

# Stanford University EE 102B Spring-2013

## Lecture 04 Sampling, Aliasing, and Reconstruction April 8, 2013

## ASSIGNMENTS

- Reading for this Lecture:
  - SPF: Chapter 4
  - S&S: The sampling discussion here will be useful when we revisit sampling in a couple of weeks.
- HW#1 and Lab #1 are posted
  - Both due on Weds. April 10 at 5pm in Packard 263.

4/8/13

© 2003, JH McClellan & RW Schafer

2

## Office Hours for Course Staff – Come see us.

- Ron Schafer: Mon. 2:00-3:00 pm and Weds. 4:00-5:00 pm in Packard 211
- Dookun Park: Tues. 1:30 ~ 3:30 pm in Packard 107
- Ruo Yu Gu (Roy): Tues. 4:00 ~ 6:00 pm in Packard 106
- <https://class2go.stanford.edu/EE102B/Spring2013/pages/staff>

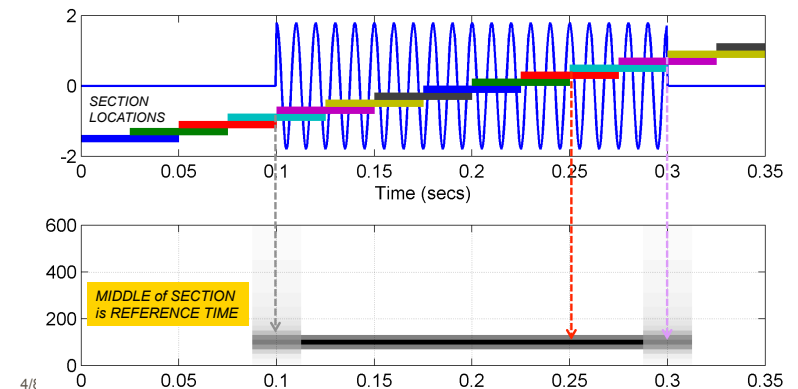
4/8/13

© 2003, JH McClellan & RW Schafer

3

## Overlapping Sections in Spectrograms

- 50% overlap. More is required for smooth image.
- Consider edge effects when analyzing a short sinusoid



# LECTURE OBJECTIVES

- SAMPLING causes ALIASING
  - Sampling Theorem**
  - Sampling Rate > 2(Highest Frequency)
- Spectrum for digital signals,  $x[n]$ 
  - Normalized Frequency in SPF  $\hat{\omega} = \omega T_s$
  - In Oppenheim and Willsky,  $\Omega = \omega T_s$
- Reconstruction from samples

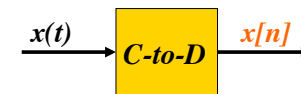
4/8/13

© 2003, JH McClellan & RW Schaffer

5

# SAMPLING $x(t)$

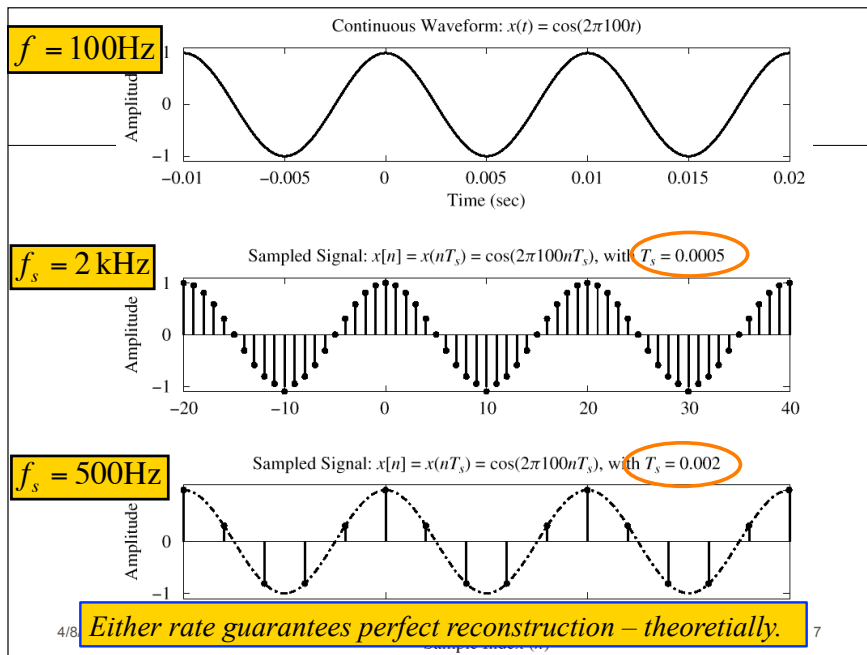
- SAMPLING PROCESS
  - Convert  $x(t)$  to **numbers**  $x[n]$
  - "n" is an integer;  $x[n]$  is a sequence of values
  - Think of "n" as the storage address in memory
- UNIFORM SAMPLING at  $t = nT_s$ 
  - IDEAL:  $x[n] = x(nT_s)$



4/8/13

© 2003, JH McClellan & RW Schaffer

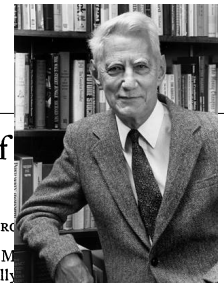
6



# Shannon, 1949

## Communication in the Presence of Noise

CLAUDE E. SHANNON†, MEMBER, IRE



**Summary**—A method is developed for representing any communication system geometrically. Messages and the corresponding signals are points in two "function spaces," and the modulation process is a mapping of one space into the other. Using this representation, a number of results in communication theory are deduced concerning expansion and compression of bandwidth and the threshold effect. Formulas are found for the maximum rate of transmission by various methods, and the conditions under which the number of bits per second is calculated.

