

# **REVIEW 5 SAMPLING**

MODIFIED TLH

DSP First, 2/e

## **Sampling & Aliasing**

**CHAPTER 4 PRESENTATION 2**

# System IMPLEMENTATION

- **ANALOG/ELECTRONIC:**

- Circuits: resistors, capacitors, op-amps



- **DIGITAL/MICROPROCESSOR**

- Convert  $x(t)$  to **numbers** stored in memory



# SAMPLING THEOREM THE BIG DEAL!!

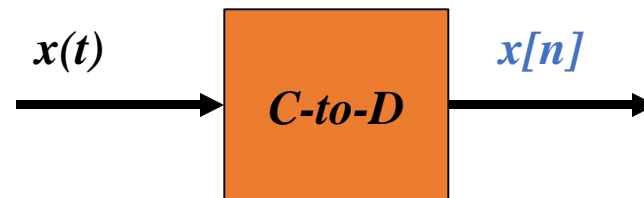
- HOW OFTEN DO WE NEED TO SAMPLE?
  - DEPENDS on FREQUENCY of SINUSOID
  - ANSWERED by SHANNON/NYQUIST Theorem
  - ALSO DEPENDS on “**RECONSTRUCTION**”

## *Shannon Sampling Theorem*

A continuous-time signal  $x(t)$  with frequencies no higher than  $f_{\max}$  can be reconstructed exactly from its samples  $x[n] = x(nT_s)$ , if the samples are taken at a rate  $f_s = 1/T_s$  that is greater than  $2f_{\max}$ .

# SAMPLING $x(t)$

- SAMPLING PROCESS
  - Convert  $x(t)$  to **numbers**  $x[n]$
  - “ $n$ ” is an integer index;  $x[n]$  is a sequence of values
  - Think of “ $n$ ” as the storage address in memory
- UNIFORM SAMPLING at  $t = nT_s$ 
  - IDEAL:  $x[n] = x(nT_s)$



# SAMPLING RATE, $f_s$

- SAMPLING RATE ( $f_s$ )

- $f_s = 1/T_s$

- NUMBER of SAMPLES PER SECOND

**SOMETIMES GIVEN IN Hz**

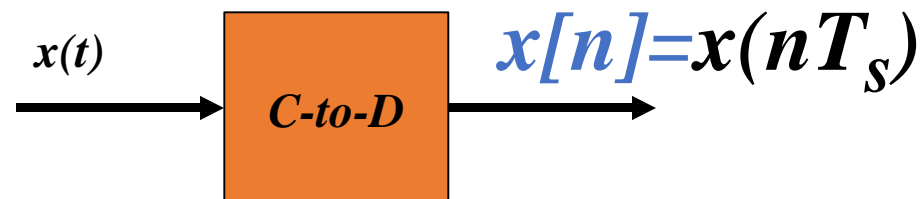
- $T_s = 125$  microsec **7**  $f_s = 8000$  samples/sec

- UNITS of  $f_s$  ARE HERTZ: 8000 Hz

**Thus -  $F_{max} = 4000$  Hz**

- UNIFORM SAMPLING at  $t = nT_s = n/f_s$

- IDEAL:  $x[n] = x(nT_s) = x(n/f_s)$



# STORING DIGITAL SOUND

- $x[n]$  is a SAMPLED SIGNAL
    - A list of numbers stored in memory
  - EXAMPLE: audio CD
  - CD rate is 44,100 samples per second
    - 16-bit samples
    - Stereo uses 2 channels
  - Number of bytes for 1 minute is
    - $2 \times (16/8) \times 60 \times 44100 = 10.584$  Mbytes
- THUS – Frequency range of 22,050 Hz is beyond (most) humans hearing range.**

# DIGITAL FREQUENCY $\hat{\omega}$

$$\hat{\omega} = \omega T_s = \frac{2\pi f}{f_s}$$

- $\hat{\omega}$  VARIES from **0** to  **$2\pi$** , as  $f$  varies from 0 to the sampling frequency
- UNITS are radians, **not** rad/sec
  - DIGITAL FREQUENCY is NORMALIZED

