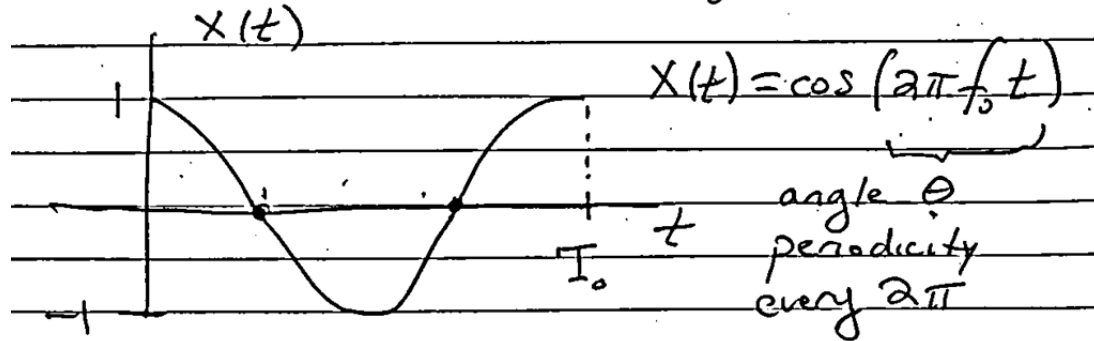


[11:00-11:35] Continuation of Lecture 1: Sampling and sound in MATLAB

Lecture Slide 1-13

EE313

August 28, 2025



$$t=0: x(0) = \cos(2\pi f_0(0)) = 1$$

$$2\pi f_0 t = 2\pi \rightarrow t = \frac{1}{f_0} = T_0$$

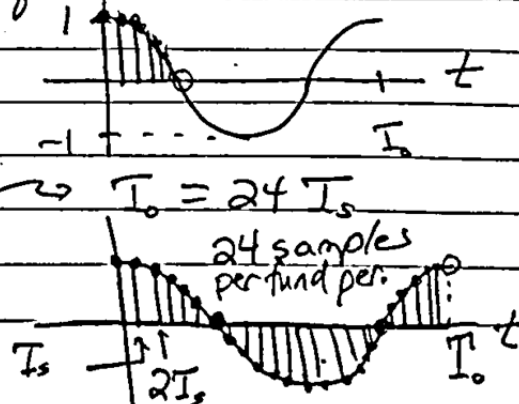
$$\text{Fundamental period is } T_0 = \frac{1}{f_0}$$

Next step is to sample $x(t)$
at uniformly spaced points in time
to create a vector of amplitude values

$$\text{Choose a sampling rate } f_s = \frac{1}{T_s} \left[\frac{1}{\text{Hz}} \right] \rightarrow T_s \left[\text{s} \right]$$

Guidance on choosing the sampling
rate is from the Nyquist Theorem
 $f_s > 2 f_0$

e.g. Pick $f_s = 24 f_0$
 $\frac{1}{T_s} = 24 \frac{1}{T_0} \rightarrow T_0 = 24 T_s$



To reconstruct a frequency f , a signal must be sampled at a rate strictly greater than $2f$.

It's common to sample much higher than $2f$. Audio is often sampled at 48,000 Hz or higher, even though the maximum audible frequency is between 15,000 Hz and 17,000 Hz for most adults.

Standard		ECE 313 Evans	
Audio Sampling Rates		Slide 1-15	
speech sampling →	8000 Hz	11,025	:
	16000 Hz	22,050	:
	32000 Hz	44,100 Hz (CD)	48000 Hz (DAT)
		88,200 Hz	96000 Hz
	31,250 Hz	196,400 Hz	192000 Hz
MIDI	24000 Hz	:	:
↳ Musical Instrument Digital Interface			
CD Compact Disc			
DAT Digital Audio Tape			

MATLAB's `sound(x)` command will play an audio signal represented by an array x using the default sample rate of 8192 Hz. A different sampling rate f_s can be specified using `sound(x, f_s)`, in which case MATLAB will automatically [resample](#) the sound to a rate that is compatible with the computer's sound system.

The `soundsc` command will scale the range of the signal to $[-1, 1]$ prior to playing it. This increases the volume of small amplitude signals and prevents clipping when playing a signal whose amplitude exceeds the range $[-1, 1]$.

[11:35 - 12:05] Systems

Systems operate on one or more input signals to produce one or more output signals.

Examples:

Averaging a continuous signal with a one-second delayed version of itself:

$$y(t) = \frac{1}{2}x(t) + \frac{1}{2}x(t - 1)$$

Discrete time low-pass averaging filter:

$$y[n] = \frac{1}{2}x[n] + \frac{1}{2}x[n - 1]$$

Squaring block:

$$y(t) = x(t)^2$$

The squaring block is nonlinear and changes the frequency or frequencies of the input signal $x(t)$. The output $y(t)$ of the squaring block contains:

- Double the frequency (or frequencies) of $x(t)$
- A DC offset (zero Hertz)

[11:55-12:05] Lecture 2: Periodic signals

A signal $x(t)$ is periodic with period T if $x(t - nT) = x(t)$ for any integer n .

A periodic signal can have multiple periods. The smallest positive period is the fundamental period.

Time shift and phase shift

A common system involves shifting a signal $x(t)$ in time by T seconds. If T is positive, this system is also called a delay.

$$D_T\{x(t)\} = x(t - T)$$

For sinusoids, a time shift causes a phase shift:

$$D_T\{A \cos(2\pi f_0 t)\} = A \cos(2\pi f_0 (t - T)) = A \cos(2\pi f_0 t - 2\pi f_0 T)$$

Phase shifts are commonly used in wireless communications, sonar, and radar systems to separate signals spatially.