# EE120 - Fall'19 - Lecture 12<sup>1</sup> Murat Arcak 17 October 2019

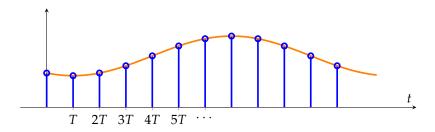
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## Sampling

Suppose we have samples of a continuous-time signal x taken every T units of time, that is, with the sampling frequency:

$$\omega_{\rm s} = \frac{2\pi}{T}.\tag{1}$$

Can we reconstruct x(t) from the samples  $\{x(nT)\}_{n\in\mathbb{Z}}$  with an appropriate interpolation?



The Sampling Theorem by Shannon and Nyquist shows that the answer is yes if *x* is bandlimited, i.e.,

$$X(\omega) = 0 \quad |\omega| > \omega_M \tag{2}$$

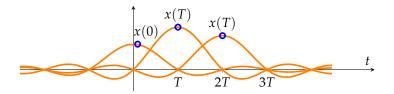
for some finite frequency  $\omega_M$ , and the sampling frequency satisfies

$$\omega_s > 2\omega_M.$$
 (3)

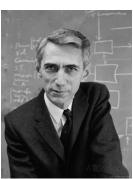
The proof of this theorem, stated more precisely below, uses scaled sinc functions for interpolation:

$$x_r(t) = \sum_{n=-\infty}^{\infty} x(nT) \operatorname{sinc}\left(\frac{t-nT}{T}\right).$$
 (4)

Each sinc function in this sum is centered at a sample point. Verify that (4), when evaluated at a sample point, returns the value of x at that point. This is illustrated in the figure below.



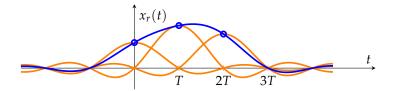
Chapter 7 in Oppenheim & Willsky



Claude Shannon (1916-2001)



Harry Nyquist (1889-1976)



Sampling Theorem: If *x* is bandlimited as in (2) and we select *T* such that the sampling frequency (1) satisfies (3), then  $x_r(t) = x(t)$ .

Proof: We view  $x_r$  in (4) as the output of a LTI system with the following impulse response and input:

$$h_r(t) = \operatorname{sinc}\left(\frac{t}{T}\right) \tag{5}$$

$$x_p(t) := \sum_{n = -\infty}^{\infty} x(nT)\delta(t - nT).$$
 (6)

Show that the convolution of these two indeed gives  $x_r$  in (4). Thus,

$$X_r(\omega) = H_r(\omega)X_p(\omega) \tag{7}$$

where  $H_r(\omega)$  is obtained from the Fourier Transform of (5):

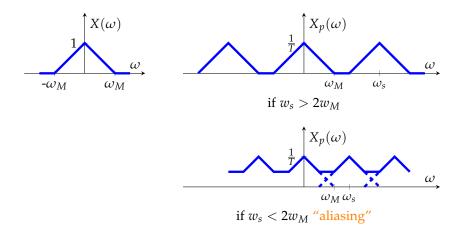
$$H_r(\omega) = \begin{cases} T & |\omega| < \frac{\pi}{T} \\ 0 & |\omega| > \frac{\pi}{T} = \frac{\omega_s}{2}. \end{cases}$$
 (8)

$$X_{p}(\omega) \longrightarrow \underbrace{T \xrightarrow{\omega_{s}}_{\frac{\omega_{s}}{2}}}_{T_{r}(\omega)} X_{r}(\omega)$$

Moreover, as shown in the next section,

$$X_p(\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} X(\omega - k\omega_s)$$
 (9)

and (3) guarantees that the shifted copies in this sum do not overlap:



Thus,  $X_p(\omega) = \frac{1}{T}X(\omega)$  when  $|\omega| \leq \frac{\omega_s}{2}$ , and (7) and (8) give

$$X_r(\omega) = X(\omega).$$

This means that the interpolated signal matches the continuous-time signal from which the samples were obtained.  $\Box$ 

There are two critical steps in the proof of the Sampling Theorem above. The first is to express the sinc interpolation in (4) as low pass filtering of  $x_p$  defined in (6). The second is to use (9), which states that the Fourier Transform of  $X_p(\omega)$  consists of shifted copies of  $X(\omega)$ . When  $\omega_M < \omega_s/2$ , these copies do not overlap and the lowpass filter recovers x. When the copies overlap the low-pass filter will output a signal that differs from x, a phenomenon called "aliasing."

### Impulse Train Sampling

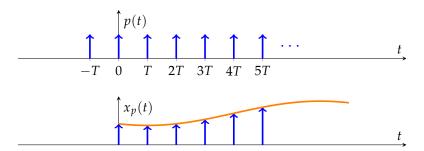
To see where (9) comes from, we note that  $x_p$  in (6) can be written as

$$x_p(t) = x(t) \cdot p(t) \tag{10}$$

where

$$p(t) := \sum_{n = -\infty}^{\infty} \delta(t - nT) \tag{11}$$

is called an "impulse train" and  $x_p$  is called the impulse train sampled signal. Both signals are illustrated below.