**I. INTRODUCTION**

A GENERAL COMMUNICATION SYSTEM is shown schematically in Fig. 1, consisting of five elements.

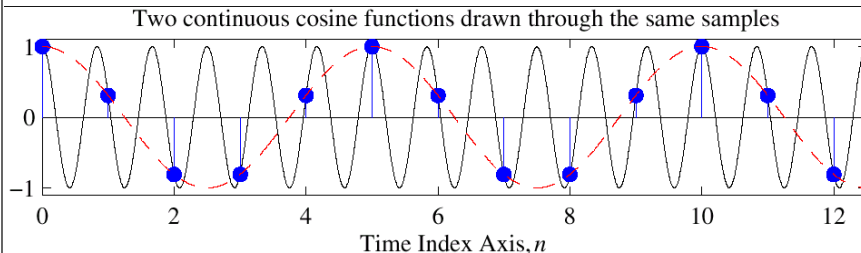
1. An information source. The source selects one message from a set of possible messages, as shown in Fig. 1. The message is then transmitted through a channel to the receiver, as shown in Fig. 1. The message is then reconstructed at the receiver, as shown in Fig. 1.

**THEOREM 1:** If a function  $f(t)$  contains no frequencies higher than  $W$  cps, it is completely determined by giving its ordinates at a series of points spaced  $1/2W$  seconds apart.

Proceedings of the I.R.E., January, 1949

## Reconstruction? Which One?

Given the samples, draw a sinusoid through the values



$$x[n] = \cos(0.4\pi n)$$

When  $n$  is an integer  
 $\cos(0.4\pi n) = \cos(2.4\pi n)$

Reconstruction method picks lowest frequency sinusoid

4/8/13

9

## DISCRETE-TIME SINUSOID

- Sample  $x(t)$  to obtain the sequence  $x[n]$

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

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

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

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

$$\hat{\omega} = \omega T_s = \frac{\omega}{f_s} \quad \text{DEFINE DIGITAL FREQUENCY}$$

4/8/13

© 2003, JH McClellan & RW Schaefer

10

## 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  $f_s$
- UNITS are radians, not rad/sec
  - DIGITAL FREQUENCY is NORMALIZED

4/8/13

© 2003, JH McClellan & RW Schaefer

11

## SPECTRUM (DIGITAL)

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

$$f_s = 1 \text{ kHz}$$

$$\frac{1}{2} X^*$$

$$-0.2\pi$$

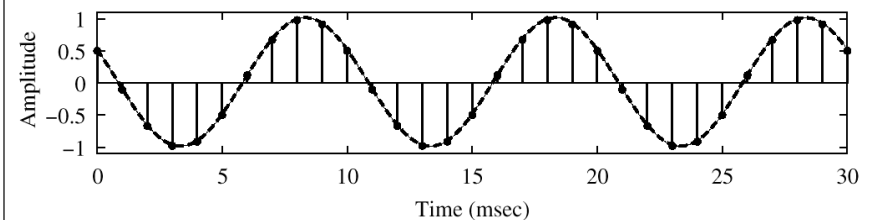
$$\frac{1}{2} X = \frac{1}{2} A e^{j\varphi}$$

$$2\pi(0.1) = 0.2\pi$$

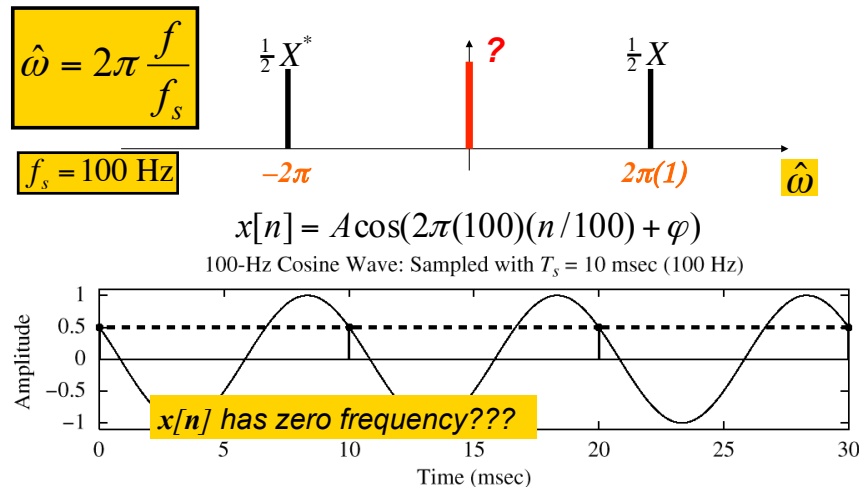
$$\hat{\omega}$$

$$x[n] = A \cos(2\pi(100)(n/1000) + \varphi)$$

100-Hz Cosine Wave: Sampled with  $T_s = 1$  msec (1000 Hz)



