SAMPLING

A signal that is bandlimited to a frequency of B Hz can (theoretically) be reproduced exactly from samples taken at a rate of $2B$ Hz or greater.

Sampling Theorem: $\overline{f}(t) = f(t)\delta_T(t) = \sum_n f(nT)\delta(t-nT)$	$\overline{f}(t)$ is the sampled signal $f(t)$ is the signal $\delta_T(t)$ is the impulse train (regularly spaced pulses of size 1) T is the sampling interval in seconds n is an integer denoting a particular impulse $F_s = 1/T$ is the sampling rate in Hz.

from Dr. Sandberg: $\frac{1}{T} f(t) = \sum_{n=-\infty}^{\infty} f(nT) \frac{\sin \overline{w}(t-nT)}{\pi(t-nT)}$, $\omega_0 < 2w$, small ε	$f(t)$ is the signal $f(nT)$ is the sampled signal T is the sampling interval in seconds n is an integer denoting a particular impulse t is time in seconds $2w$ is called the **Nyquist frequency** ε is the width of the impulse ω_0 is the radian frequency of the signal

Aliasing is a type of distortion found in signal sampling which may be viewed in a graph of the signal in the time domain as overlapping tails. It may be eliminated by bandlimiting the signal before sampling.

The **Nyquist interval** is the minimum sampling interval required to recover the original signal from its samples, equal to the reciprocal of twice the bandwidth.

The **Nyquist rate** is the minimum sampling rate required to recover the original signal from its samples, equal to 2× the bandwidth.

The **sinc function**, also known as the filtering or interpolating function is defined as

$$\mathrm{sinc}(x) = \frac{\sin x}{x}.$$