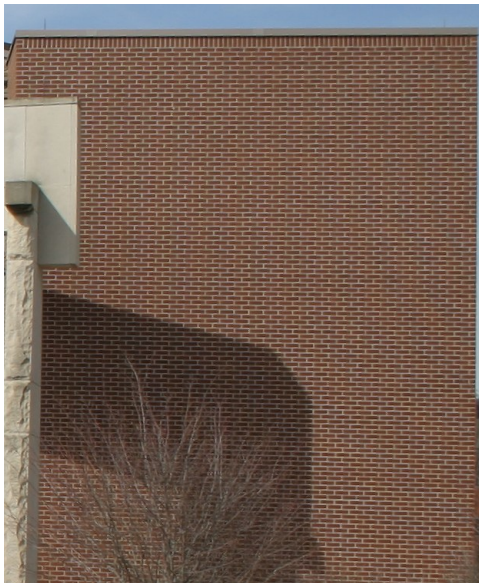

Figure 7.2 Relationships among the measured signal, control signal, and their representations in the computer.

Source: Åström & Wittenmark

Challenges with computerized control - aliasing

60Hz mains hum

noise

external disturbances

noise

Σ → Actuators → System → Sensors → Σ → Output

Process

Clock

D/A ← Computer ← A/D ← Filter
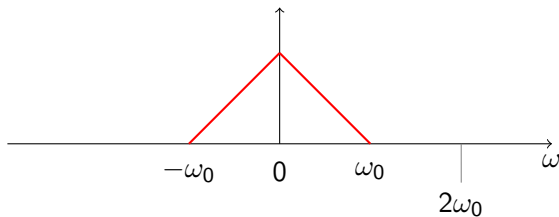
operator input

90Hz sampling freq

Controller

Spatial aliasing

The sampling theorem

Shannon and Nyquist:

A continuous-time signal with Fourier transform that is zero outside the interval $(-\omega_0, \omega_0)$ can be completely reconstructed from equidistant samples of the signal, as long as the sampling frequency is at least $2\omega_0$.
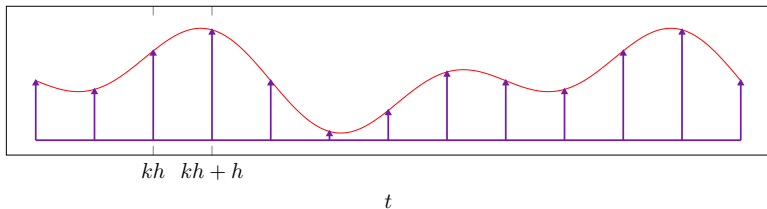
The impulse modulation model

The impulse train, a.k.a the Dirac comb:

$$m(t) = \sum_{k=-\infty}^{\infty} \delta(t - kh)$$

$$f_s(t) = f(t)m(t) = f(t) \sum_{k=-\infty}^{\infty} \delta(t - kh) = \sum_{k=-\infty}^{\infty} f(t)\delta(t - kh) = \sum_{k=-\infty}^{\infty} f(kh)\delta(t - kh)$$

$kh \quad kh + h$

t

Fourier transform of the sampled signal

The Fourier transform of f_s and the Fourier transform of f are related as

$$F_s(\omega) = \frac{1}{h} \sum_n F(\omega + n\omega_s).$$

Because the Fourier transform of the sampled signal equals the Fourier transform of the continuous-time signal repeated at every multiple of the sampling frequency and added, we get *frequency-folding* or *aliasing*.
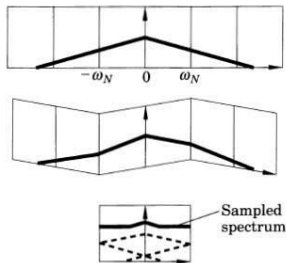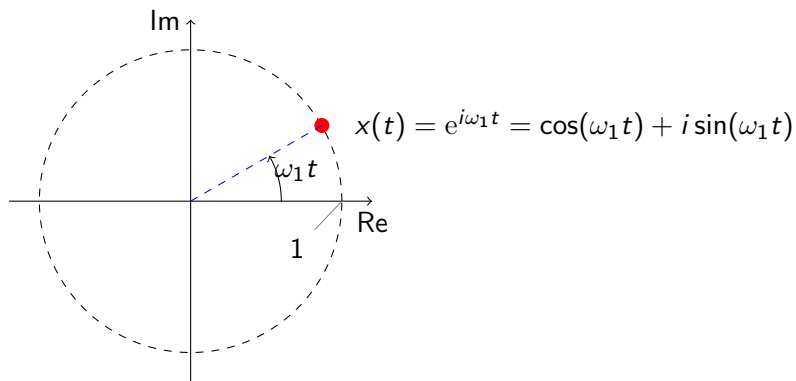

Figure 7.11 Frequency folding.