## SPECTRUM (DIGITAL) ???



## Stroboscope Effect

- Check out this video on U-tube
- <http://www.youtube.com/watch?v=YpMtanpsVeE>

4/8/13

EE-2025 Fall-2004 JMc

14

## The REST of the STORY

- Spectrum of  $x[n]$  has more than one line for each complex exponential
  - Called **ALIASING**
  - MANY SPECTRAL LINES**
- SPECTRUM is PERIODIC with period =  $2\pi$ 
  - Because

$$A \cos(\hat{\omega}n + \varphi) = A \cos((\hat{\omega} + 2\pi\ell)n + \varphi)$$

4/8/13

© 2003, JH McClellan & RW Schaefer

15

## ALIASING DERIVATION

- Other Frequencies give the same  $\hat{\omega}$

If  $x(t) = A \cos(2\pi(\underline{f} + \ell f_s)t + \varphi)$

$t \leftarrow \frac{n}{f_s}$

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

then :  $\hat{\omega} = \frac{2\pi(f + \ell f_s)}{f_s} = \frac{2\pi f}{f_s} + \frac{2\pi \ell f_s}{f_s}$

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

4/8/13

16

## ALIASING CONCLUSIONS

- Adding an **INTEGER multiple** of  $f_s$  or  $-f_s$  to the frequency of a sinusoid  $x_c(t)$  gives **exactly the same values** for  $x[n] = x_c(n/f_s)$
- 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**

## NORMALIZED FREQUENCY

- DIGITAL FREQUENCY

*Normalized Radian Frequency*

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

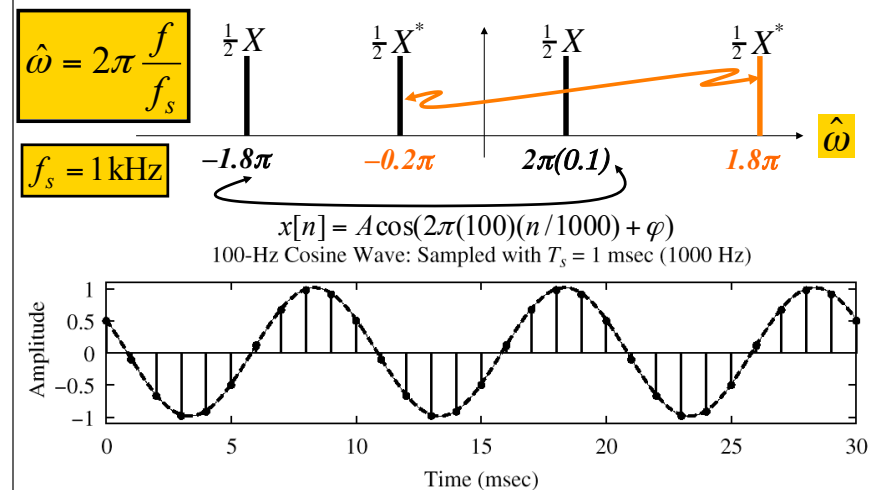
*Normalized Cyclic Frequency*

$$\hat{f} = \hat{\omega}/(2\pi) = f T_s = f/f_s$$

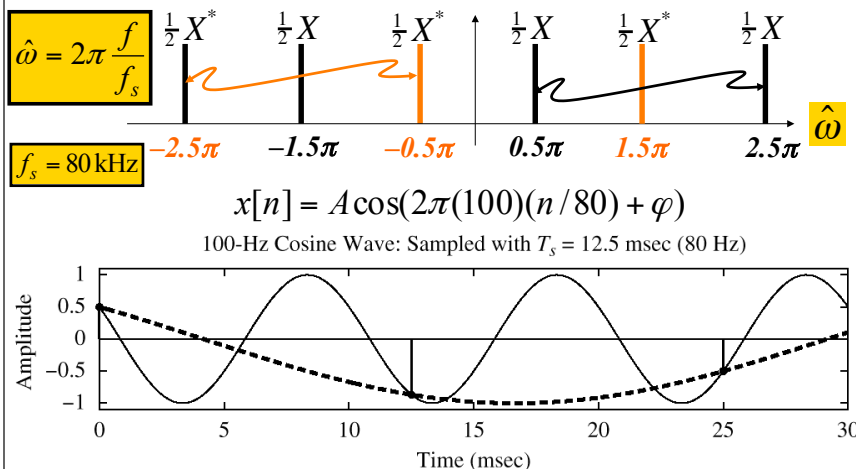
## SPECTRUM for $x[n]$

- PLOT versus NORMALIZED FREQUENCY
- INCLUDE **ALL** SPECTRUM LINES
  - ALIASES
    - ADD MULTIPLES of  $2\pi$
    - SUBTRACT MULTIPLES of  $2\pi$
  - FOLDED ALIASES
    - (to be discussed later)
    - ALIASES of NEGATIVE FREQS

