



Biomedical Signal Analysis

Hsiao-Lung Chan, Ph.D.
Department of Electrical Engineering,
Chang-Gung University, Taiwan
chanhl@mail.cgu.edu.tw

Outline

- Spectral analysis
- Digital filters



Joseph Fourier

lived from 1768 to 1830

Fourier studied the mathematical theory of heat conduction. He established the partial differential equation governing heat diffusion and solved it by using infinite series of trigonometric functions.

Find out more at:

<http://www-history.mcs.st-andrews.ac.uk/history/Mathematicians/Fourier.html>

Fourier transform

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt$$

$\underbrace{\hspace{1.5cm}}$ $\underbrace{\hspace{1.5cm}}$
Spectrum at Basis function
frequency f for frequency f

$$X(\omega) = \int_{-\infty}^{\infty} x(t) e^{-j\omega t} dt$$

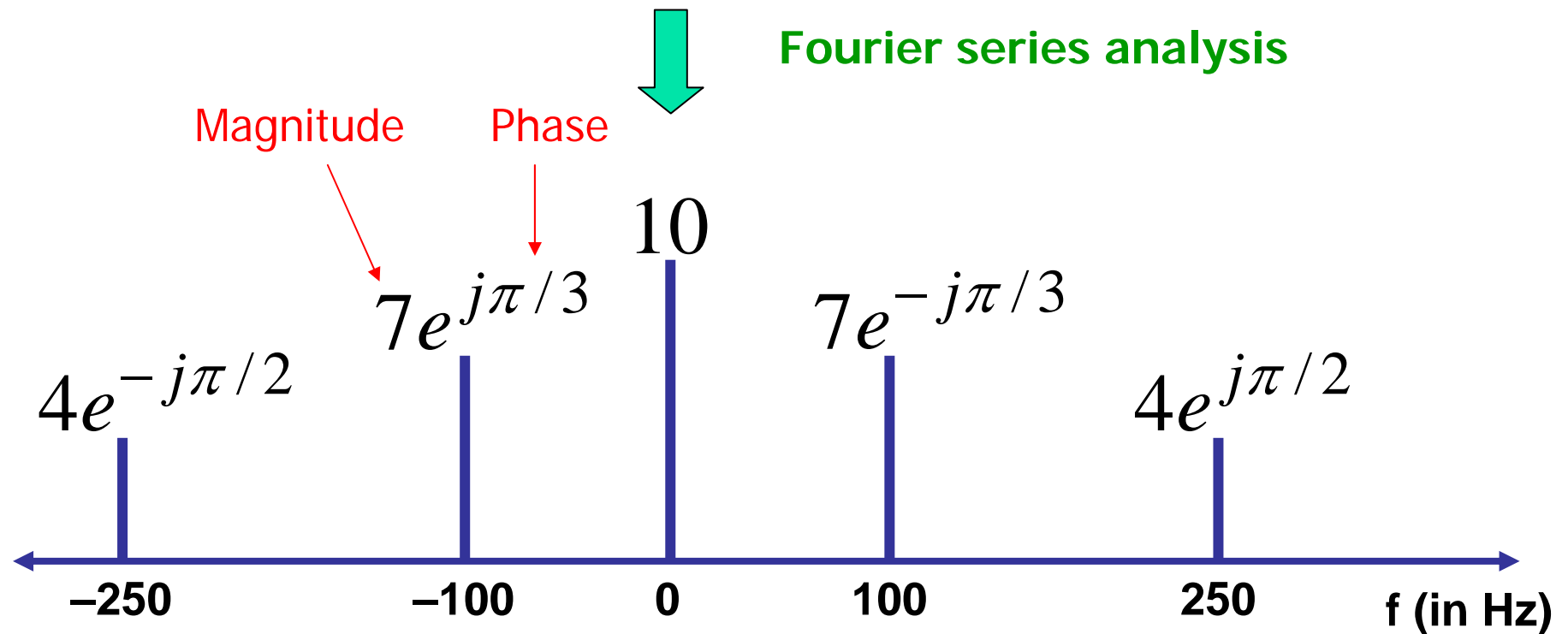
Inverse Fourier transform

$$x(t) = \int_{-\infty}^{\infty} X(f) e^{-j2\pi ft} df$$

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega) e^{j\omega t} d\omega$$

Example

$$x(t) = 10 + 14 \cos(2\pi(100)t - \pi/3) + 8 \cos(2\pi(250)t + \pi/2)$$



Magnitude and phase

$$X(f) = |X(f)| e^{j\phi(f)}$$

where magnitude and phase spectra

$$|X(f)| = \sqrt{\{\operatorname{Re}[X(f)]\}^2 + \{\operatorname{Im}[X(f)]\}^2}$$

$$\phi(f) = \tan^{-1} \left\{ \frac{\operatorname{Im}[X(f)]}{\operatorname{Re}[X(f)]} \right\}$$

% Generating two sinusoids and one DC component

fs=2000; % sampling at 2 kHz

t=0:1/fs:0.1;

$x = 10 + 14 \cdot \cos(2\pi \cdot 100 \cdot t - \pi/3) + 4 \cdot \cos(2\pi \cdot 250 \cdot t - \pi/2);$

subplot(2,2,1)

plot(t,x)

ylabel('x(n)')

xlabel('Time, s')

title('Two sinusoids + DC')

axis([min(t) max(t) min(x)*1.1 max(x)*1.1])

% Spectral analysis

```
Xf=fft(x);
```

```
resolution=fs/length(Xf);
```

```
f=(0:length(Xf)-1)*resolution;
```

```
Xf_mag = abs(Xf); % magnitude of spectrum
```

```
subplot(2,2,2)
```

```
plot(f,Xf_mag)
```

```
xlabel('Frequency, Hz')
```

```
ylabel('Magnitude')
```

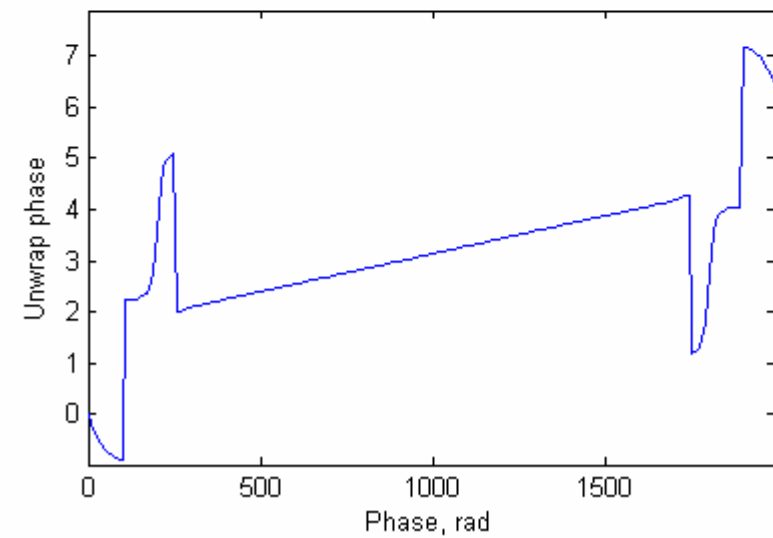
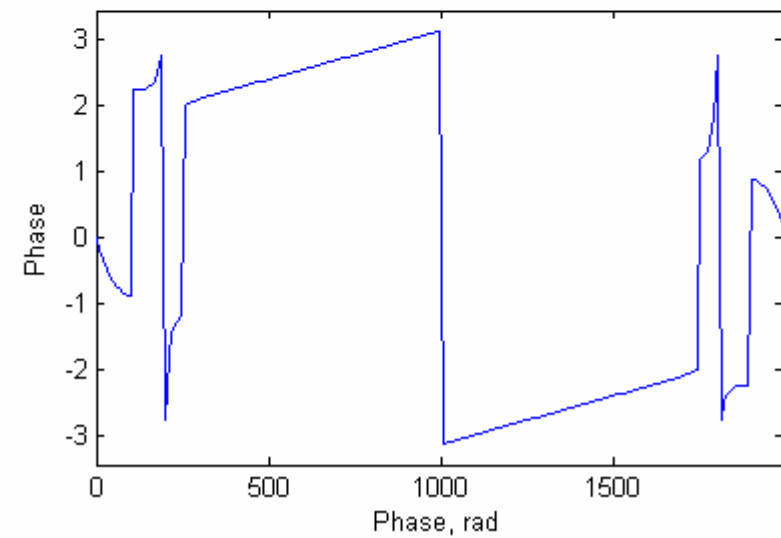
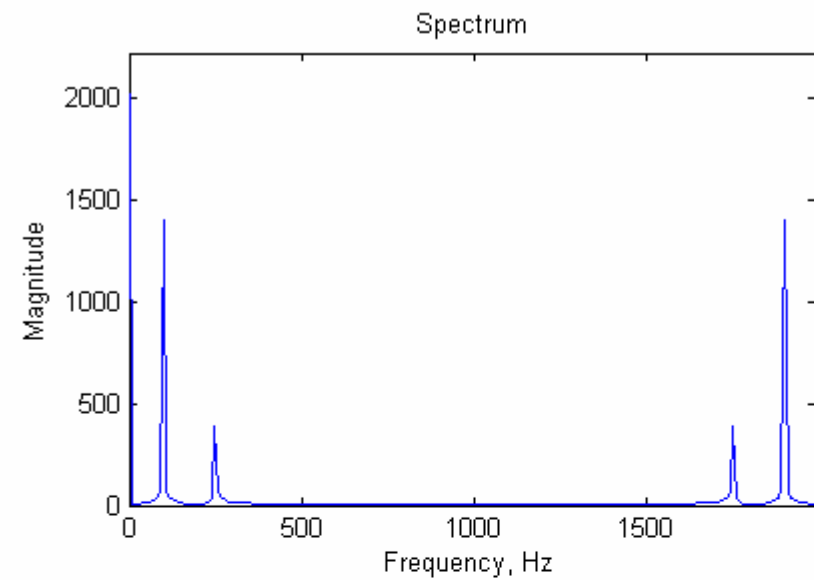
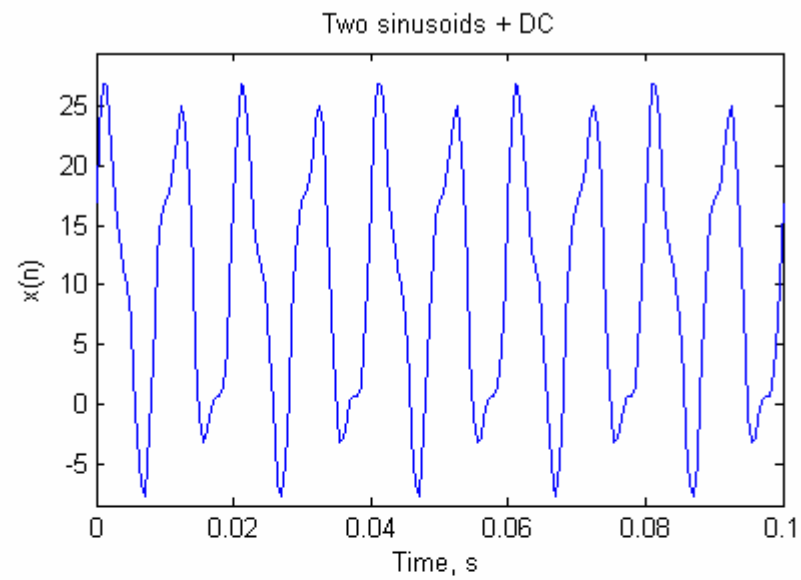
```
title('Spectrum')
```

```
axis([min(f) max(f) 0 max(Xf_mag)*1.1])
```



```
Xf_phase = angle(Xf);           % phase of spectrum  
subplot(2,2,3)  
plot(f,Xf_phase)  
xlabel('Phase, rad'); ylabel('Phase');  
axis([min(f) max(f) min(Xf_phase)*1.1 max(Xf_phase)*1.1])
```

```
Xf_phase = unwrap(Xf_phase);    % Unwrap phase angle  
subplot(2,2,4)  
plot(f,Xf_phase)  
xlabel('Phase, rad'); ylabel('Unwrap phase');  
axis([min(f) max(f) min(Xf_phase)*1.1 max(Xf_phase)*1.1])
```



