

# What is Digital Signal Processing?

## Techniques include (e.g. )

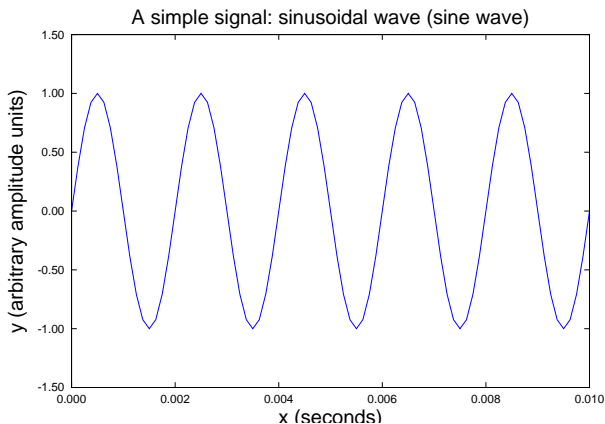
- ▶ Filtering
- ▶ Frequency domain techniques (*i.e.* Fourier)
- ▶ Time domain techniques
- ▶ Random signals
- ▶ Predication and Estimation (e.g. time series estimation)

## Example Applications

- ▶ Audio processing
- ▶ Communication systems
- ▶ Image processing
- ▶ Video processing
- ▶ Data compression
- ▶ Vehicle control
- ▶ Financial engineering

# What is a Signal?

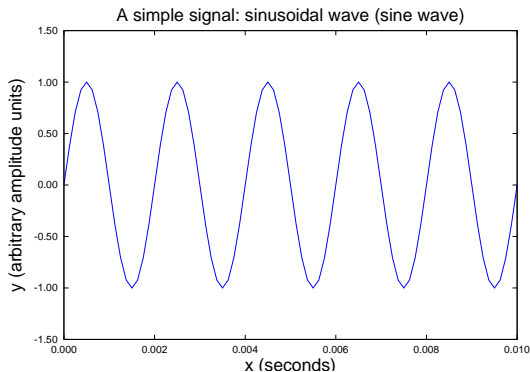
A simple example.



# What is a Signal?

Can contain information for

- ▶ *Communication*
- ▶ *Storage*
- ▶ *Calculation*



# Example of *What is a Signal?*

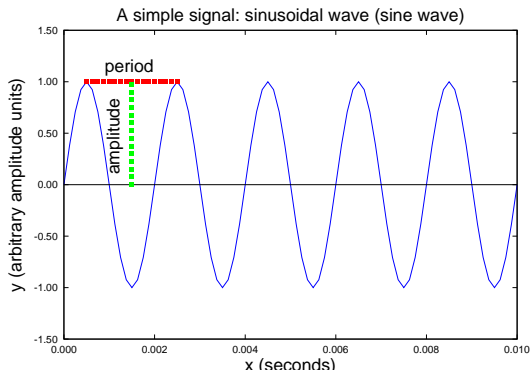
Information is carried in the

- ▶ *amplitude*, “ $a$ ”;
- ▶ *period*, “ $T$ ”;
- ▶ *frequency*, “ $f = 1/T$ ”;
- ▶ and *phase*, “ $\phi$ ”.

Equation for a sine wave:

$$y(x) = a \sin(2\pi f x + \phi)$$

where “ $x$ ” is time in seconds for this example. Amplitude “ $a = 1$ ” controls the height of the wave.

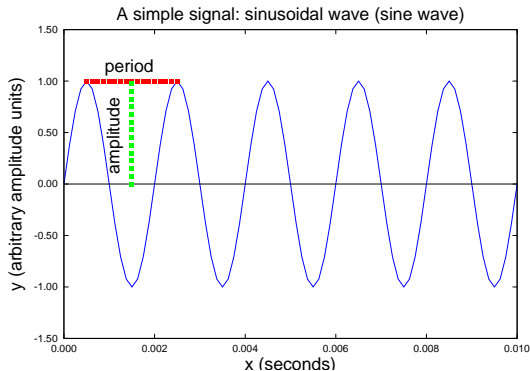


# Frequency and Period

Equation for a sine wave:

$$y(x) = a \sin(2\pi f x + \phi)$$

- ▶  $f$  is the frequency
- ▶ Measured in Hertz or Hz
- ▶ Here period,  
 $T = 0.002\text{s}$
- ▶  $f = 1/T$  Hz, therefore  
 $f = 1/0.002 = 500\text{Hz}$ .