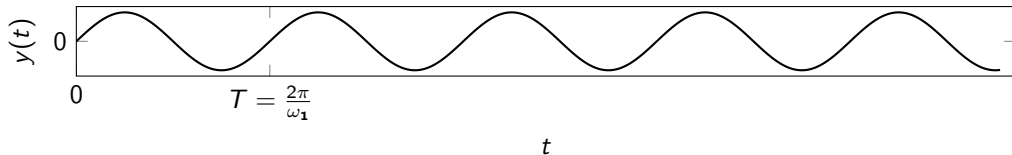# Fourier transform of a complex exponential

The function $x(t) = \mathrm{e}^{i\omega_1 t}$

$$x(t) = \mathrm{e}^{i\omega_1 t} = \cos(\omega_1 t) + i\sin(\omega_1 t)$$

has Fourier transform

$$X(i\omega) = \int_{-\infty}^{\infty} x(t)\mathrm{e}^{-i\omega t}dt = \int_{-\infty}^{\infty} \mathrm{e}^{i(\omega_1 - \omega)t}dt = \delta(\omega_1 - \omega)$$

Fourier transform of a sinusoid

A sinusoidal signal $y(t) = \sin(\omega_1 t)$ has all its power concentrated at one single frequency, $\omega = \omega_1 \ \mathrm{rad/s}$.
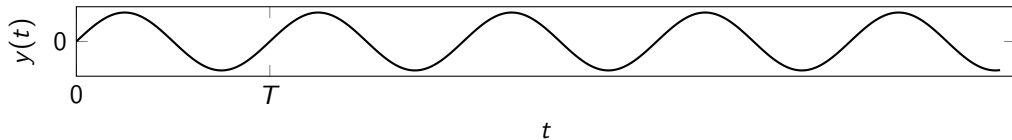

Since

$$y(t) = \sin(\omega_1 t) = \frac{1}{2i} \left(e^{i\omega_1 t} - e^{-i\omega_1 t} \right)$$

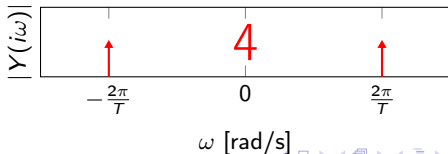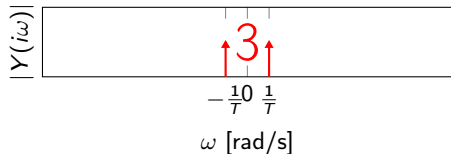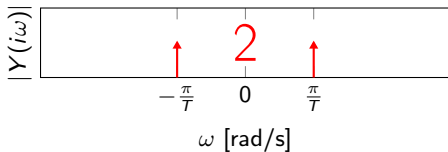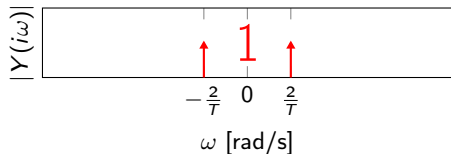the Fourier transform of a sinusoid becomes

$$Y(i\omega) = \frac{1}{2i} \left(\delta(\omega_1 - \omega) - \delta(\omega_1 + \omega) \right)$$

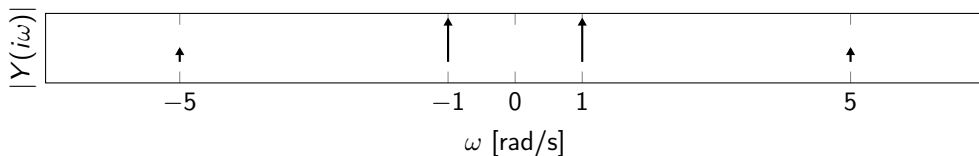# Exercise 1: Fourier transform of a sinusoid