Effect of data points on spectral analysis

```
fs=500;  
t=0:1/fs:0.3;  
x = cos(2*pi*25*t) + cos(2*pi*35*t+pi/10);  
  
Xf=fft(x);  
resolution=fs/length(Xf);  
f=(0:length(Xf)-1)*resolution;  
Xf_power = Xf.*conj(Xf); % power spectral density  
index=1:length(Xf)/4;  
stem(f(index),Xf_power(index))
```

Effect of data points on spectral analysis

% Improve resolution by increase data length

```
t=0:1/fs:0.3*5;
```

```
x = cos(2*pi*25*t) + cos(2*pi*35*t+pi/10);
```

```
Xf=fft(x);
```

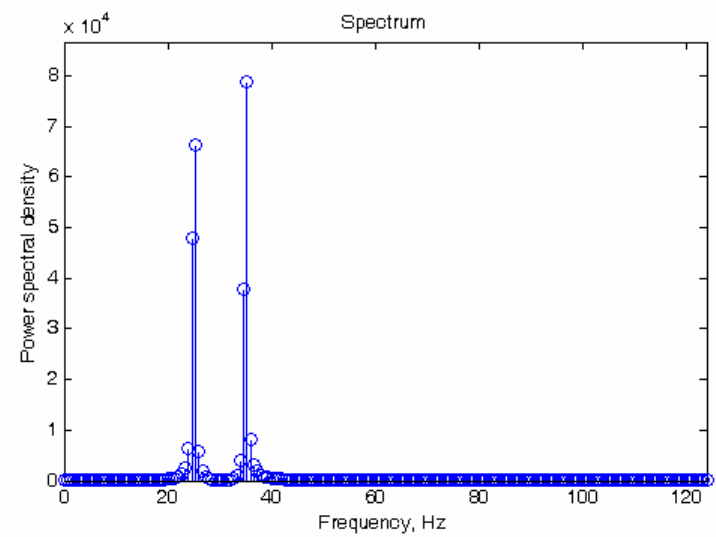
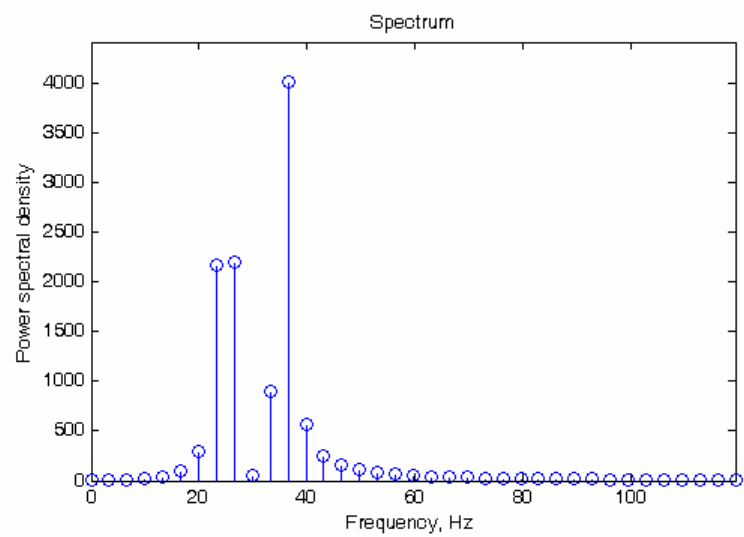
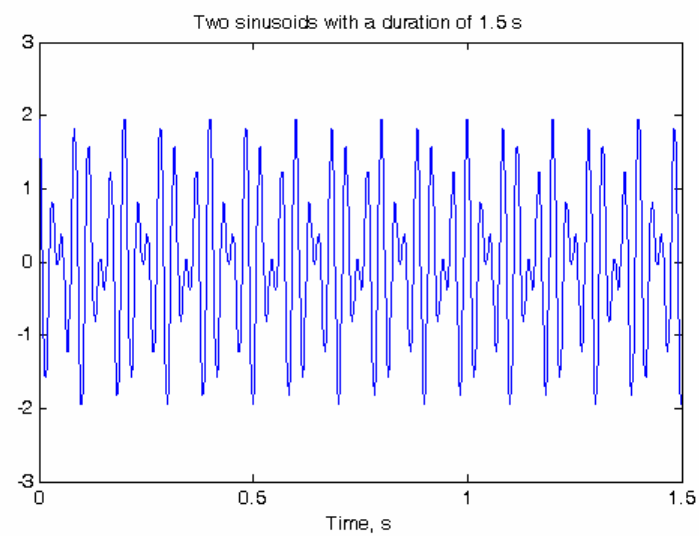
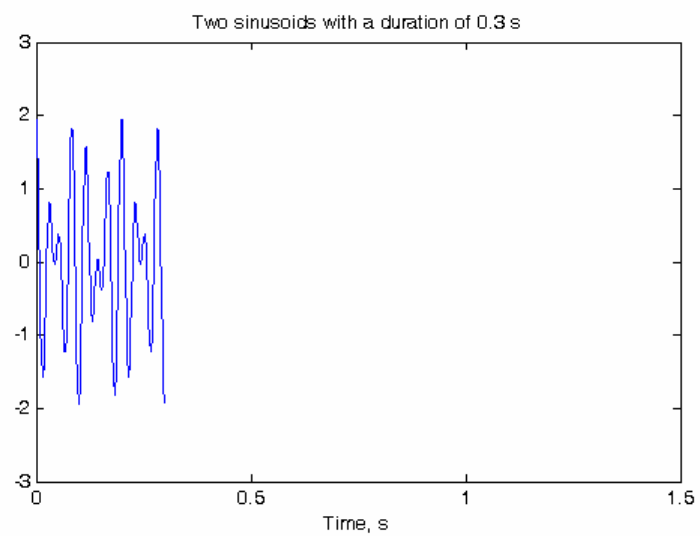
```
resolution=fs/length(Xf);
```

```
f=(0:length(Xf)-1)*resolution;
```

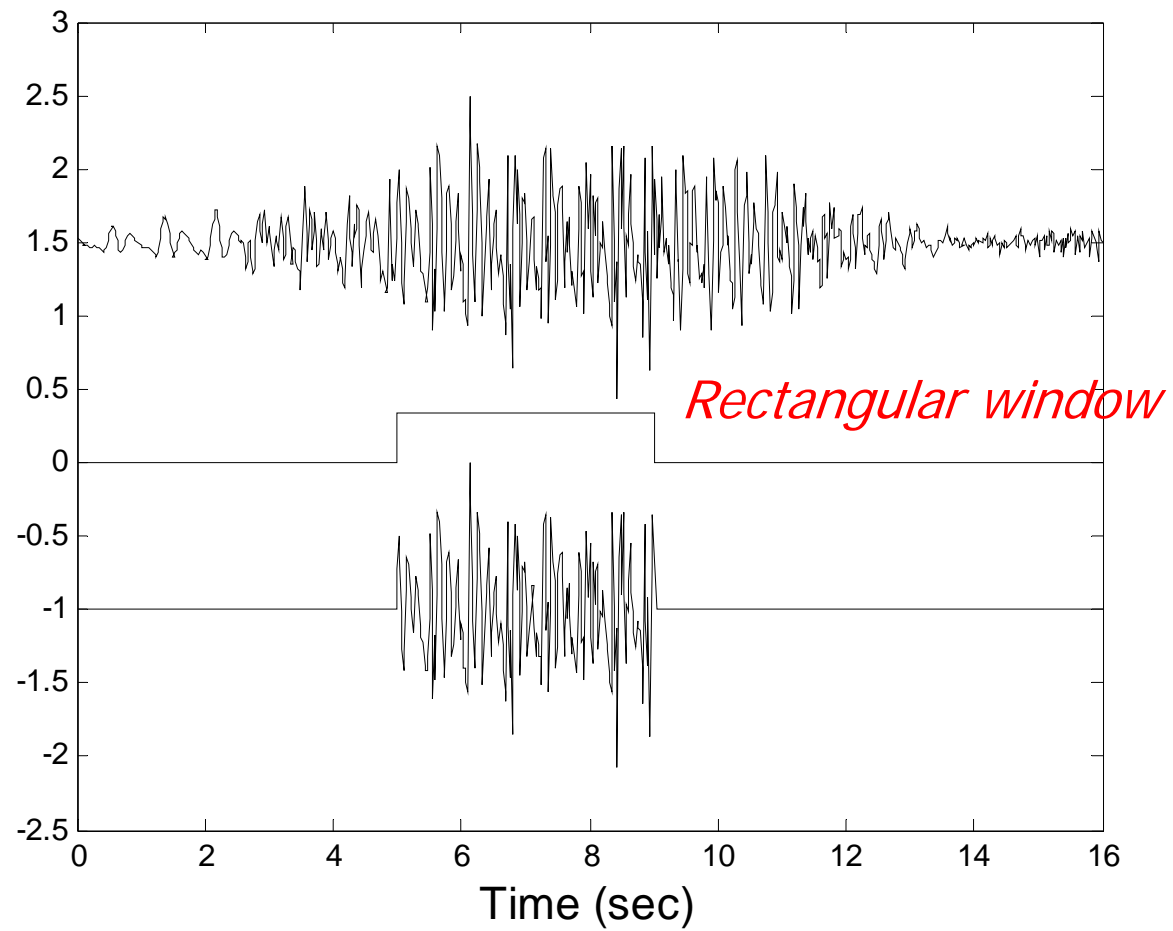
```
Xf_power = Xf.*conj(Xf); % power spectral density
```

```
index=1:length(Xf)/4;
```

```
stem(f(index),Xf_power(index))
```



Data length: truncation



Window functions

Rectangular:

$$w(n) = 1$$

Blackman:

$$w(n) = 0.42 - 0.5 \cos\left(\frac{2\pi n}{N-1}\right) + 0.08 \cos\left(\frac{4\pi n}{N-1}\right)$$

Hamming:

