Sample rate, number of bits, subsampling, and aliasing

Xavier Raick[1]

[1] Laboratory of Functional and Evolutionary Morphology, FOCUS, University of Liège, Liège, Belgium

xavier.raick@uliege.be

During the **digitalization process**, recording systems transform an infinite signal into a finite signal (Fig. 1). In this process, it is necessary to specify the **sampling rate**, i.e., the number of samples of audio recorded every second (in Hz). The **number of bits** give information about the quantification, i.e., the range of amplitude that can be recorded. The **subsampling** process reduces the quantity of information (number of points) in the signal, allowing for significant computational time savings. However, it is important to avoid excessive subsampling. According to the Nyquist-Shannon sampling theorem, the sample rate must be at least twice as high as the highest frequency of interest [1,2] in order to prevent **aliasing**, which can lead to misinterpretation of the data (Fig. 2). Since fish and benthic invertebrate sounds differ greatly in terms of temporal and frequency characteristics, subsampling at 4 kHz was performed only in chapters focusing on fish sounds.

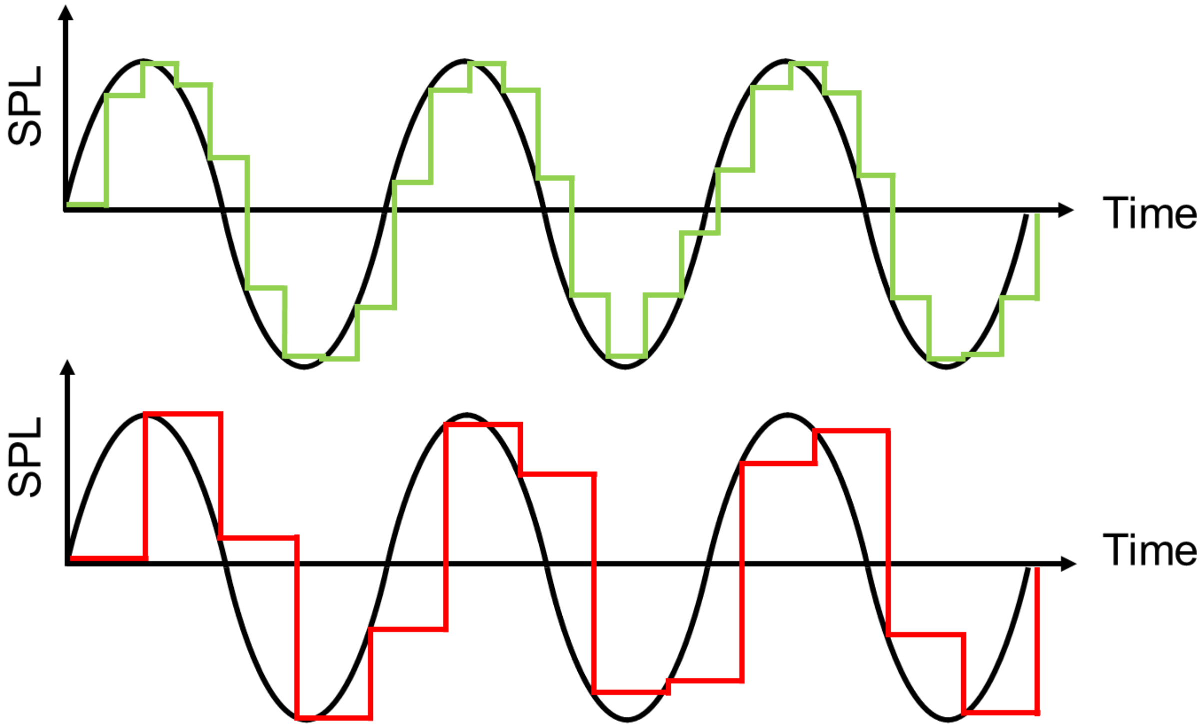

Fig. 1 Illustration of the digitalization process with two examples. The original signal is depicted in black, while the digitalized signals are shown in green and red.

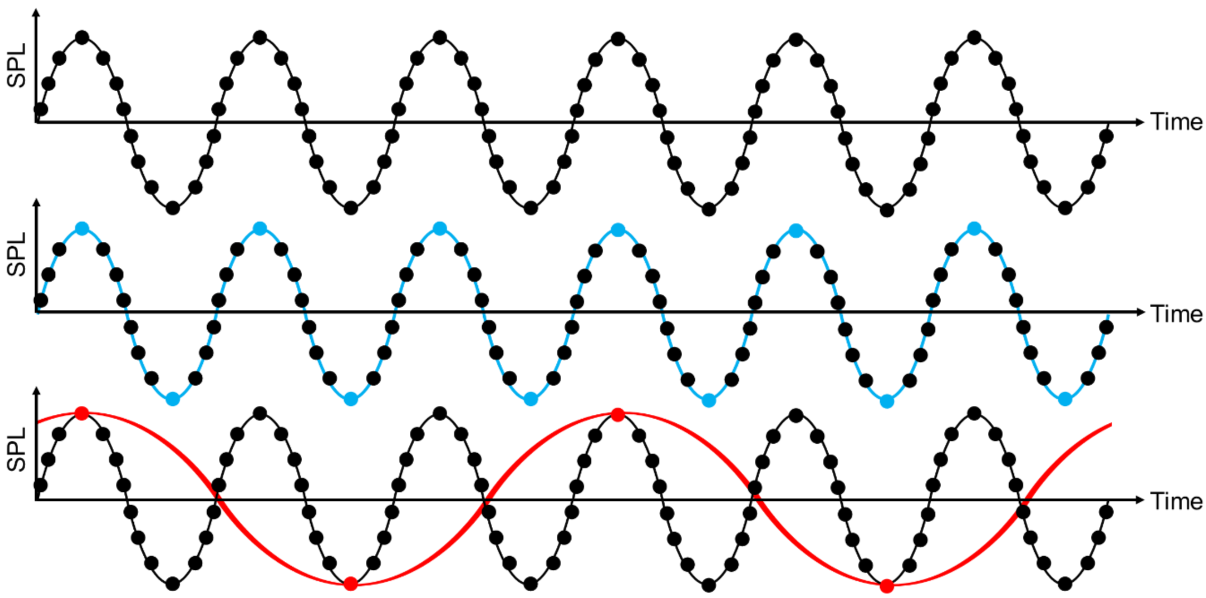

Fig. 2 Schematization of aliasing. The subsampled signal with a correct sample rate is represented in blue, while the non-correct sample rate, resulting in a new signal (in red) different from the original signal (in black).

REFERENCES

1. Nyquist, H. Certain Topics in Telegraph Transmission Theory. *Trans. Am. Inst. Electr. Eng.* **1928**, *47*, 617–644, doi:10.1109/T-AIEE.1928.5055024.
2. Shannon, C.E. A Mathematical Theory of Communication. *Bell Syst. Tech. J.* **1948**, *27*, 379–423, doi:10.1002/j.1538-7305.1948.tb01338.x.