# Phase

Equation for a sine wave:

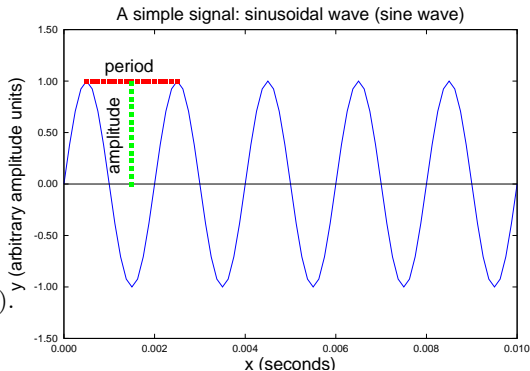
$$y(x) = a \sin(2\pi f x + \phi)$$

►  $\phi$  is the phase

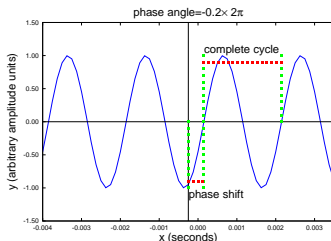
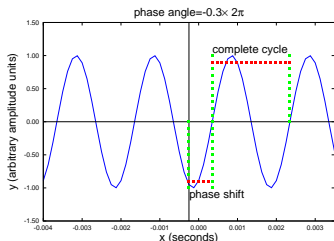
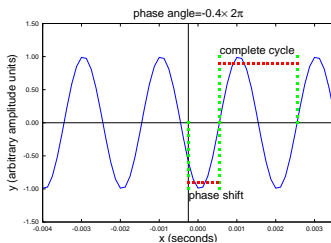
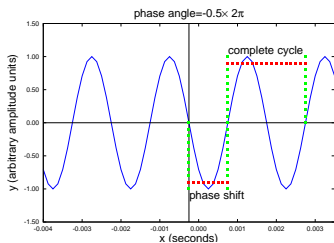
► Here  $\phi = 0$

Therefore here,

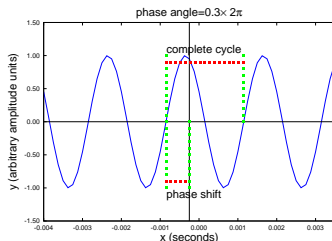
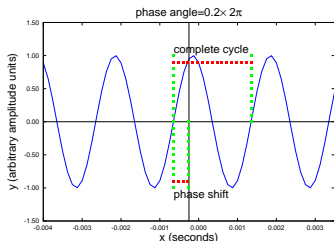
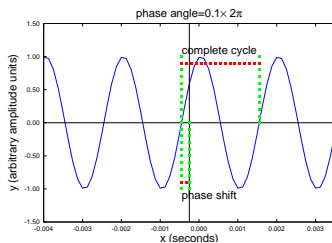
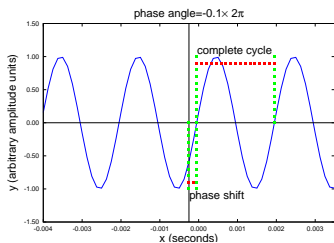
$$y(x) = y(x, \phi = 0) = a \sin(2\pi f x).$$



# Phase examples



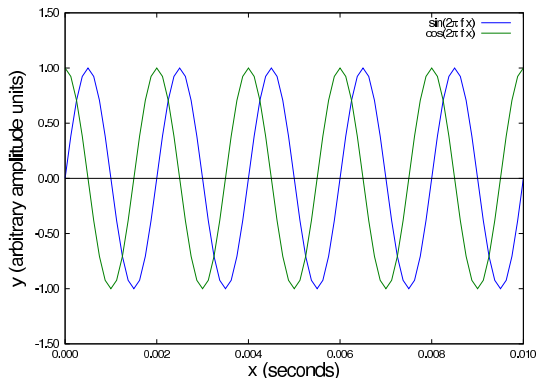
# Phase examples cont'd





# Cosine Vs Sine

Cosine and Sine functions are equivalent except for a phase shift ( $1/4 \times \text{period}$ ).