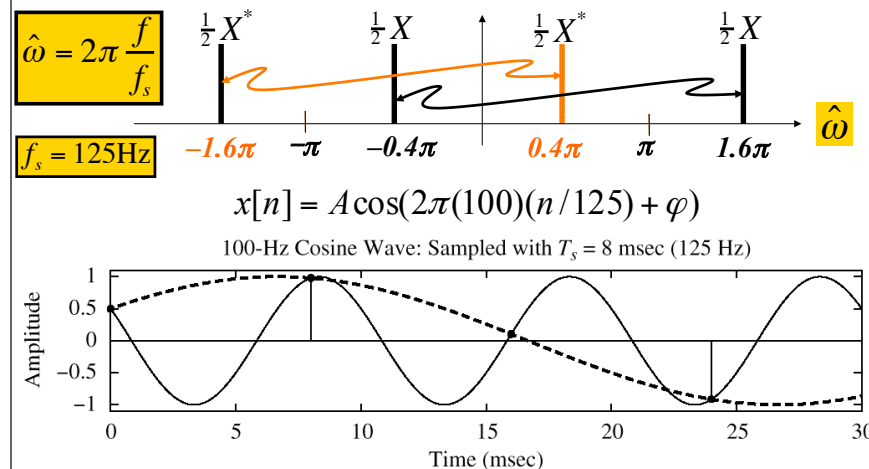
## SPECTRUM (MORE LINES)



## UNDERSAMPLING (ALIASING DISTORTION)



## SPECTRUM (FOLDING CASE)



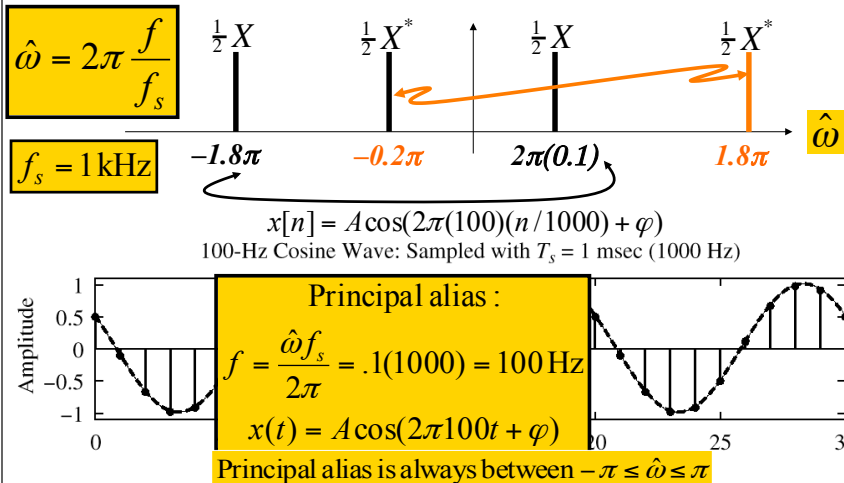
## Terminology: NYQUIST RATE

- “**Nyquist Rate**” Sampling
  - $f_s > \text{TWICE}$  the HIGHEST Frequency in  $x(t)$
  - “Sampling above the Nyquist rate”
- **BANDLIMITED SIGNALS**
  - DEF: HIGHEST FREQUENCY COMPONENT in SPECTRUM of  $x(t)$  is finite
  - PERIODIC SQUARE WAVE is **NOT** BANDLIMITED

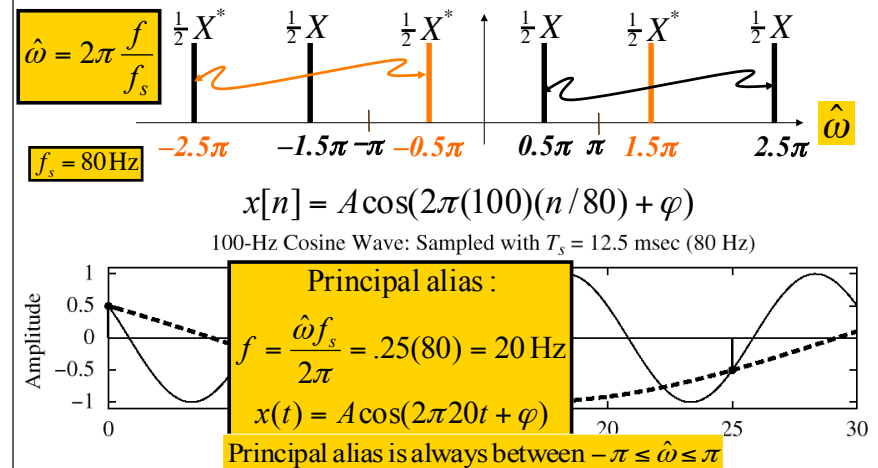
## EXAMPLE: SPECTRUM

- $x[n] = A \cos(0.2\pi n + \phi)$
- FREQS @  $0.2\pi$  and  $-0.2\pi$
- ALIASES:
  - $\{2.2\pi, 4.2\pi, 6.2\pi, \dots\}$  &  $\{-1.8\pi, -3.8\pi, \dots\}$
  - EX:  $x[n] = A \cos(4.2\pi n + \phi)$
- ALIASES of **NEGATIVE** FREQ:
  - $\{1.8\pi, 3.8\pi, 5.8\pi, \dots\}$  &  $\{-2.2\pi, -4.2\pi, \dots\}$