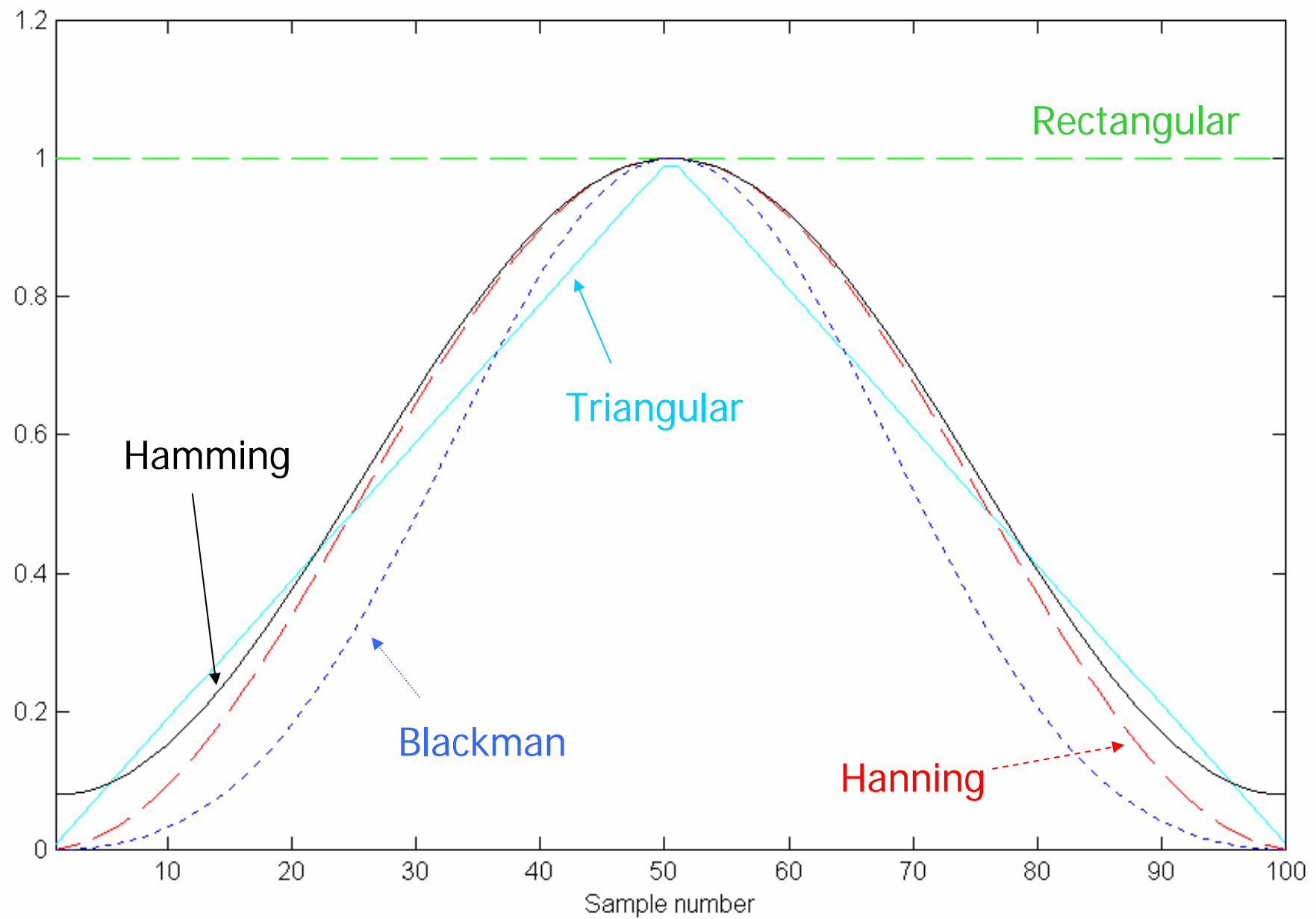
$$w(n) = 0.54 - 0.46 \cos\left(\frac{2\pi n}{N-1}\right)$$

Bartlett (Triangular):

$$w(n) = \begin{cases} \frac{2n}{N-1}, & 0 \leq n \leq \frac{N-1}{2} \\ 2 - \frac{2n}{N-1}, & \frac{N-1}{2} \leq n \leq N-1 \end{cases}$$

Hanning:

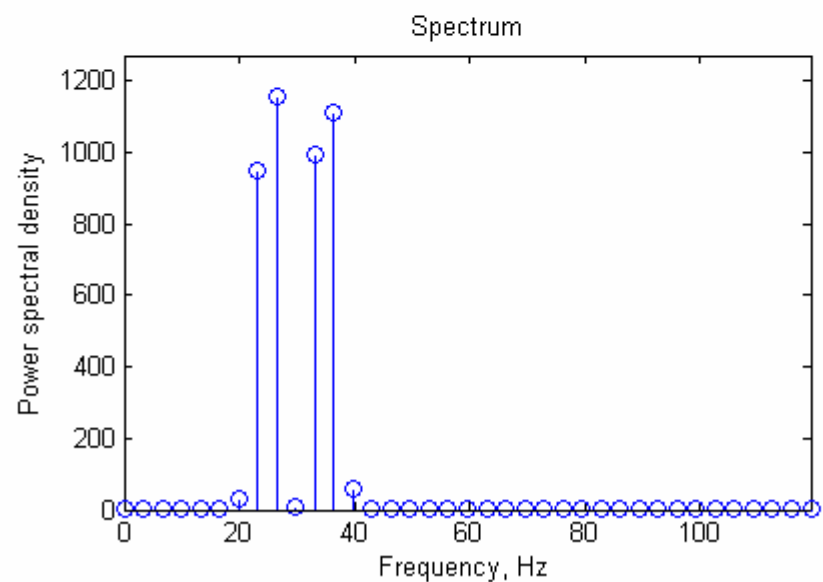
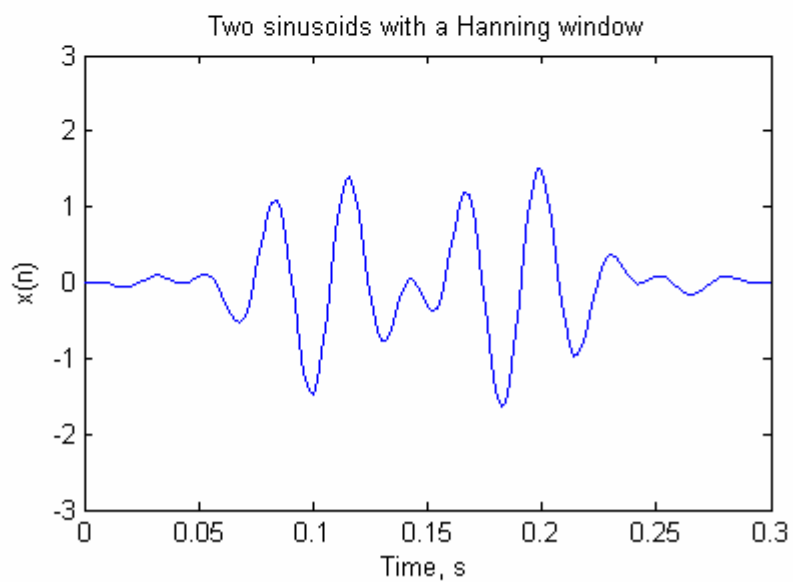
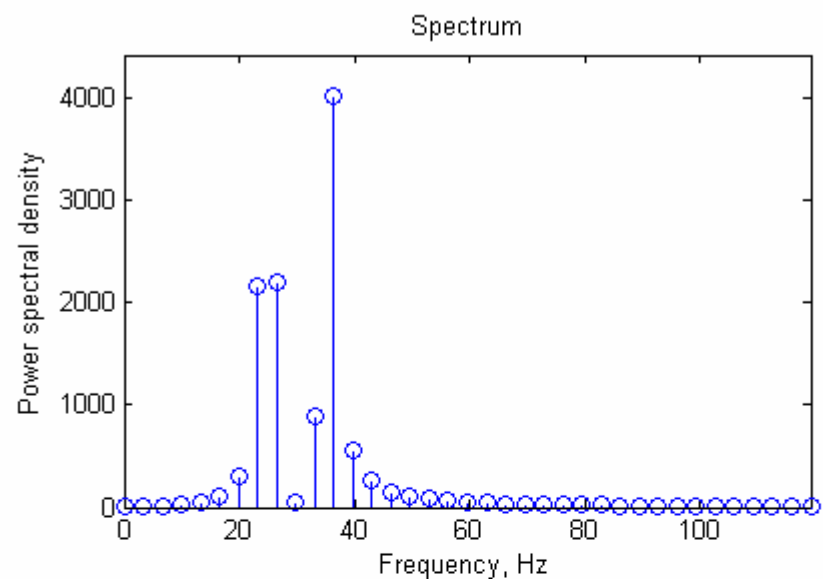
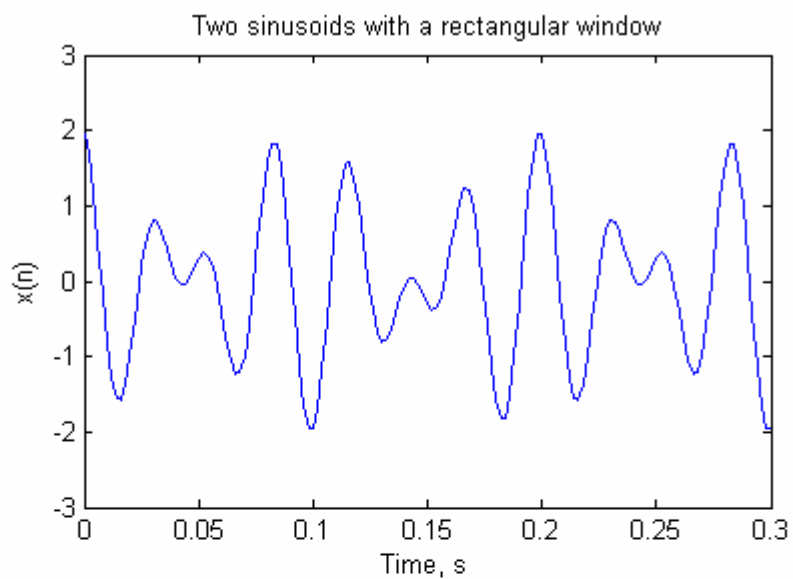
$$w(n) = 0.5 - 0.5 \cos\left(\frac{2\pi n}{N-1}\right)$$



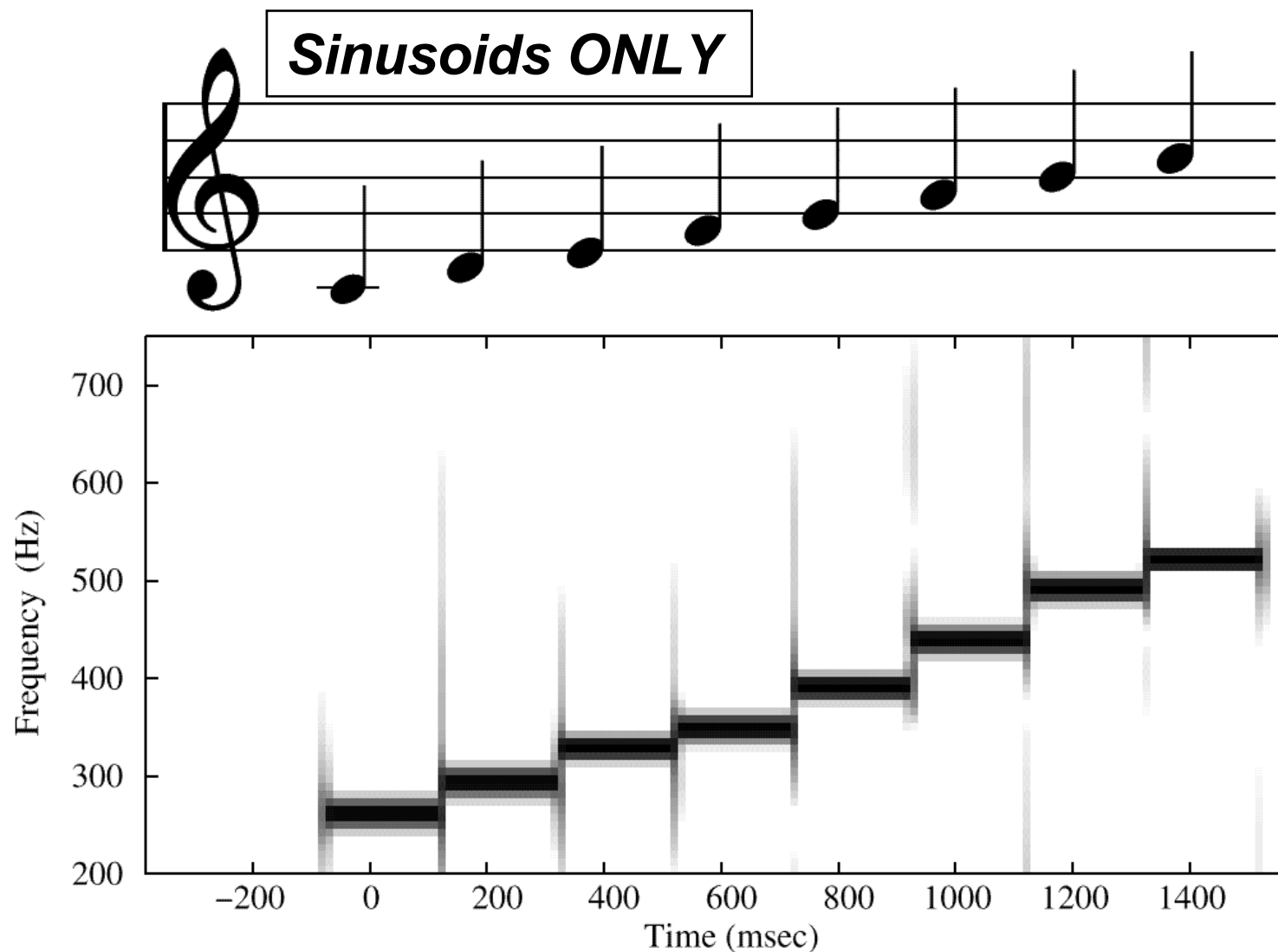
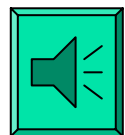

```
fs=500;  
t=0:1/fs:0.3;  
x = cos(2*pi*25*t) + cos(2*pi*35*t+pi/10);  
  
Xf=fft(x);  
resolution=fs/length(Xf);  
f=(0:length(Xf)-1)*resolution;  
Xf_power = Xf.*conj(Xf); % power spectral density  
index=1:length(Xf)/4;  
stem(f(index),Xf_power(index))
```