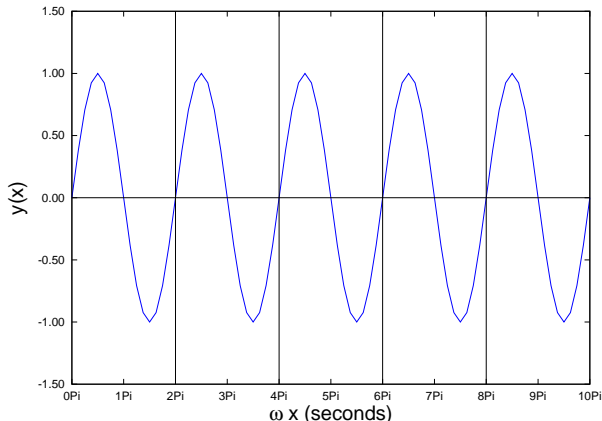


- ▶  $\cos(2\pi f x) = \sin(2\pi f x + \phi)$  where  $\phi = \pi/2$ .
- ▶  $\sin(2\pi f x) = \cos(2\pi f x + \phi)$  where  $\phi = -\pi/2$ .

# Angular Frequency

- Frequency,  $f = 1/T$
- Angular frequency,  
 $\omega = 2\pi f$
- 1 period or cycle =  $2\pi$  radians

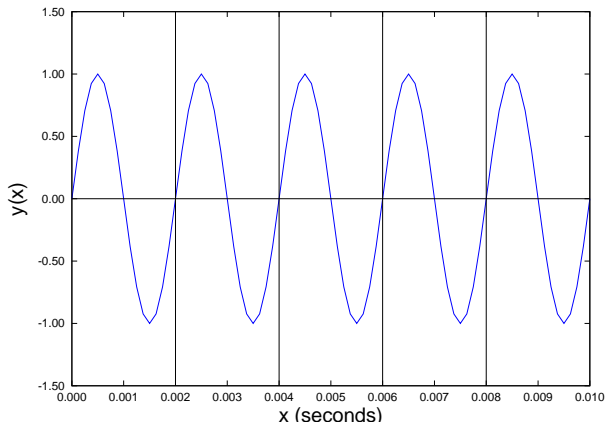
$$\begin{aligned}y(x) &= \sin(2\pi f x + \phi) \\ &= \sin(\omega x + \phi)\end{aligned}$$



# Angular Frequency

- Frequency,  $f = 1/T$
- Angular frequency,  
 $\omega = 2\pi f$
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$$\begin{aligned}y(x) &= \sin(2\pi f x + \phi) \\ &= \sin(\omega x + \phi)\end{aligned}$$



# Phasor Representation

A cosine (or sine) wave:

$$y(x) = a \cos(\omega x + \phi)$$

can be represented as a phasor.

A phasor is a complex number:

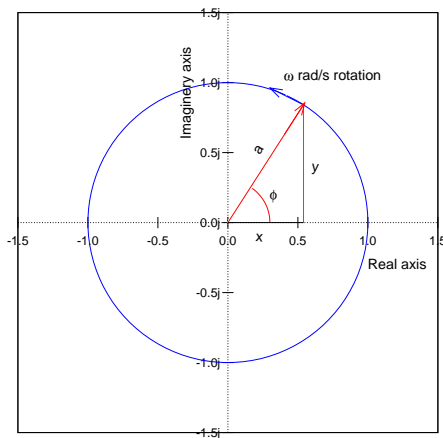
$$z = x + jy = a(\cos(\phi) + j \sin(\phi))$$

where  $x$  is known as the real part or  $\text{Re}(z) = x$  and  $y$  is known as the imaginary part or  $\text{Im}(z) = y$ .

$x$  and  $y$  can be calculated with  
 $x = a \cos(\phi)$  and  $y = a \sin(\phi)$ .