## SPECTRUM (MORE LINES)



## SPECTRUM (ALIASING CASE)



## DIGITAL FREQ $\hat{\omega}$ AGAIN

Normalized Radian Frequency

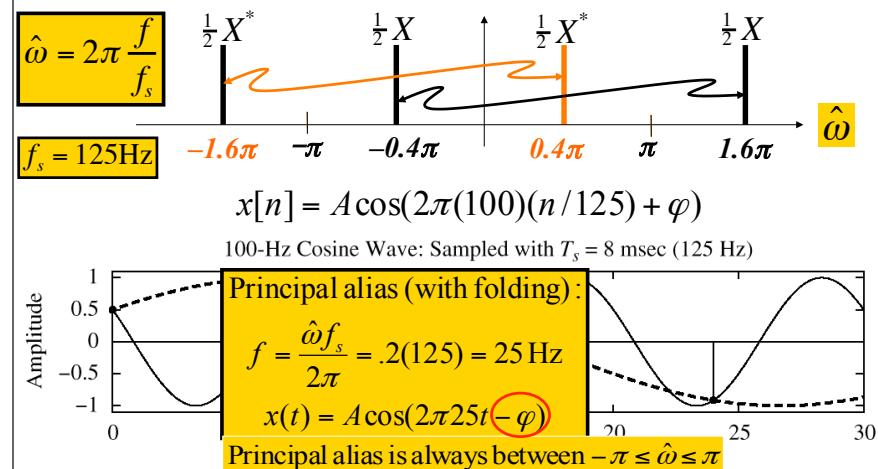
$$\hat{\omega} = \omega T_s = \frac{2\pi f}{f_s} + 2\pi \ell$$

**ALIASING**

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

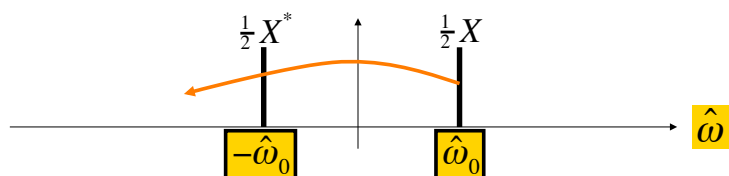
**FOLDED ALIAS**

## SPECTRUM (FOLDING CASE)



## SPECTRUM Explanation of SAMPLING THEOREM

- How do we prevent aliasing?
- Guarantee original signal is principal alias:



$$\hat{\omega}_0 - 2\pi < -\hat{\omega}_0 \Rightarrow \hat{\omega}_0 < \pi$$

$$\hat{\omega}_0 = \frac{2\pi f_0}{f_s} < \pi \Rightarrow f_0 < \frac{f_s}{2}$$

4/8/13

29

## D-to-A Reconstruction



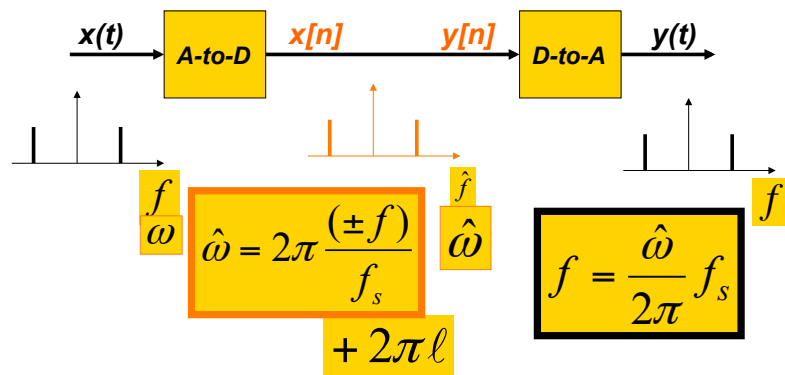
- Create continuous  $y(t)$  from  $y[n]$ 
  - IDEAL D-to-A:**
    - If you have formula for  $y[n]$
    - Invert sampling ( $t=nT_s$ ) by  $n=f_s t$
    - $y[n] = A \cos(0.2\pi n + \phi)$  with  $f_s = 8000$  Hz
    - $y(t) = A \cos(0.2\pi(8000t) + \phi) = A \cos(2\pi(800)t + \phi)$

4/8/13

© 2003, JH McClellan & RW Schaffer

30

## FREQUENCY DOMAINS



4/8/13

© 2003, JH McClellan & RW Schaffer

31

## D-to-A is AMBIGUOUS !

- ALIASING
  - Given  $y[n]$ , which  $y(t)$  do we pick ???
  - INFINITE NUMBER of  $y(t)$ 
    - PASSING THRU THE SAMPLES,  $y[n]$
  - D-to-A RECONSTRUCTION MUST CHOOSE ONE OUTPUT
- RECONSTRUCT THE **SMOOTHEST** ONE
  - THE **LOWEST** FREQ, if  $y[n] = \text{sinusoid}$