% Using Hanning window

```
x=x.*hanning(length(x))';  
Xf=fft(x);  
resolution=fs/length(Xf);  
f=(0:length(Xf)-1)*resolution;  
Xf_power = Xf.*conj(Xf); % power spectral density  
index=1:length(Xf)/4;  
stem(f(index),Xf_power(index))
```

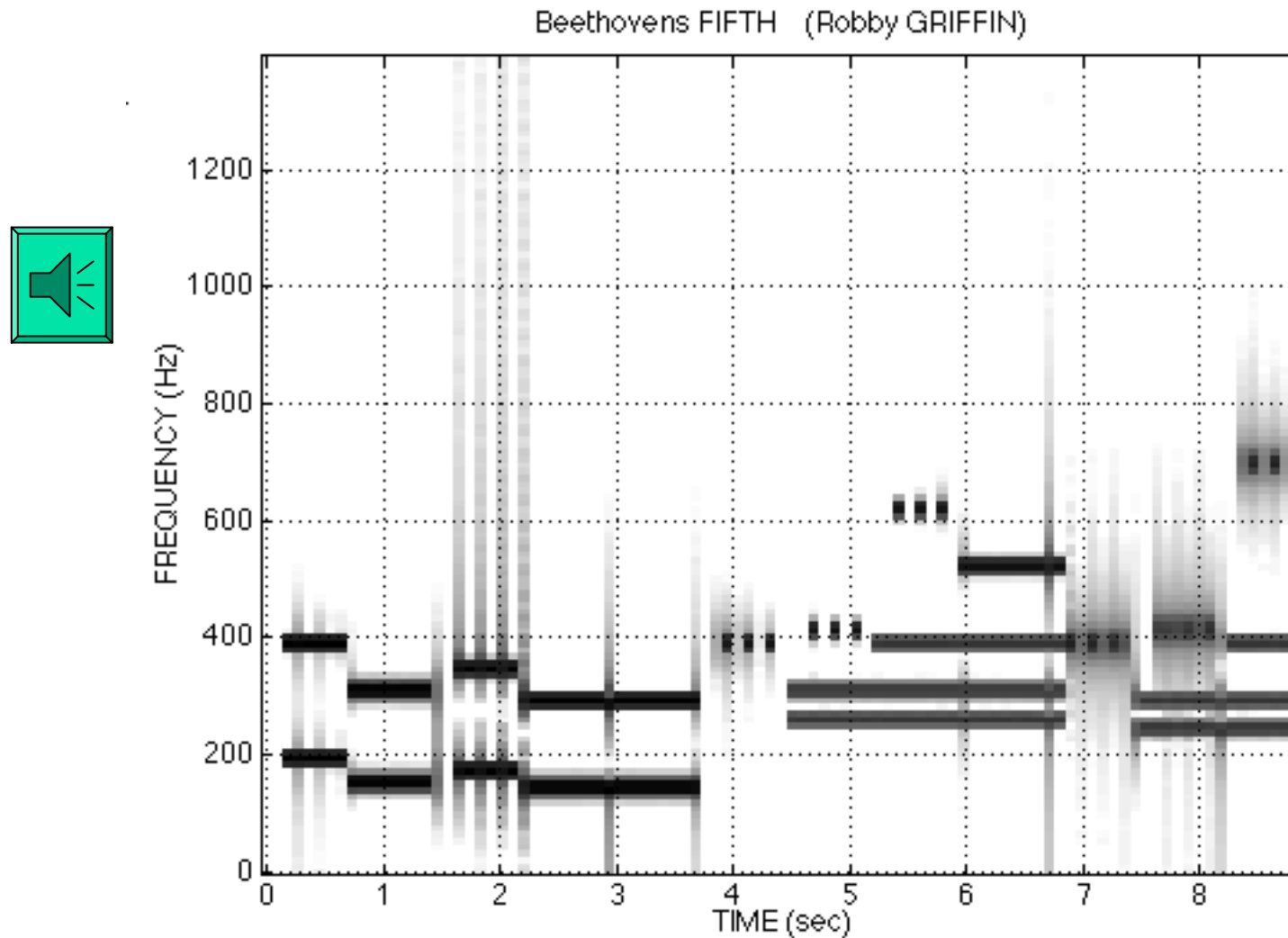


Spectrogram of C-Scale



From J.H. McClellan, R.W. Schafer, Signal Processing First, Prentice-Hall, 2003.

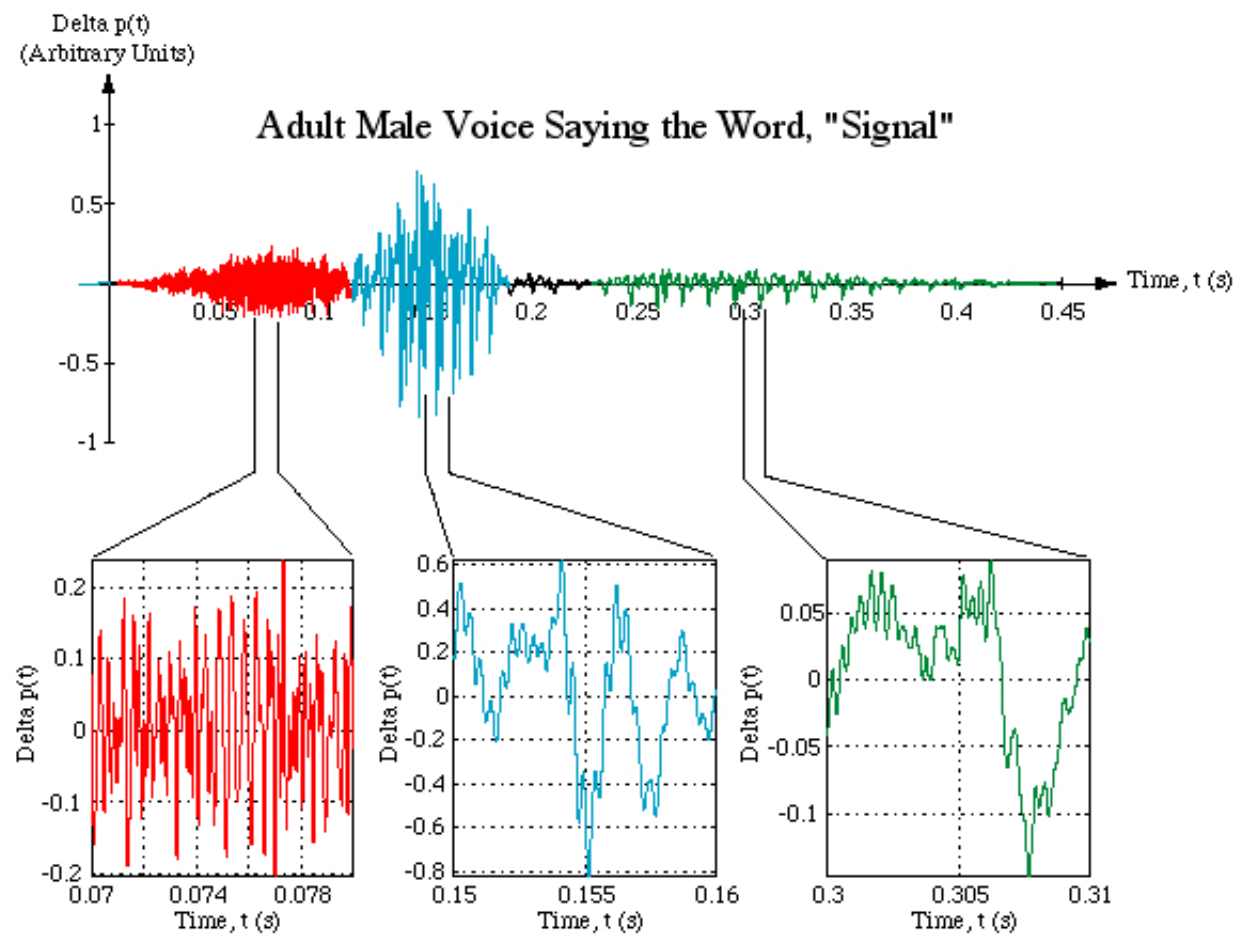
Spectrogram of LAB SONG



From J.H. McClellan, R.W. Schafer, Signal Processing First, Prentice-Hall, 2003.