Also remember  $j = \sqrt{-1}$ .

Argand or Phasor Diagram:



# Complex Numbers

The square root of minus one is not defined so a symbol,  $j$  is used (sometimes  $i$ ):

$$j = \sqrt{-1}.$$

Powers:

- ▶  $j^2 = -1$
- ▶  $j^3 = -j$
- ▶  $j^{-1} = 1/j = -j$

If  $z = x + jy$  (rectangular form) then alternative representations are:

▶ Polar form:  $z = a\angle\phi$

▶ Exponential form:  
 $z = a \exp(j\phi)$

where  $a = \sqrt{x^2 + y^2}$  and  
 $\phi = \tan^{-1}(y/x)$ .

# Properties of Complex Numbers

If  $z = x + jy$ ,  $z_1 = x_1 + jy_1$  and  $z_2 = x_2 + jy_2$  then

► Addition:

$$z_1 + z_2 = x_1 + x_2 + j(y_1 + y_2)$$

► Subtraction:

$$z_1 - z_2 = x_1 - x_2 + j(y_1 - y_2)$$

► Multiplication:

$$z_1 z_2 = a_1 a_2 \angle(\phi_1 + \phi_2)$$

► Division:

$$z_1 / z_2 = a_1 / a_2 \angle(\phi_1 - \phi_2)$$

► Reciprocal:  $1/z = 1/a \angle(-\phi)$

► Square root:  $\sqrt{z} = \sqrt{a} \angle(\phi/2)$

► Complex conjugate:

$$z^* = x - jy = a \angle -\phi$$

The polar form simplifies some operations such as multiplication and division of complex numbers.

# Phasor Representation

Euler's identity:

$$\exp(j\phi) = \cos(\phi) + j \sin(\phi)$$

Therefore

- ▶  $\cos(\phi) = \operatorname{Re}(\exp(j\phi)) \longrightarrow$  or the real part,  $x$
- ▶  $\sin(\phi) = \operatorname{Im}(\exp(j\phi)) \longrightarrow$  or the imaginary part,  $y$

Recall the cosine wave:

$$y(x) = \cos(\omega x + \phi)$$

which can be written as:

$$\begin{aligned} y(x) &= \operatorname{Re}(a \exp(j(\omega x + \phi))) = \operatorname{Re}(a \exp(j\omega x) \exp(j\phi)) \\ &= \operatorname{Re}(A \exp(j\omega x)) \end{aligned}$$

where  $A$  is the phasor representation of  $y(x)$  given by

$$A = a \exp(j\phi) = a \angle(\phi).$$

# Complex Exponentials, Sines and Cosines

Given

- ▶  $y_1(x) = b \exp(j\omega x) = b \cos(\omega x) + jb \sin(\omega x)$
  - ▶  $y_2(x) = b \exp(-j\omega x) = b \cos(\omega x) + jb \sin(-\omega x)$
- as
- ▶  $\cos(-\omega x) = \cos(\omega x)$  (even function)
  - ▶  $\sin(-\omega x) = -\sin(\omega x)$  (odd function)

Then

$$y_1(x) + y_2(x) = 2b \cos(\omega x).$$

So that

$$b \cos(\omega x) = \frac{a}{2} \exp(j\omega x) + \frac{a}{2} \exp(-j\omega x).$$

A similar approach can be used to derive a sine function.



# Outline

## Course Format

Course Outline

## Digital Signal Processing

What is Digital Signal Processing?

Phase

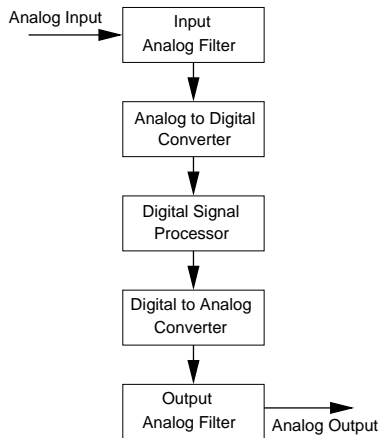
Phasors and Complex Numbers

## A Typical Digital Signal Processing System

## Summary

Lecture summary

# A Typical Digital Signal Processing System



- ▶ Input Analog Filter  
(antialiasing):  
*Limits frequency range*
- ▶ Analog to Digital Converter  
*Converts signal to digital samples*
- ▶ Digital Signal Processor  
*Storage, Communication and or Calculations*
- ▶ Digital to Analog Converter  
*Convert to continuous signal*
- ▶ Output Analog Filter  
*Removes sharp transitions*