Consider the signal below

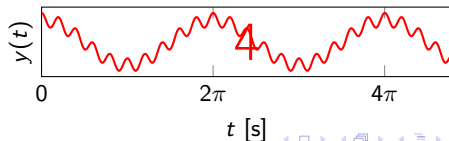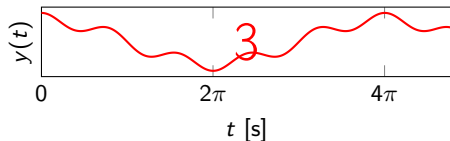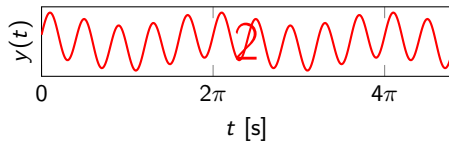Which of the below is the correct Fourier transform (magnitude plot shown)?

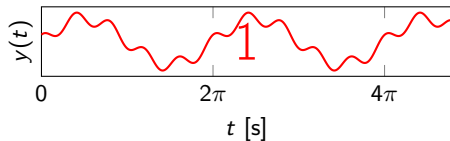# Exercise 2: Two sinusoids

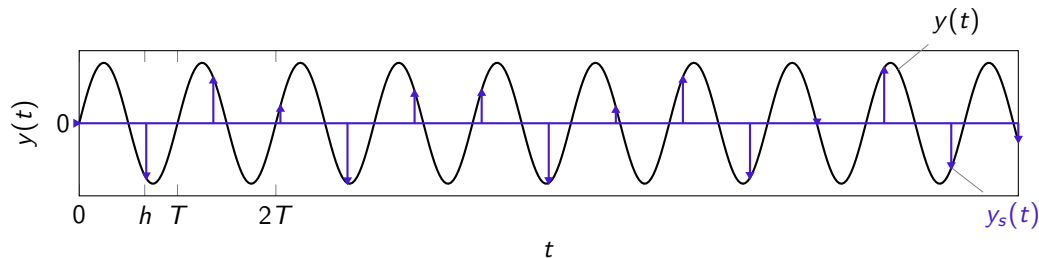Consider a signal with Fourier transform

Which of the below time series could this Fourier transform correspond to?

Exercise 3: Fourier transform of a sampled sinusoid

Consider the continuous and sampled signals with sampling period $h = \frac{2}{3}T$

What is the frequency of the sinusoid? What is the sampling frequency ω_s and the Nyquist frequency ω_N?

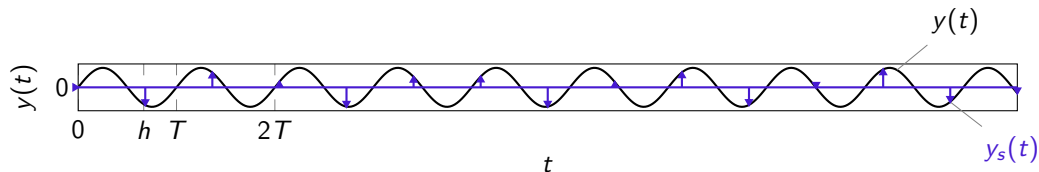# Exercise 3: Fourier transform of a sampled sinusoid

Consider the continuous and sampled signals with sampling period $h = \frac{2}{3}T$

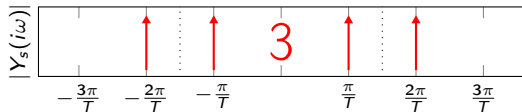Which of the below corresponds to the Fourier transform of the sampled signal?
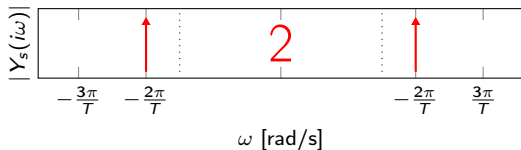
Alias frequency

To find the low frequency alias $\omega_a < \omega_N$ of a high freqency sinusoid ω_1, The expression

$$\omega_a = \left| \left((\omega_1 + \omega_N) \bmod \omega_s \right) - \omega_N \right|$$

can be used.