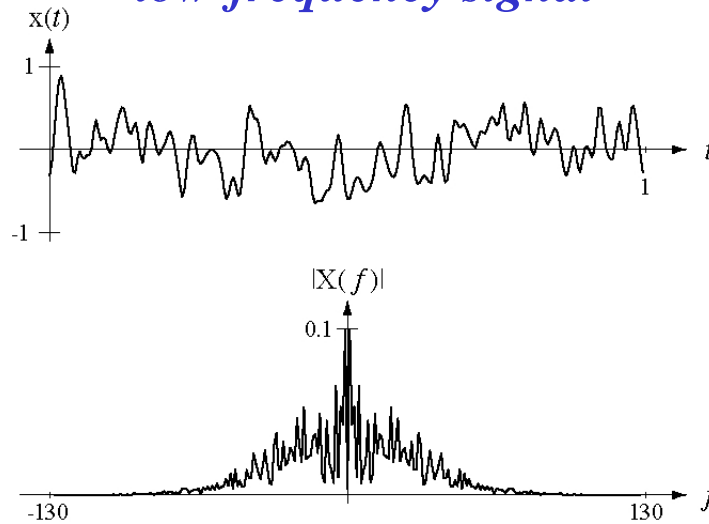
Recorded sound

“s” “i” “gn” “al”

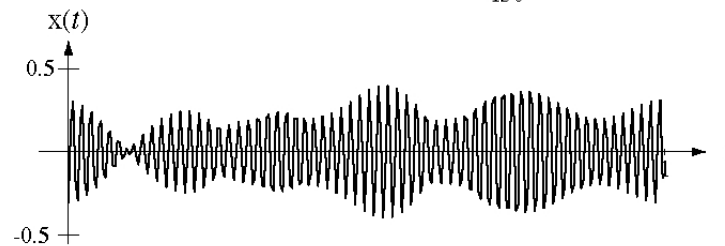
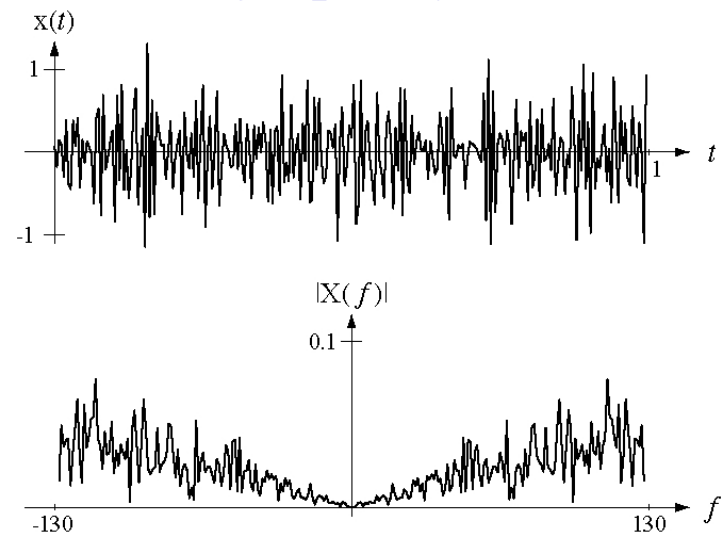


Fourier transform of signals

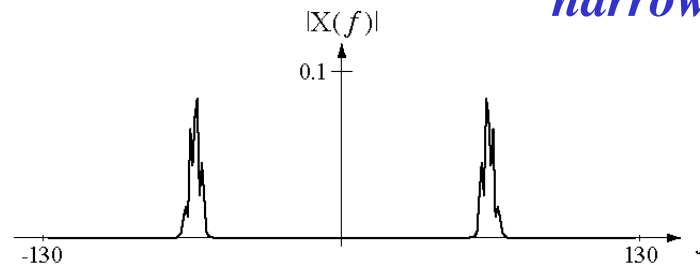
low-frequency signal



high-frequency signal

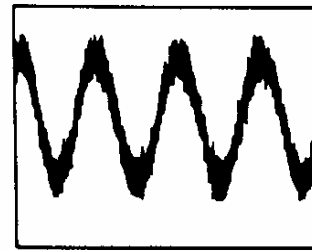


narrow-band signal

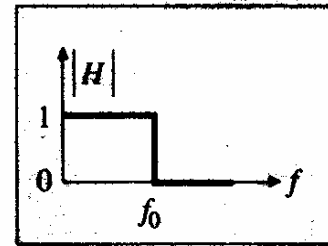


From .J.Roberts, Signals and Systems,
McGraw-Hill, 2003.

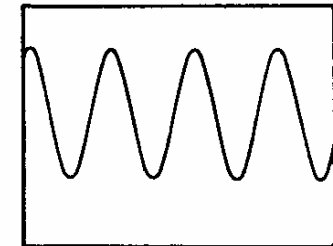
Ideal filters



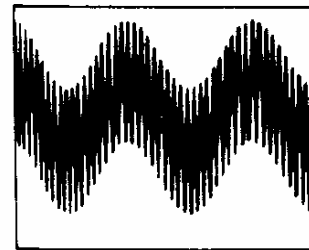
→ Time



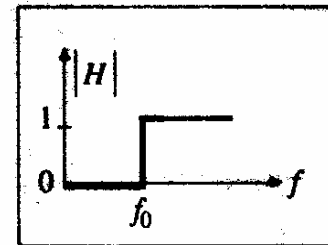
Lowpass filter



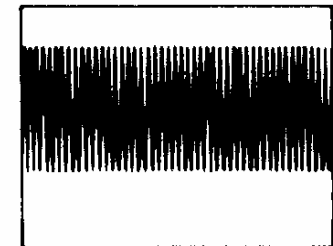
→ Time



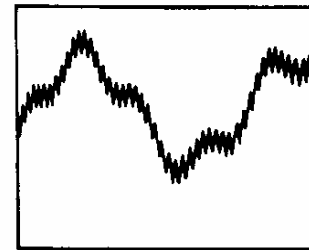
→ Time



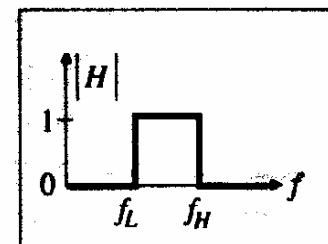
High-pass filter



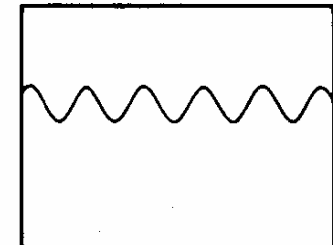
→ Time



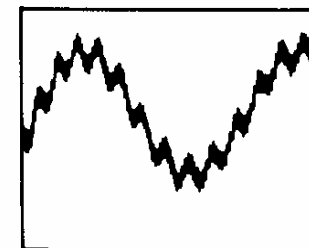
→ Time



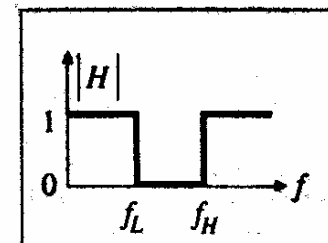
Band-pass filter



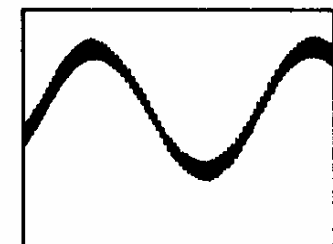
→ Time



→ Time



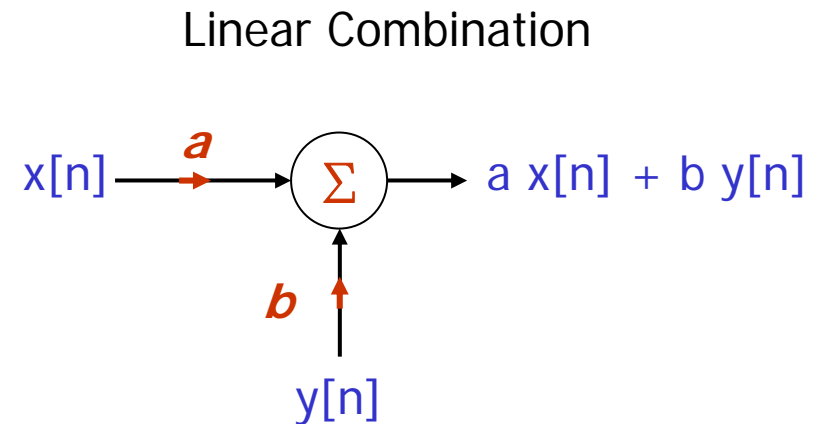
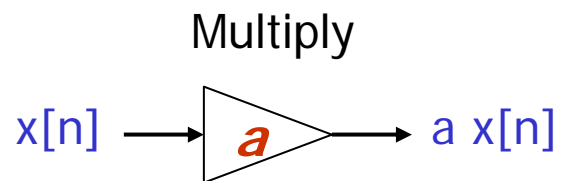
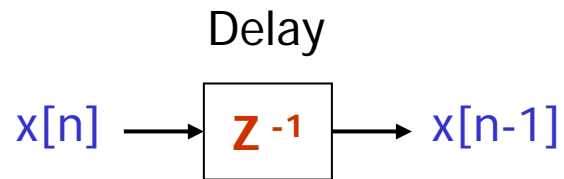
Band-reject filter



→ Time

S. Franco, "Design with Operational Amplifiers and Analog Integrated Circuits", Second Edition, 1998.

Z transform



	Digital signal	z transform	Analog signal
Input signal	$x[n]$	$X(z)$	$x(t)$
Delay one sample	$x[n-1]$	$Z^{-1} X(z)$	$x(t-T)$
Multiply	$a x[n]$	$a X(z)$	$a x(t)$
Linear combination	$a x[n] + b y[n]$	$a X(z) + b Y(z)$	$a x(t) + b y(t)$

Example 1: Perform the running average of last six digital sample

$$y[n] = \frac{x[n] + x[n-1] + x[n-2] + x[n-3] + x[n-4] + x[n-5]}{6}$$

$$Y(z) = \frac{X(z) + z^{-1}X(z) + z^{-2}X(z) + z^{-3}X(z) + z^{-4}X(z) + z^{-5}X(z)}{6}$$

Transfer function

$$H(z) = \frac{Y(z)}{X(z)} = \frac{1 + z^{-1} + z^{-2} + z^{-3} + z^{-4} + z^{-5}}{6}$$

```
fs=1000;  
t=0:1/fs:0.2;  
x = sin(2*pi*50*t) + 0.5*randn(size(t));
```

```
Xf=fft(x);  
resolution=fs/length(Xf);  
f=(0:length(Xf)-1)*resolution;  
Xf_magnitude = abs(Xf);  
subplot(3,2,2)  
index=1:length(Xf_magnitude)/2;  
plot(f(index),Xf_magnitude(index))
```