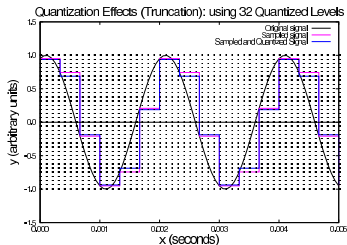
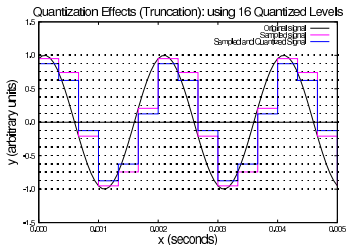
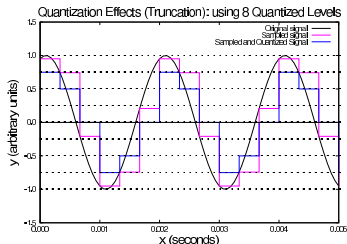
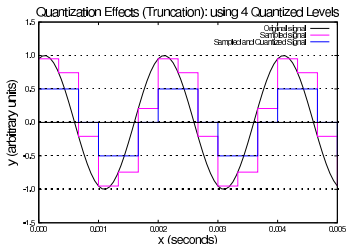
# Analog to Digital Converter (ADC)

- ▶ Real world is typically *analog* (continuous)
- ▶ Digital signal approximates analog signal with discrete quantised samples
- ▶ ADC converts an analog signal to a digital signal
- ▶ Signal is digitised in two ways:
  - ▶ Signal is sampled at a sampling rate or frequency: Information is collected about the signal at regular intervals.
  - ▶ The continuous or analog signal is then quantised: *i.e.* put into digital form, where only a finite set of numbers are represented.

# Quantisation using Truncation

- ▶ Signal can be quantised using e.g. truncation where numbers following specified position are removed.
- ▶ Examples:
  - ▶ 5.7 truncated to integer is 5
  - ▶ 5.11 truncated to 1 decimal place is 5.1
- ▶ Negative numbers are truncated in the same way (note different to the common *floor* function in matlab), e.g.
  - ▶ -5.78 truncated to integer is -5
  - ▶ -5.135 truncated to 2 decimal places is -5.13

# Truncation Quantisation examples

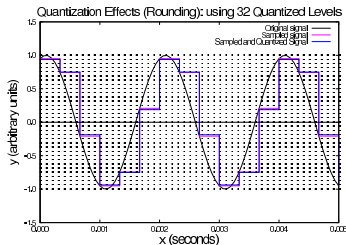
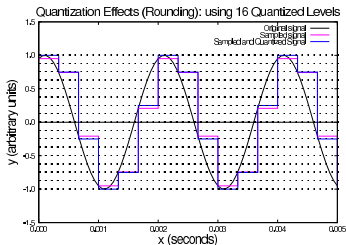
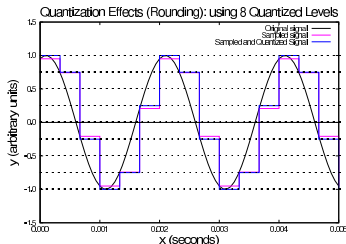
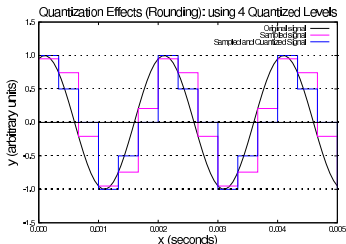


- Errors can be seen between the sampled and the sampled and quantized signals.