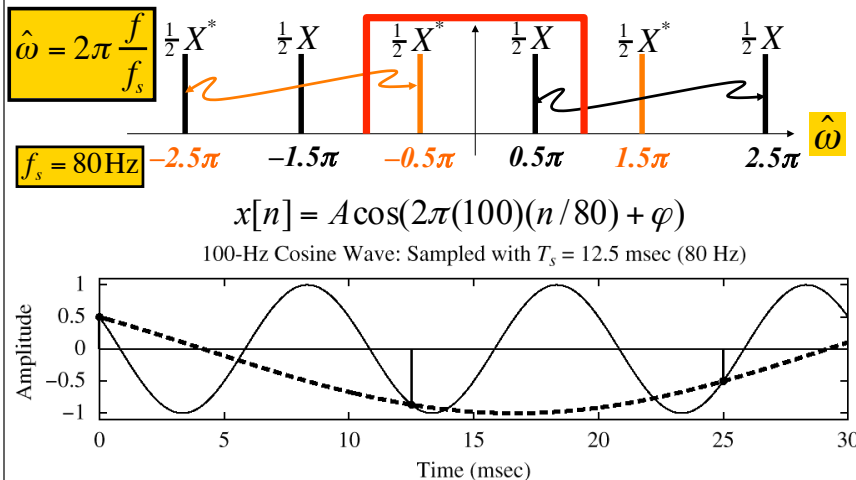
4/8/13

© 2003, JH McClellan & RW Schaffer

32



## SPECTRUM (ALIASING CASE)



## DEMOS from CHAPTER 4

- CD-ROM DEMOS
- SAMPLING DEMO (con2dis GUI)
  - Different Sampling Rates
    - Aliasing of a Sinusoid
- STROBE DEMO
  - Synthetic vs. Real
  - Television **SAMPLES** at 30 fps
- Sampling & Reconstruction

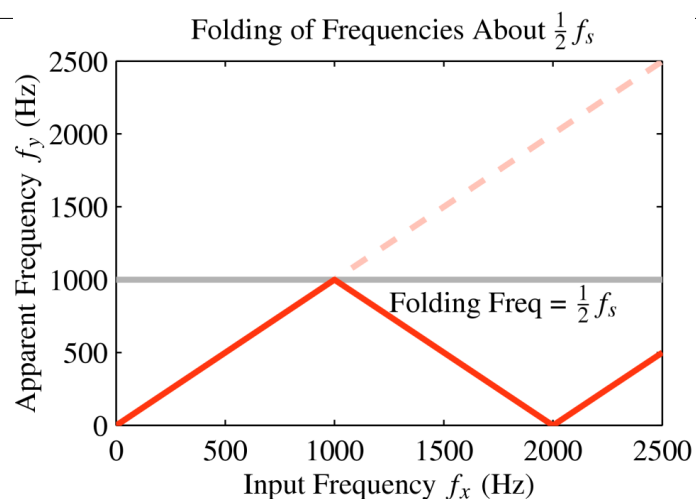
4/8/13

© 2003, JH McClellan & RW Schaffer

34

## FOLDING DIAGRAM

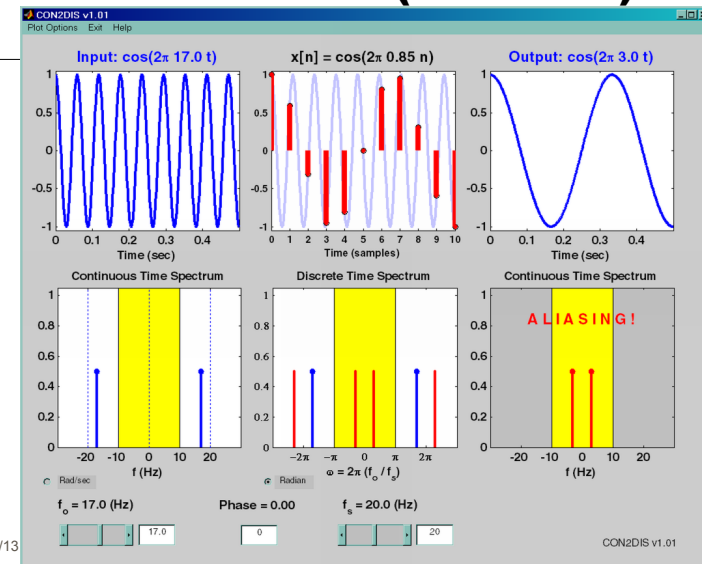
$f_s = 2000 \text{ Hz}$



4/8/13

35

## SAMPLING GUI (con2dis)



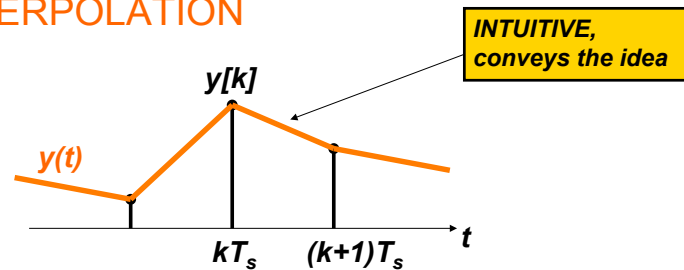
4/8/13

CON2DIS v1.01

36

## Reconstruction (D-to-A)

- CONVERT STREAM of NUMBERS to  $x(t)$
- “CONNECT THE DOTS”
- **INTERPOLATION**



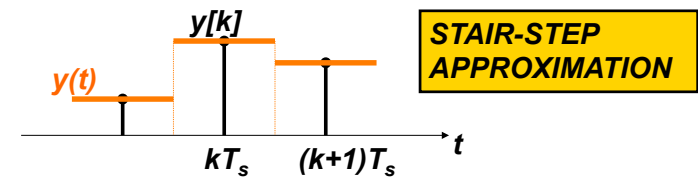
4/8/13

© 2003, JH McClellan & RW Schaefer

37

## SAMPLE & **HOLD** DEVICE

- CONVERT  $y[n]$  to  $y(t)$ 
  - $y[k]$  should be the value of  $y(t)$  at  $t = kT_s$
  - Make  $y(t)$  equal to  $y[k]$  for
    - $kT_s - 0.5T_s < t < kT_s + 0.5T_s$