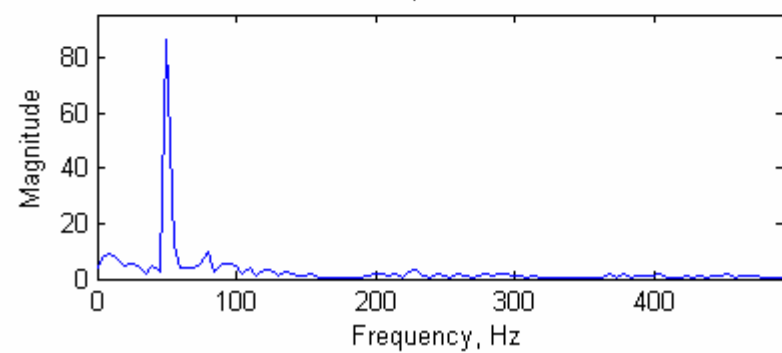
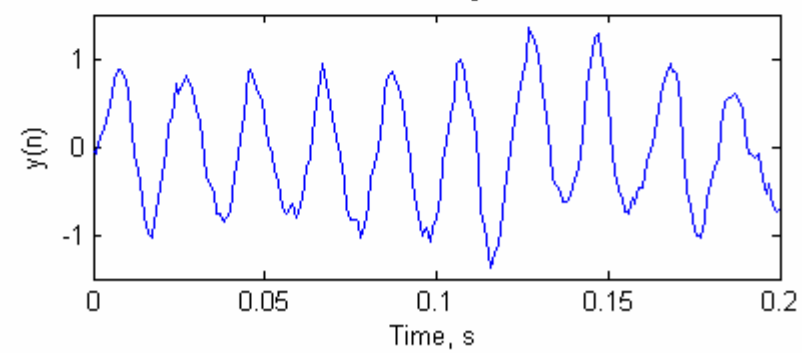
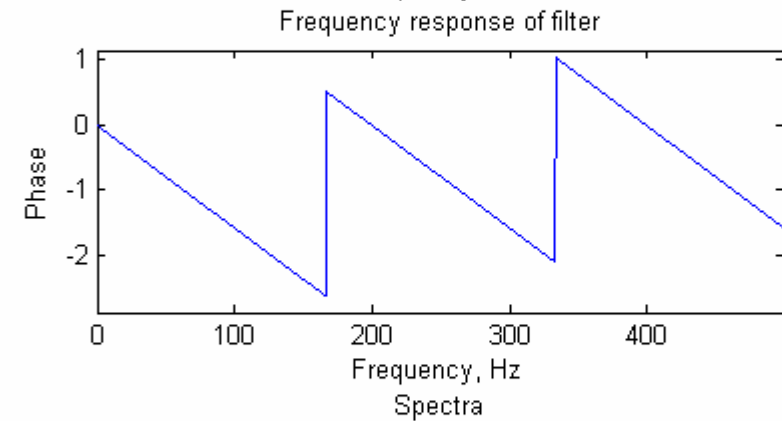
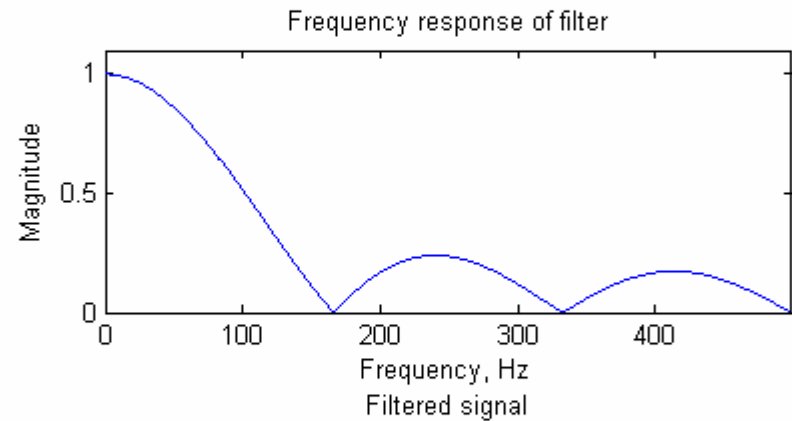
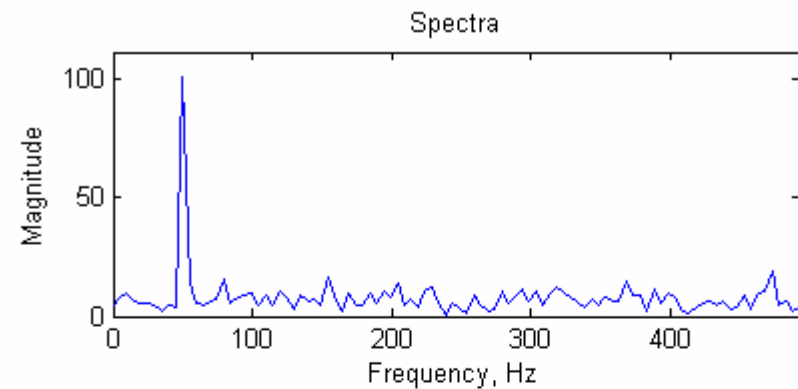
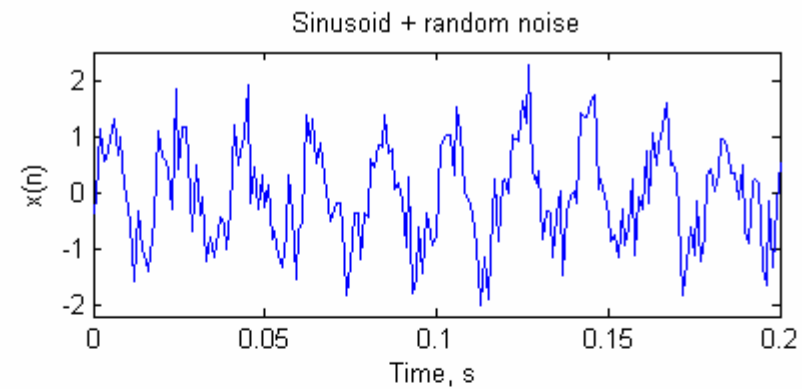
```
b=[1/6 1/6 1/6 1/6 1/6 1/6];  
a=[1];  
NFFT=1024;  
[h,f] = freqz(b,a,NFFT);
```

```
f=f/pi*fs/2;  
h_magnitude=abs(h);  
h_phase=phase(h);  
subplot(3,2,3)  
plot(f,h_magnitude);  
subplot(3,2,4)  
plot(f,h_phase);
```

```
y=filter(b,a,x);  
t=(0:length(y)-1)/fs;
```

```
Yf=fft(y);  
resolution=fs/length(Yf);  
f=(0:length(Yf)-1)*resolution;  
Yf_magnitude = abs(Yf);
```

```
subplot(3,2,6)  
index=1:length(Yf_magnitude)/2;  
plot(f(index),Yf_magnitude(index
```



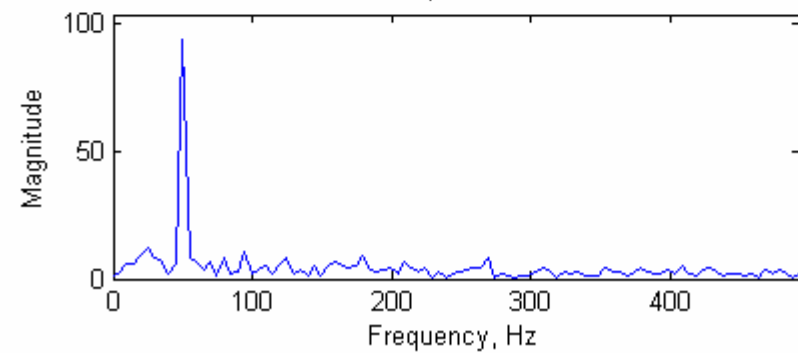
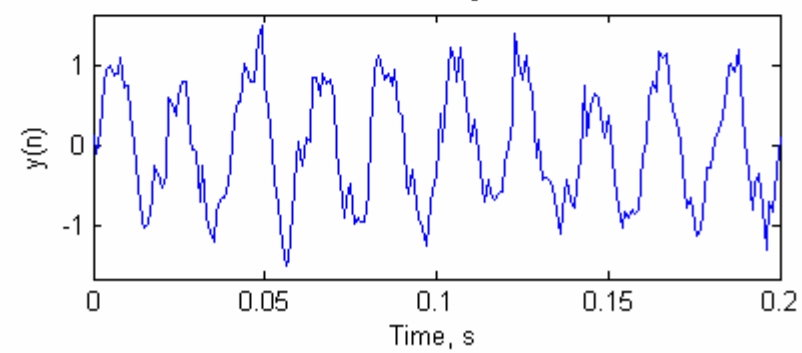
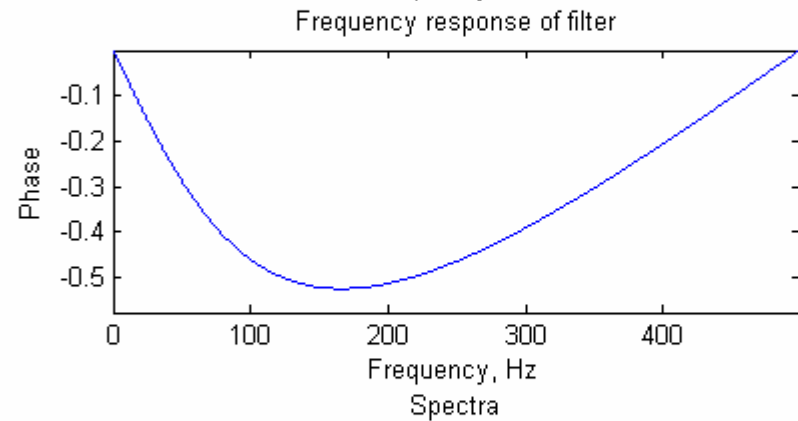
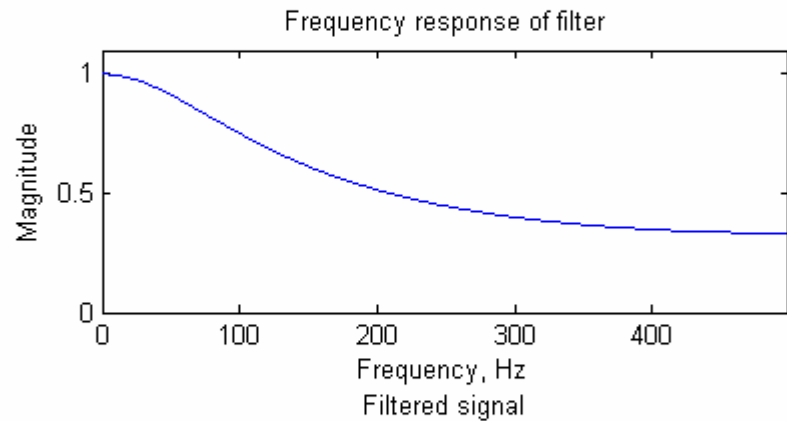
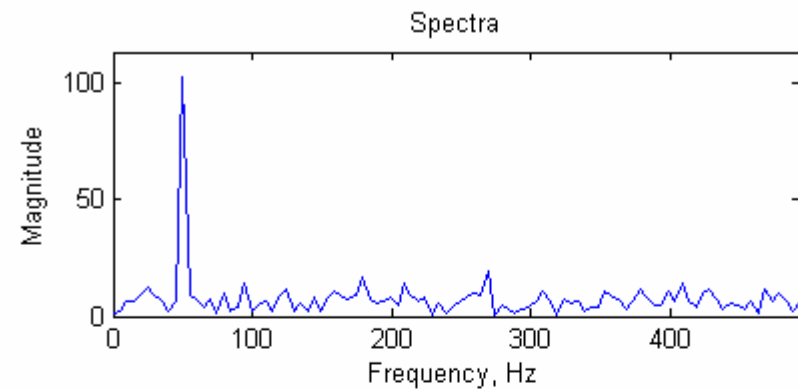
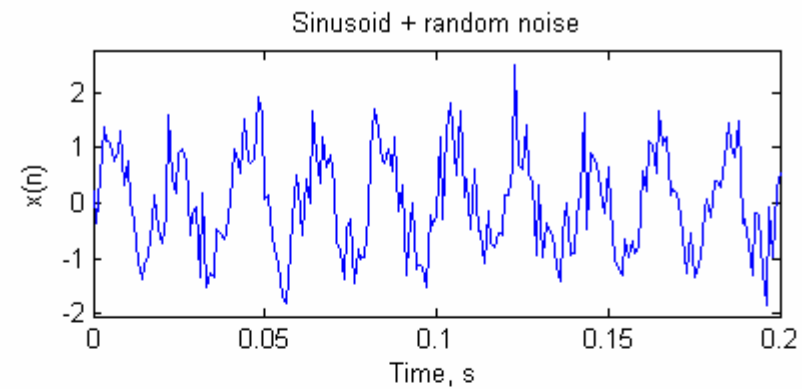
Example 2: Perform the average of current data and last filter output