# Quantisation using Rounding

- ▶ Rounding can be a quantization method associated with smaller errors, e.g.
  - ▶ 5.7 rounded to nearest integer is 6
  - ▶ 5.11 rounded to 1 decimal place is 5.1
  - ▶ -5.78 rounded to nearest integer is -6
  - ▶ -5.135 rounded to 2 decimal places is -5.14

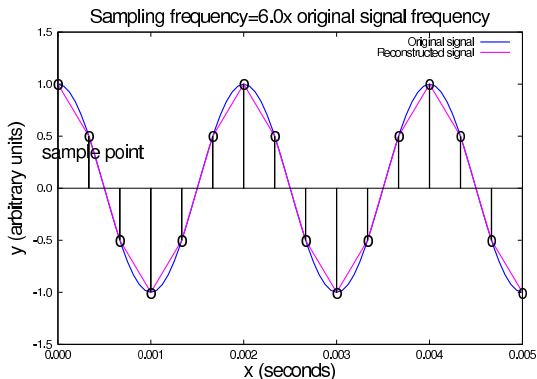
# Rounding Quantisation examples



- Errors can be seen between the sampled and the sampled and quantized signals.

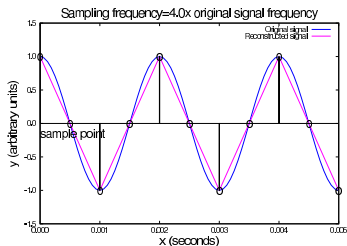
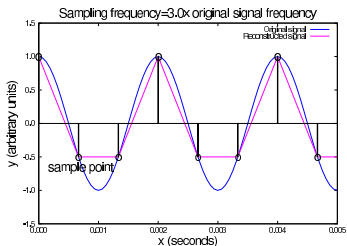
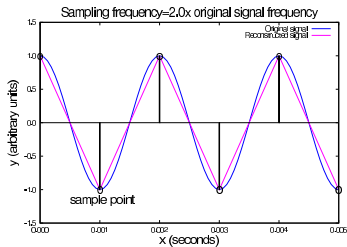
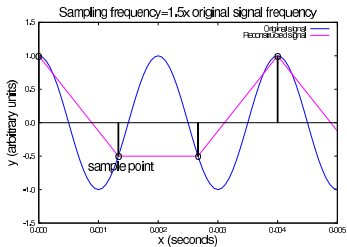
# Sampling

- ▶ Sampling also affects the quality of the digitised signal.
- ▶ Higher sampling rate reduces error and enables better representation of the original analog signal in digital form.

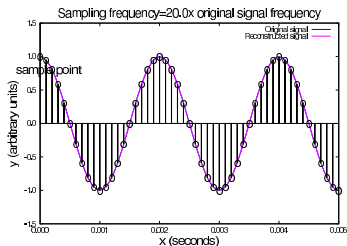
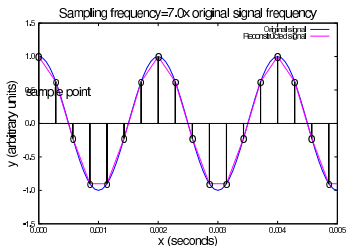
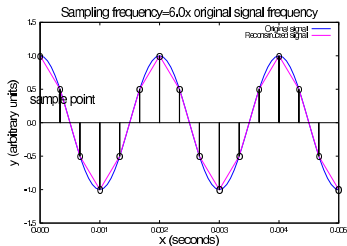
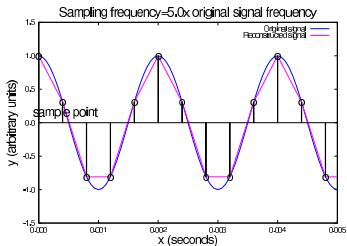




# Sampling examples

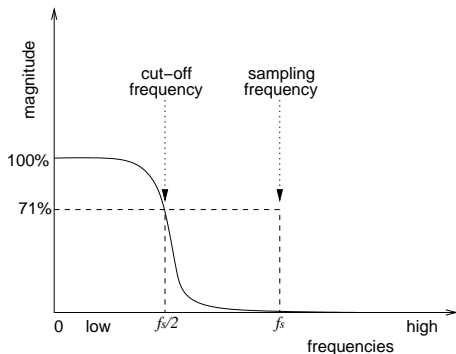


# Sampling examples cont'd



# Input Analog Filter: Antialiasing Filter

- ▶ Analog to Digital Converter (ADC) requires signal below a particular frequency (Nyquist Frequency)
- ▶  $\therefore$  Limit frequency range to below Nyquist frequency ( $f_s/2$ ) before Analog to Digital Conversion.