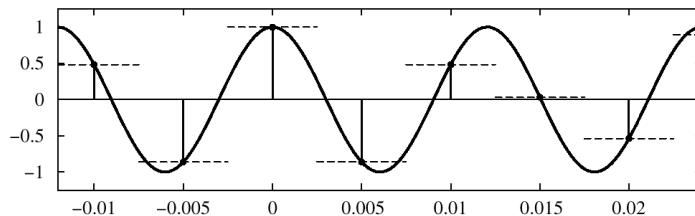
4/8/13

© 2003, JH McClellan & RW Schaefer

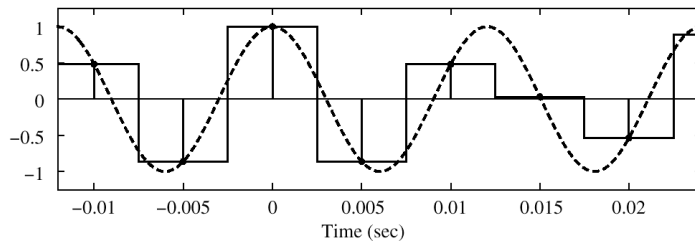
38

## SQUARE PULSE CASE

Sampling and Zero-Order Reconstruction:  $f_0 = 83$   $f_s = 200$



Original and Reconstructed Waveforms

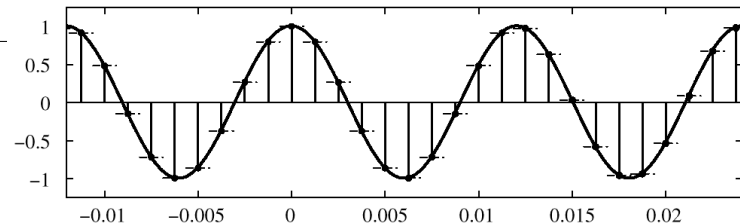


4

39

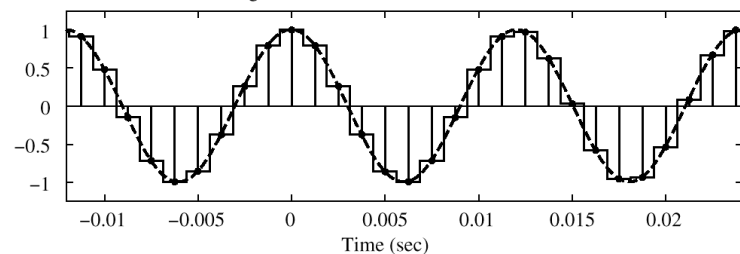
## **OVER-SAMPLING** CASE

Sampling and Zero-Order Reconstruction:  $f_0 = 83$   $f_s = 800$



**EASIER TO RECONSTRUCT**

Original and Reconstructed Waveforms



40

## MATH MODEL for D-to-A

$$y(t) = \sum_{n=-\infty}^{\infty} y[n]p(t - nT_s)$$

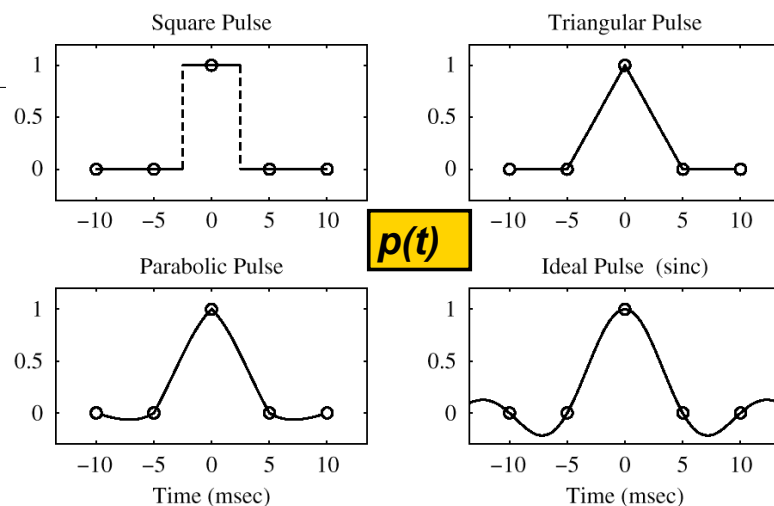
SQUARE PULSE:

$$p(t) = \begin{cases} 1 & -\frac{1}{2}T_s < t \leq \frac{1}{2}T_s \\ 0 & \text{otherwise} \end{cases}$$