Aliasing example

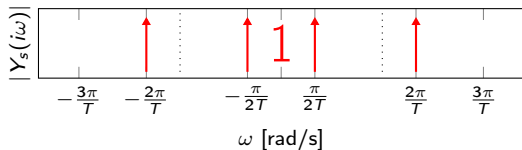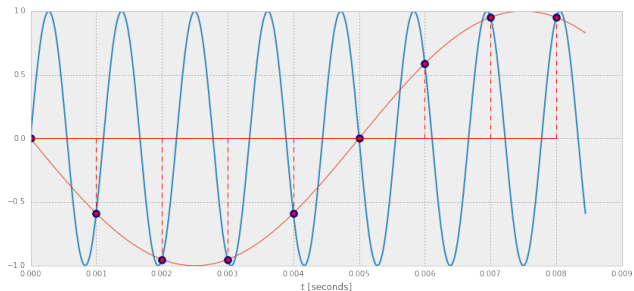If a continuous-time signal with frequency content (bandwidth) ω_B is sampled at too low sampling rate ($\omega_s < 2\omega_B$), then the energy at higher frequencies is folded onto lower frequencies.

A high-frequency sinusoid ($\omega_1 = 1800\pi$ rad/s) masquerading as a lower frequency sinusoid (200π rad/s) due to aliasing when sampled with $h = 10^{-3}$ s.

Draw the spectrum (lines) of the two sinusoids. Mark the Nyquist frequency and verify that the alias frequency is obtained by folding about the Nyquist frequency.

Noisy measurements
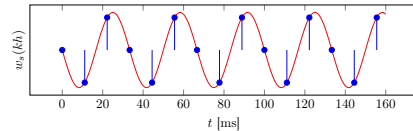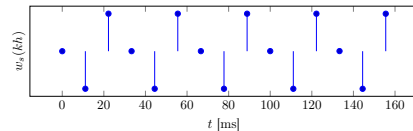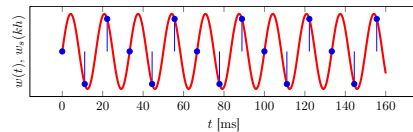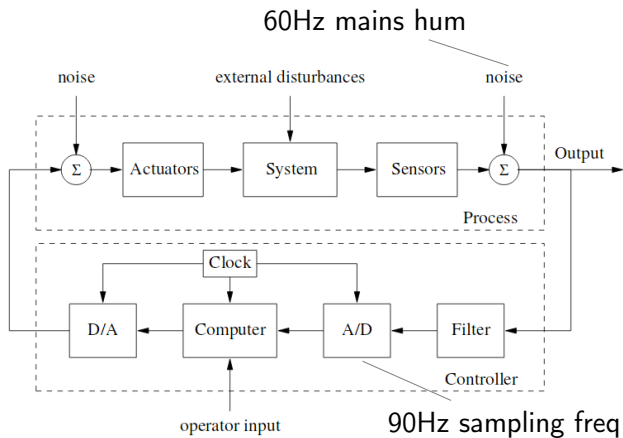
60Hz mains hum

90Hz sampling freq

noise

external disturbances

noise

Output

Σ — Actuators — System — Sensors — Σ

Process

Clock

D/A — Computer — A/D — Filter

Controller

operator input

Antialiasing filter

The Bessel filter is often used. From wikipedia:

In electronics and signal processing, a Bessel filter is a type of analog linear filter with a maximally flat group/phase delay (maximally linear phase response), which preserves the wave shape of filtered signals in the passband. Bessel filters are often used in audio systems.

Why use a Bessel filter as antialiasing filter?

Antialiasing filter

The Bessel filter is often used. From wikipedia:

In electronics and signal processing, a Bessel filter is a type of analog linear filter with a maximally flat group/phase delay (maximally linear phase response), which preserves the wave shape of filtered signals in the passband. Bessel filters are often used in audio crossover systems.

Why use as antialiasing filter?

- Preserves wave shapes $\Rightarrow$ very little distortion of signals in the passband

- Maximally linear phase response $\Rightarrow \arg H \approx -T\omega$, Can be modelled as a pure delay