Recall that the Fourier series coefficients of the impulse train are  $a_k=\frac{1}{T}$  for all k. Thus,

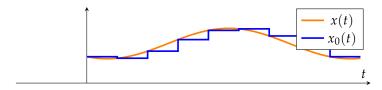
$$P(j\omega) = \sum_{k=-\infty}^{\infty} 2\pi a_k \delta(\omega - k\omega_s) = \frac{2\pi}{T} \sum_{k=-\infty}^{\infty} \delta(\omega - k\omega_s)$$
 (12)

$$X_p(j\omega) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(j\theta) P(j(\omega - \theta)) d\theta = \frac{1}{T} \sum_{k=-\infty}^{\infty} X(j(\omega - k\omega_s)),$$

which proves (9).

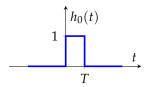
Since sinc interpolation corresponds to ideal low pass filtering, we next explore simpler forms of interpolation and compare the resulting frequency response to the ideal low-pass filter.

We first consider a zero-order hold which creates a staircase continuoustime signal with constant values in between the samples as shown below.



The resulting signal  $x_0$  can be viewed as the output of a LTI system with impulse response  $h_0$  shown below when the input is  $x_p$ :

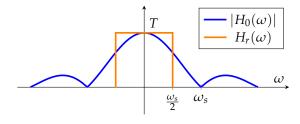
$$x_0(t) = (x_p * h_0)(t) \tag{13}$$



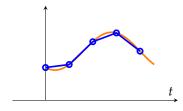
Thus,

$$H_0(\omega) = e^{-j\omega T/2} T \operatorname{sinc}\left(\frac{T}{2\pi}\omega\right)$$
 (14)

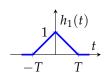
$$X_0(\omega) = H_0(\omega)X_p(\omega). \tag{15}$$

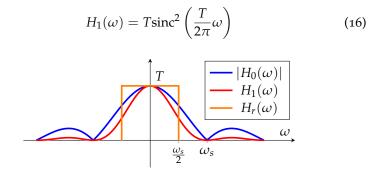


*Linear Interpolation (First-order Hold)* 



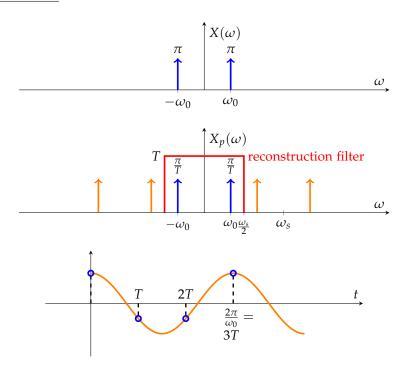
$$x_1(t) = (x_p * h_1)(t)$$



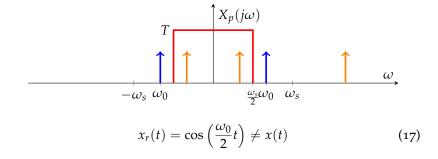


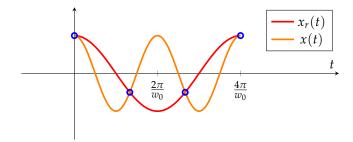
# Simple Examples of Aliasing

Example 1:  $x(t) = \cos(\omega_0 t)$ ,  $\omega_s = 3\omega_0 > 2\omega_0$  (no aliasing)



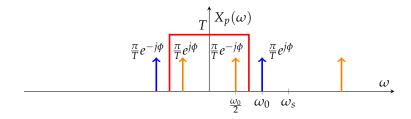
Example 2:  $x(t) = \cos(\omega_0 t)$ ,  $\omega_s = \frac{3\omega_0}{2} < 2\omega_0$  (aliasing)





#### Example 3: (phase reversal)

$$x(t) = \cos(\omega_0 t + \phi) = \frac{1}{2} e^{j\phi} e^{j\omega_0 t} + \frac{1}{2} e^{-j\phi} e^{-j\omega_0 t}, \quad \omega_s = \frac{3\omega_0}{2}$$
 (18)



$$X_{r}(\omega) = \pi e^{j\phi} \delta\left(\omega + \frac{\omega_{0}}{2}\right) + \pi e^{-j\phi} \delta\left(\omega - \frac{\omega_{0}}{2}\right)$$

$$\downarrow \qquad \qquad \downarrow \qquad \downarrow$$

$$x_r(t) = \frac{1}{2} \left( e^{j\left(\frac{\omega_0}{2}t - \phi\right)} + e^{-j\left(\frac{\omega_0}{2}t - \phi\right)} \right) \tag{20}$$

$$= \cos\left(\frac{\omega_0}{2}t - \phi\right) \to \text{ phase reversal} \tag{21}$$

$$=\cos\left(-\frac{\omega_0}{2}t + \phi\right) \tag{22}$$

Wagon wheel effect in movies: Wheel appears to rotate more slowly and in the opposite direction when actual speed exceeds half of the sampling rate (18-24 frames/second).

Example 4: Suppose we sample the signal

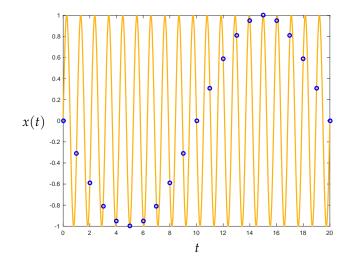
$$x(t) = \sin(1.9\pi t) \tag{23}$$

with T = 1 as shown in the figure below. It follows from Fourier analysis similar to the examples above that sinc interpolation gives<sup>2</sup>

<sup>2</sup> Show (24).

$$x_r(t) = \cos(0.1\pi t + \pi/2) = -\sin(0.1\pi t),$$
 (24)

which is evident from the samples in the figure. The negative sign of  $-\sin(0.1\pi x)$  is a result of the phase reversal discussed above.



## Example 5: (critical frequency)

