

Nyquist and aliasing: how fast to sample sensor data

After reading, you can set a sampling rate from your analysis band, recognise a false low-frequency peak caused by aliasing, and see why slow PLC or SCADA logs fall short for spectra.

In short

- The sampling rate must be at least twice the highest frequency of interest, or components fold back into the spectrum.
- Practical FFT analysers commonly sample at about 2.56 times the highest analysed frequency, as a margin.
- PLC polling and historian compression add aliasing-like distortion and cannot replace high-rate vibration acquisition.

01 / The Nyquist limit: f_s and f_{max}

When you digitise a signal, the sampling rate f_s sets the time between samples. The Nyquist-Shannon sampling theorem states that if the highest frequency in a signal is f_{max} , the sampling rate must be at least twice f_{max} for the signal to be reconstructed without distortion. That threshold is the Nyquist limit. In a digital measurement, $f_s/2$ is the highest frequency that can be represented correctly.

When the condition is not met, the error is not only cosmetic. A false peak appears in the spectrum at a wrong frequency. High-frequency content you never intended to measure can fold down into the low-frequency range you are looking at. The theorem assumes a band-limited signal. Real sensor outputs always contain some high-frequency noise, so the limiting has to be done by an analogue filter before sampling.

02 / Aliasing: how a frequency folds back

In sampled data, a component above $f_s/2$ looks identical to a lower frequency. The apparent frequency is $|f - k \cdot f_s|$, where k is the integer nearest to f/f_s . This effect is called aliasing, or frequency folding.

Two examples make the rule concrete. A 60 Hz vibration sampled at 50 Hz gives $k = 1$ and an apparent frequency of $|60 - 50| = 10$ Hz. A pulsation at 0.9 Hz read once per second has $f_s = 1$ Hz, so the apparent frequency is $|0.9 - 1| = 0.1$ Hz. In both cases the data suggests a low-frequency event that is not in the signal.

03 / Filter before the converter

Aliasing cannot be undone after sampling. For that reason, the anti-aliasing filter must be analogue and placed before the converter. It attenuates components above the band of interest before they are sampled. A digital filter applied later in software cannot remove an alias that has already been created.

Downsampling follows the same logic. Before you keep only every n th sample, apply a low-pass filter that removes content above the new Nyquist limit. Otherwise, high-frequency content folds into the new, lower band. The cut-off frequency should sit above the band of interest but below $f_s/2$. If it is set too low, the analysis band itself is attenuated; if it is set too high, components near $f_s/2$ fold down without being removed.

04 / Hidden folding in PLC and historian data

A separate problem appears in the data path. A PLC or SCADA system reads values at fixed intervals. For example, with a one-second polling period, samples taken once per second can only represent changes below 0.5 Hz faithfully. Historian deadband or compression adds further distortion: it stores a new value only when the signal has changed enough, so the stored series is no longer a regular sampling of the process. The result can look like aliasing.

For that reason, do not build a high-frequency spectrum from slowly polled data. The effective sampling rate depends on every link in the acquisition chain, not only on the sensor.

05 / Choosing the sampling rate

A practical rule starts with the highest frequency of interest, $f_{\max}$, and sets f_s from it. For spectra, FFT analysers commonly use f_s of about $2.56 \times f_{\max}$. That leaves a margin above the theoretical Nyquist minimum. If you also want to see the shape of the waveform, a rate of 5 to 10 times $f_{\max}$ is more suitable. For example, with $f_{\max} = 400$ Hz, $f_s = 2.56 \times 400 = 1\,024$ Hz, so $f_s/2 = 512$ Hz. The transition band of the anti-aliasing filter must fit between 400 Hz and 512 Hz.

Choosing a band much wider than the fault frequencies you need also means more data and a higher f_s than necessary, so limit the band to the frequencies of interest.

After you choose, check the spectrum near $f_s/2$ and confirm it is flat or low there. Significant energy close to $f_s/2$ can mean real content near the limit, or an anti-aliasing filter that is too gentle. Until you understand the cause, do not draw conclusions from data near $f_s/2$.

06 / A first check on your own system

Before you install a new measurement, write down the existing settings: the channel, the read or sampling rate, whether there is an analogue filter, and whether the historian compresses the data. Then run a small test. Connect a reference signal of known frequency, such as a function generator, and check that its peak appears where it should. If a peak moves when you change f_s , it is most likely an alias.

This check matters when the data later feed machine learning or spectral diagnosis. Verify the measurement chain first, then choose the analysis method. Record the test results with the reference frequency, the chosen f_s , the peak seen and an explanation of any difference. The record can then serve as a comparison after later changes to the system.

Try it

Open a built-in synthetic signal in spectra, look at the spectrum near half the sampling rate, and note how you would tell a real peak from a folded one.

References and online guide

[Open the online guide](#)

[Nyquist-Shannon sampling theorem \(Wikipedia\)](#)

[Aliasing \(Wikipedia\)](#)

[Discrete Fourier Transforms \(scipy.fft\) \(SciPy\)](#)

[Accelerometer \(Wikipedia\)](#)

Proposed pilot steps do not claim field accuracy. Synthetic examples are not measurements from a production site.