## EXPAND the SUMMATION

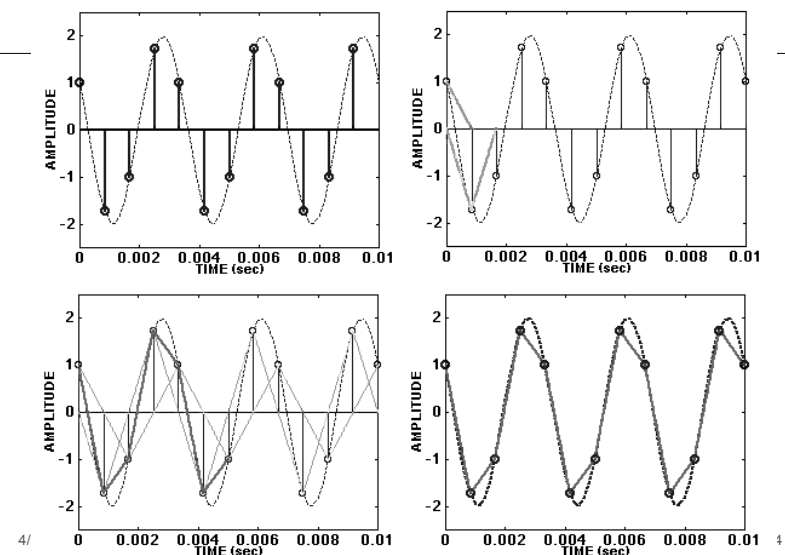
$$\sum_{n=-\infty}^{\infty} y[n]p(t - nT_s) = \dots + y[0]p(t) + y[1]p(t - T_s) + y[2]p(t - 2T_s) + \dots$$

- SUM of SHIFTED PULSES  $p(t - nT_s)$ 
  - “WEIGHTED” by  $y[n]$
  - CENTERED at  $t = nT_s$
  - SPACED by  $T_s$ 
    - RESTORES “REAL TIME”



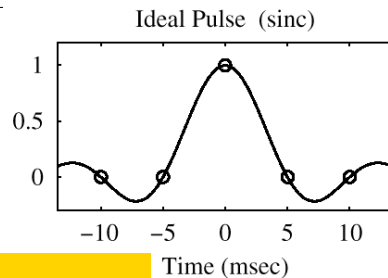
**Figure 4.17** Four different pulses for D-to-C conversion. The sampling period is  $T_s = 0.005$ , i.e.,  $f_s = 200$  Hz. Note that the duration of each pulse is approximately one or two times  $T_s$ .

## TRIANGULAR PULSE (2X)



## OPTIMAL PULSE ?

**CALLED  
"BANDLIMITED  
INTERPOLATION"**



$$p(t) = \frac{\sin \frac{\pi t}{T_s}}{\frac{\pi t}{T_s}} \quad \text{for } -\infty < t < \infty$$

$$p(t) = 0 \quad \text{for } t = \pm T_s, \pm 2T_s, \dots$$

4/8/13

© 2003, JH McClellan & RW Schaffer

45

## BANDLIMITED INTERPOLATION

- Interpolation formula

$$y(t) = \sum_{n=-\infty}^{\infty} y[n]p(t - nT_s)$$

- Corresponding Fourier transform

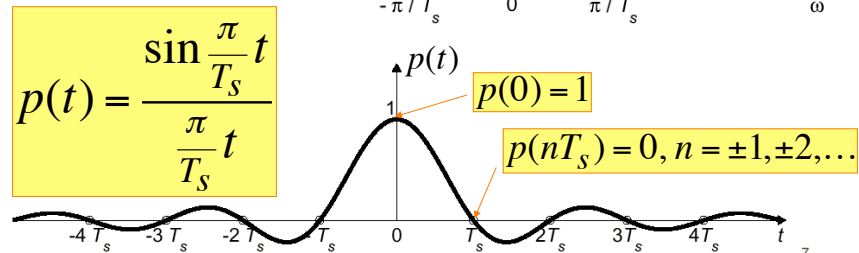
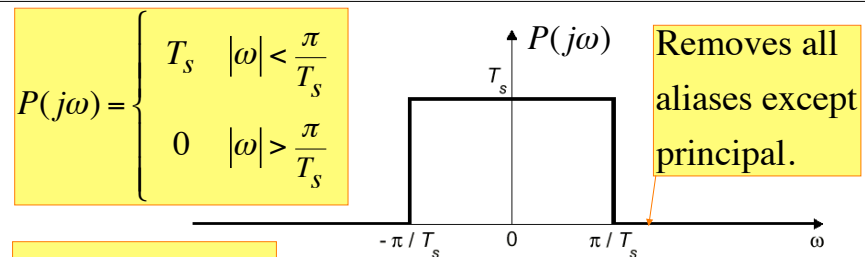
$$Y(j\omega) = \sum_{n=-\infty}^{\infty} y[n]e^{-j\omega n}P(j\omega) = \left( \sum_{n=-\infty}^{\infty} y[n]e^{-j\omega n} \right) P(j\omega)$$

4/8/13

EE-2025 Fall-2004 JMc

46

## Ideal Reconstruction Filter



## Spatial Aliasing



4/8/13

EE-2025 Fall-2010 JMc-BHJ

49

## Spatial Aliasing