# DISCRETE-TIME SINUSOID

- Change  $x(t)$  into  $x[n]$  **DERIVATION**

$$x(t) = A \cos(\omega t + \varphi)$$

$$x[n] = x(nT_s) = A \cos(\omega nT_s + \varphi)$$

$$x[n] = A \cos((\omega T_s)n + \varphi)$$

$$x[n] = A \cos(\hat{\omega}n + \varphi)$$

$$\hat{\omega} = \omega T_s = \frac{\omega}{f_s}$$

**DEFINE DIGITAL FREQUENCY**



- SAMPLING can cause ALIASING
  - Sampling Theorem
  - Sampling Rate > 2(Highest Frequency)
- Spectrum for digital signals,  $x[n]$ 
  - Normalized Frequency

$$\hat{\omega} = \omega T_s = \frac{2\pi f}{f_s} + 2\pi \ell$$



Example

$$\cos(200\pi t + \theta) \quad f = 100 \text{ Hz}$$

a) Need  $f_s = 5 \times 200$  samples/sec

b) frequencies

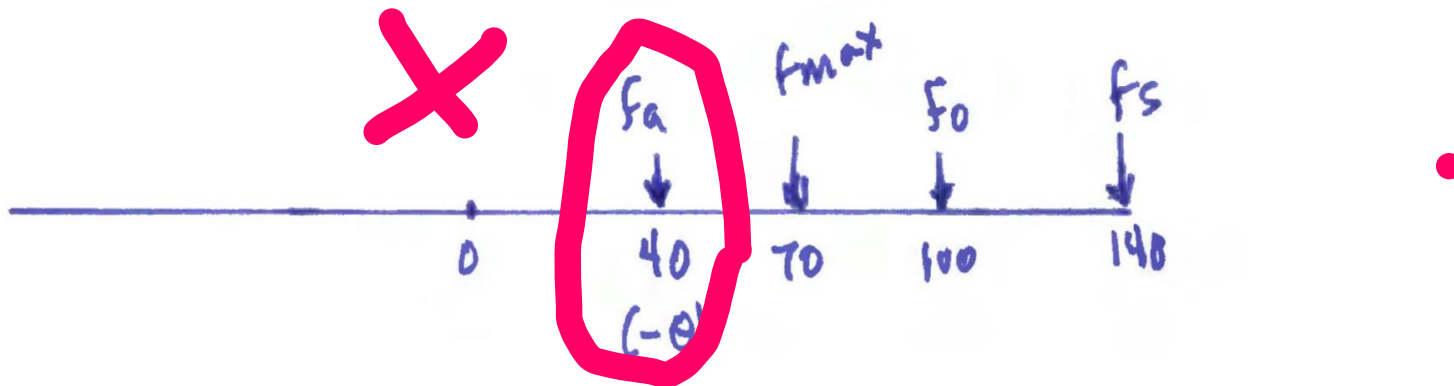
$$f_s = 240 \text{ Hz} \quad f = 100 \text{ Hz} \quad \text{OK}$$

$f_s = 140 \text{ Hz}$  - Aliased

$$f_a = 100 - 140 = -40 \text{ Hz}$$

$$x_a(t) = \cos(-2\pi(40)t + \theta) = \cos(80\pi t - \theta)$$

Phase Reversal



# ALIASING CONCLUSIONS

- Adding an **INTEGER multiple** of  $f_s$  or  $-f_s$  to the frequency of a continuous sinusoid  $x_c(t)$  gives **exactly the same values** for the sampled signal  $x[n] = x_c(n/f_s)$
- **GIVEN  $x[n]$ , we CAN'T KNOW** whether it came from a sinusoid at  $f_0$  or  $(f_0 + f_s)$  or  $(f_0 + 2f_s)$  ...
- **This is called ALIASING**

# Video Aliasing

**Why car wheels rotate  
backwards in movies 4:25**

<https://www.youtube.com/watch?v=SFbINinFsxk&feature=youtu.be>