- ▶ Otherwise next stage produces frequency errors (*i.e.* aliasing)
- ▶ Sampling produces copies of signal at multiples of sampling frequency
- ▶ Aliasing occurs when copies of signal overlap each other

# Digital Signal Processor

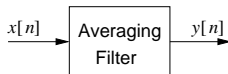
- ▶ After digitisation (with the ADC) digital signal processing may then be performed on the digitised signal.
- ▶ Simple example
  - ▶ Averaging filter:

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

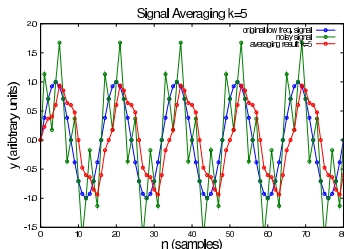
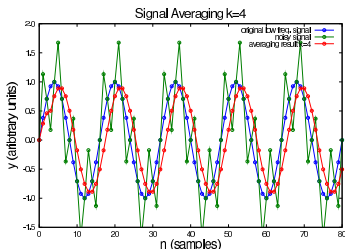
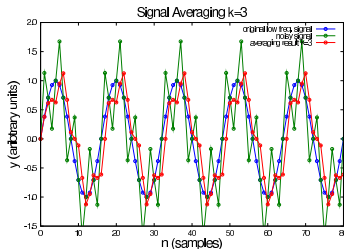
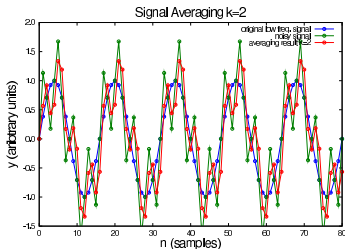
for window width  $k = 3$

$$y[n] = \frac{x[n] + x[n-1] + x[n-2]}{3}$$

where  $x[n]$  is an input value at sample time  $n$  and  $y[n]$  is an output at sample time  $n$

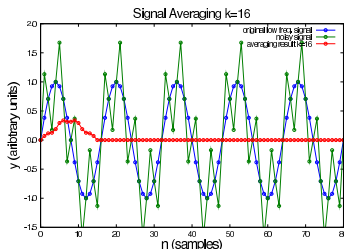
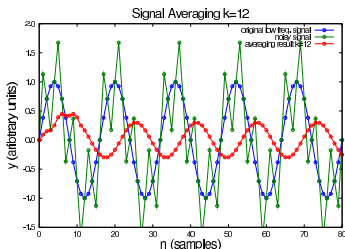
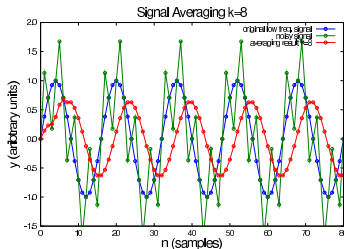
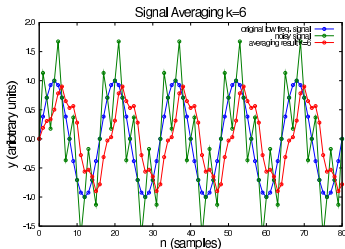


# Averaging Filter Examples



Window width  $k$  controls the response of the filter. If  $k$  is too low, there is little benefit on output signal.

# Averaging Filter Examples cont'd



Window width  $k$  controls the response of the filter. If  $k$  is too high, the filter removes all of the output signal.

# Outline

## Course Format

Course Outline

## Digital Signal Processing

What is Digital Signal Processing?

Phase

Phasors and Complex Numbers

## A Typical Digital Signal Processing System

## Summary

Lecture summary

# What have we covered today?

- ▶ Course content
- ▶ Definition of digital signal processing
- ▶ Description of phase
- ▶ Cosine and Sine functions
- ▶ Complex numbers and alternative representations
- ▶ A typical digital signal processing system