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

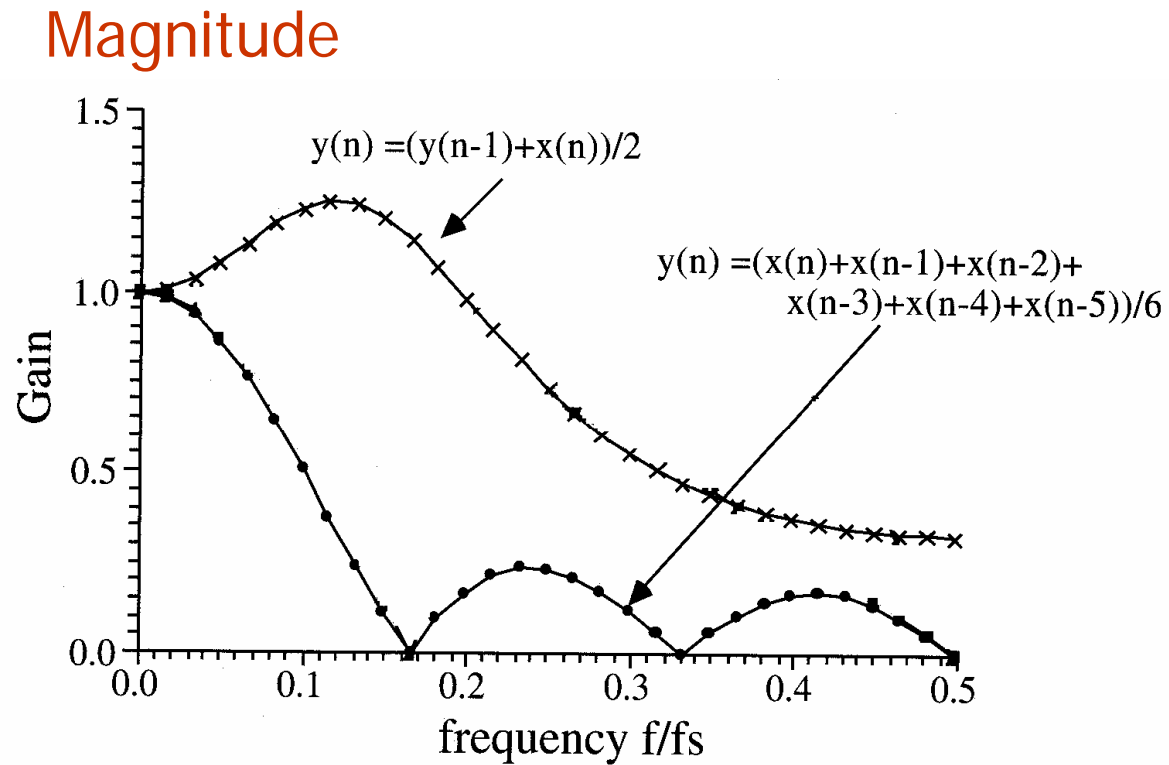
$$Y(z) = \frac{z^{-1}Y(z) + X(z)}{2}$$

Transfer function

$$H(z) = \frac{Y(z)}{X(z)} = \frac{1}{2 - z^{-1}}$$



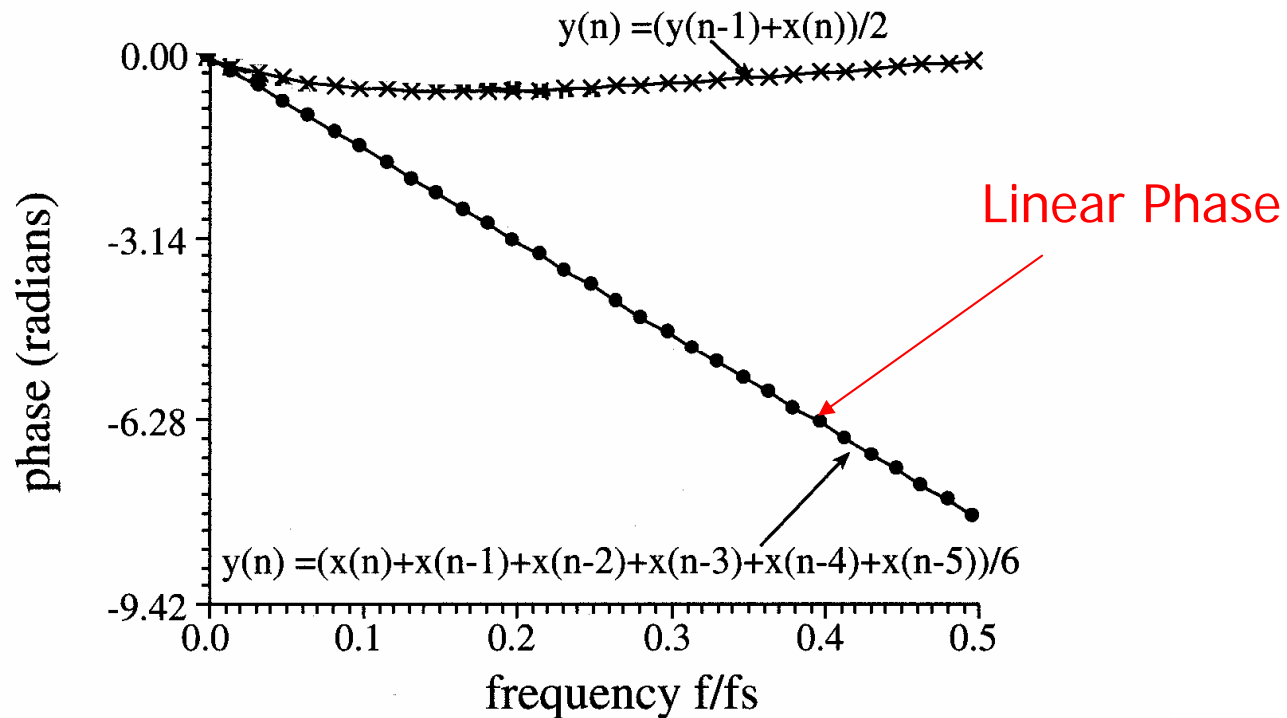
Frequency response of example 1 and 2



From Jonathan W. Valvano, Embedded Microcomputer Systems, real time interfacing, Brooks/Cole, 2000.

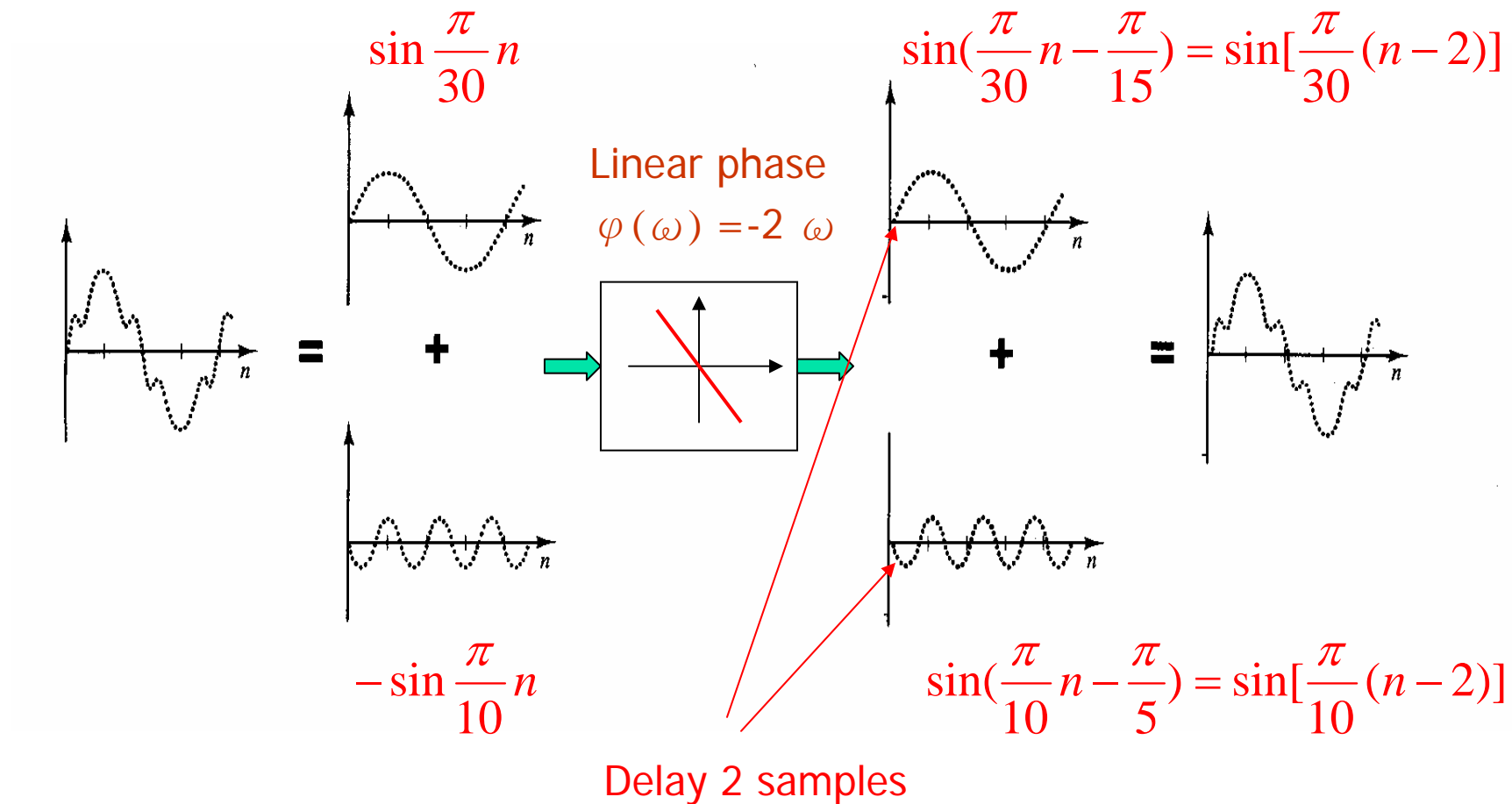
Frequency Response of Example 1 and 2

Phase



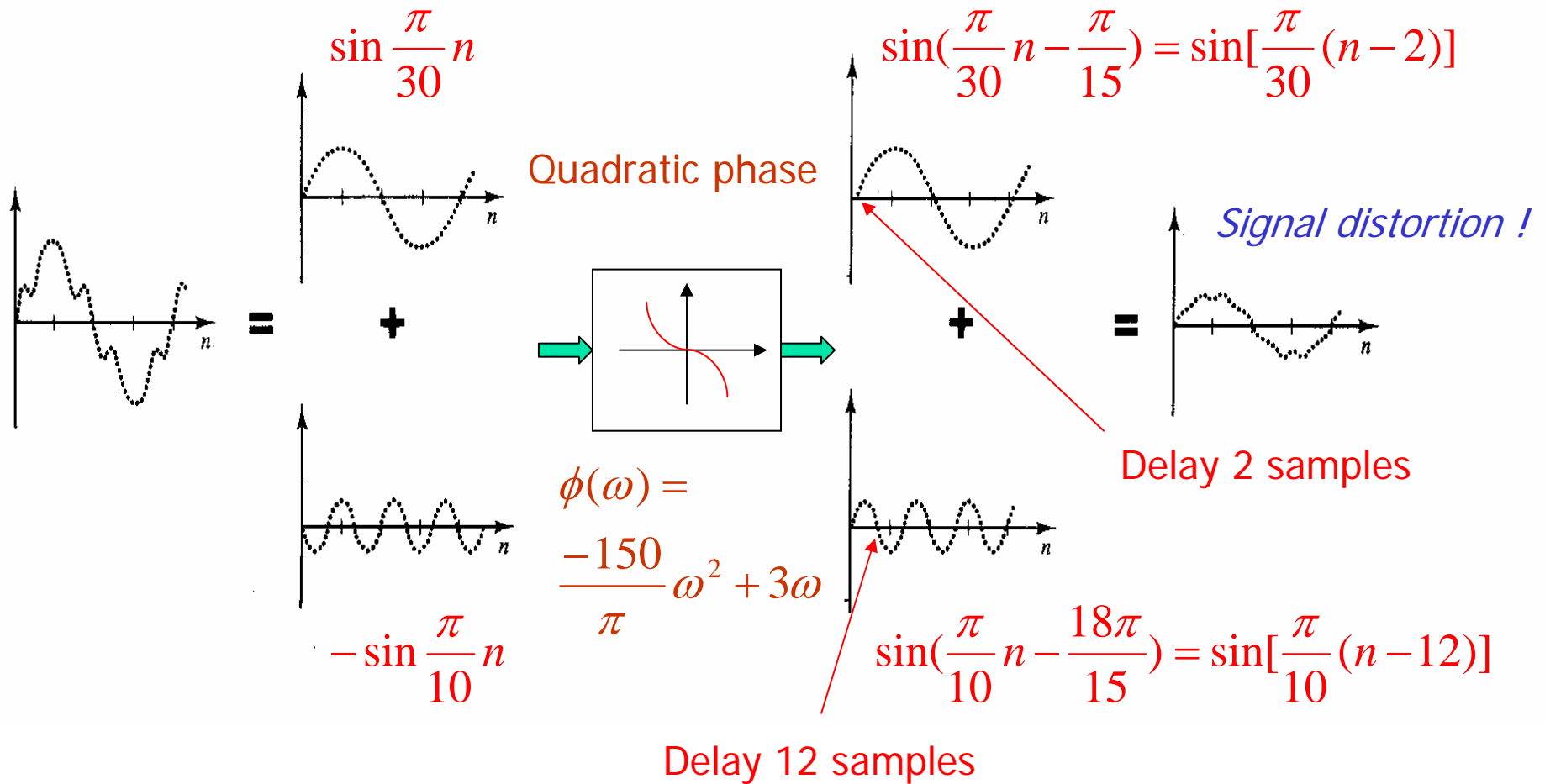
From Jonathan W. Valvano, Embedded Microcomputer Systems, real time interfacing, Brooks/Cole, 2000.

Linear Phase



Modified from L.Ludeman, Fundamentals of digital signal processing, Harper & Row, 1986.

Nonlinear Phase



Modified from L.Ludeman, Fundamentals of digital signal processing, Harper & Row, 1986.

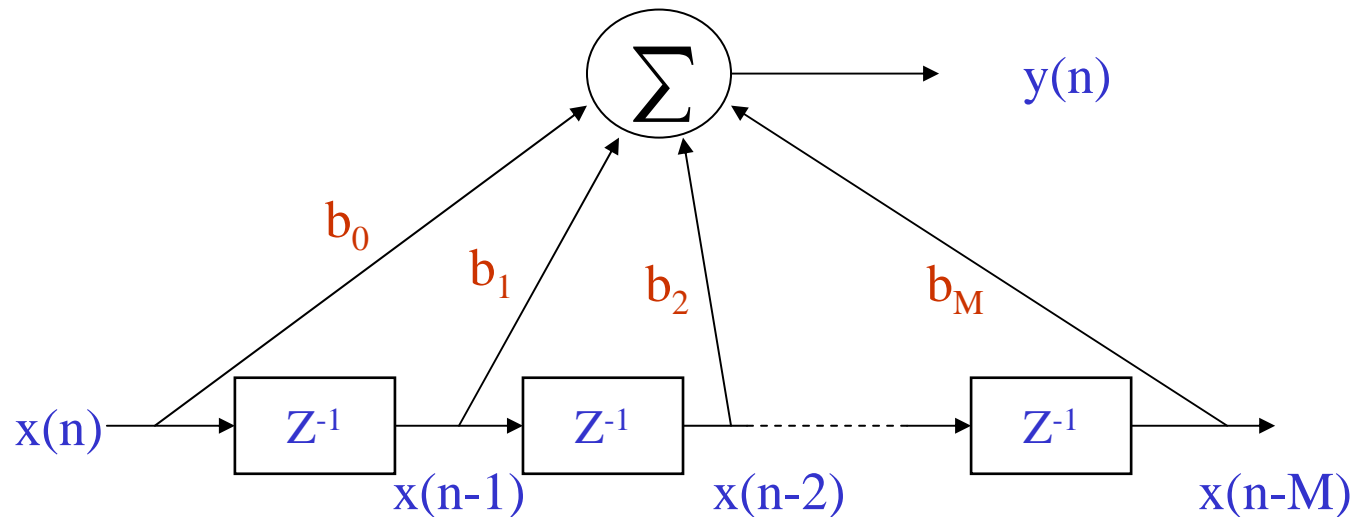
Group delay

$$\tau = -\frac{d\phi(\omega)}{d\omega}$$

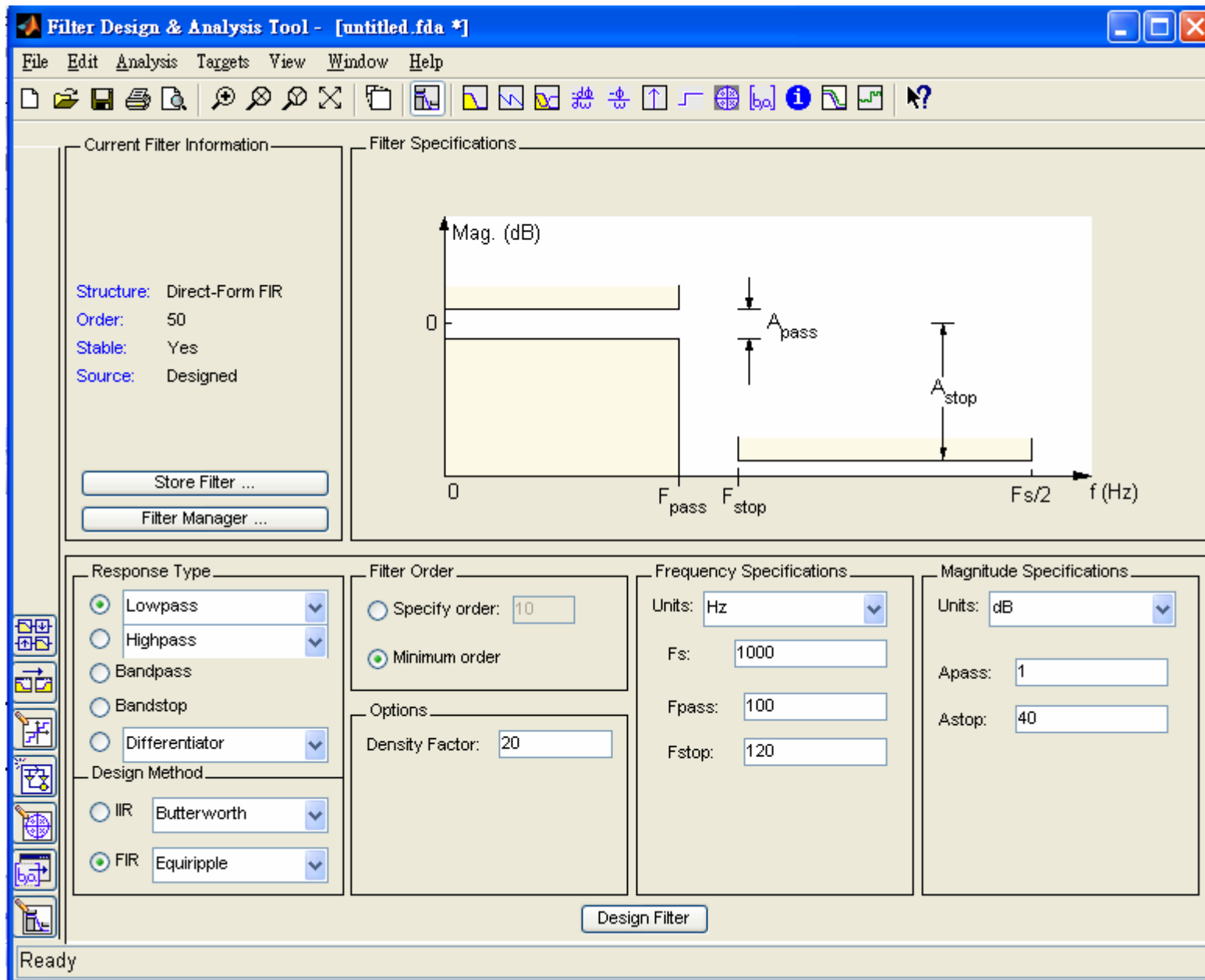
- Linear phase yield constant group delay

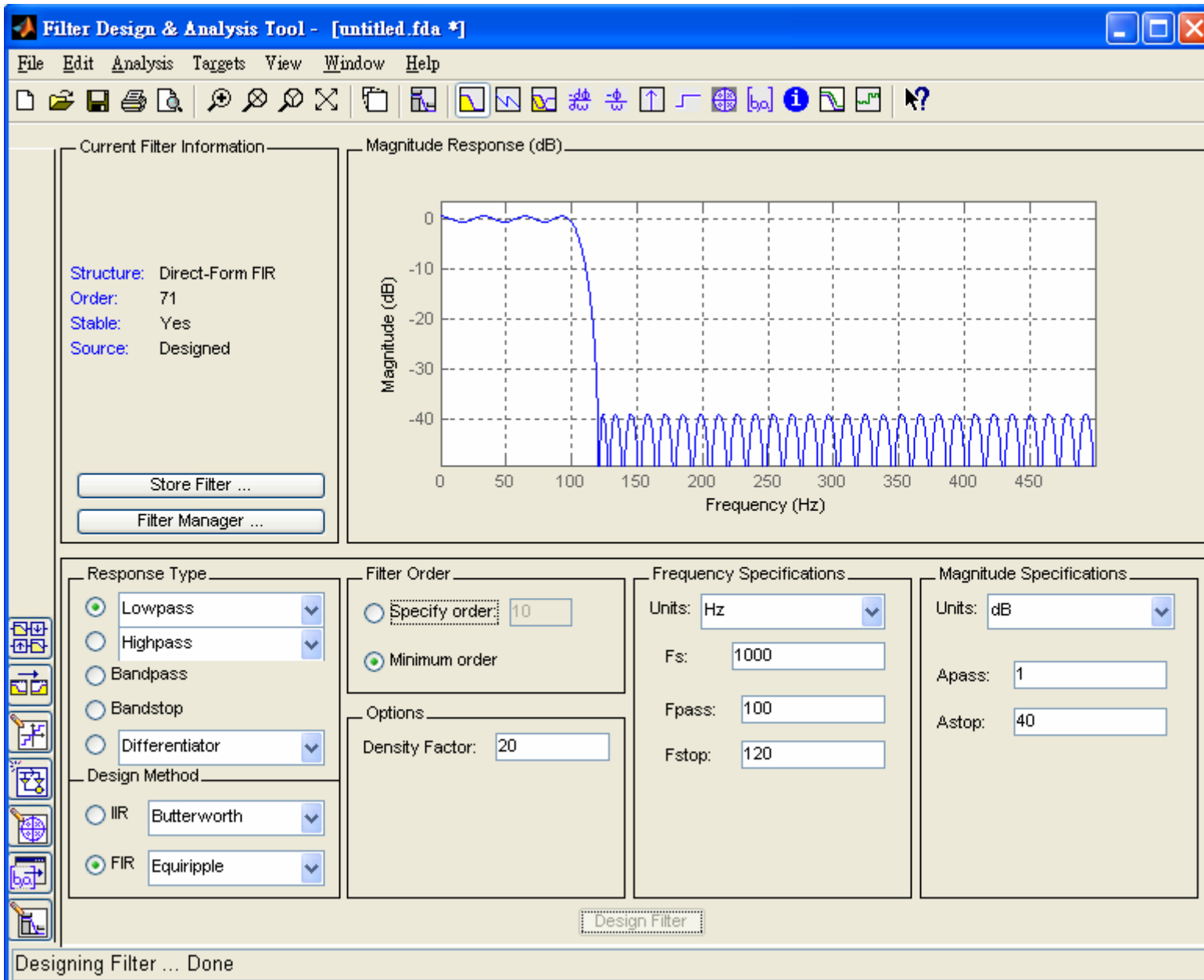
Finite impulse response (FIR) filter

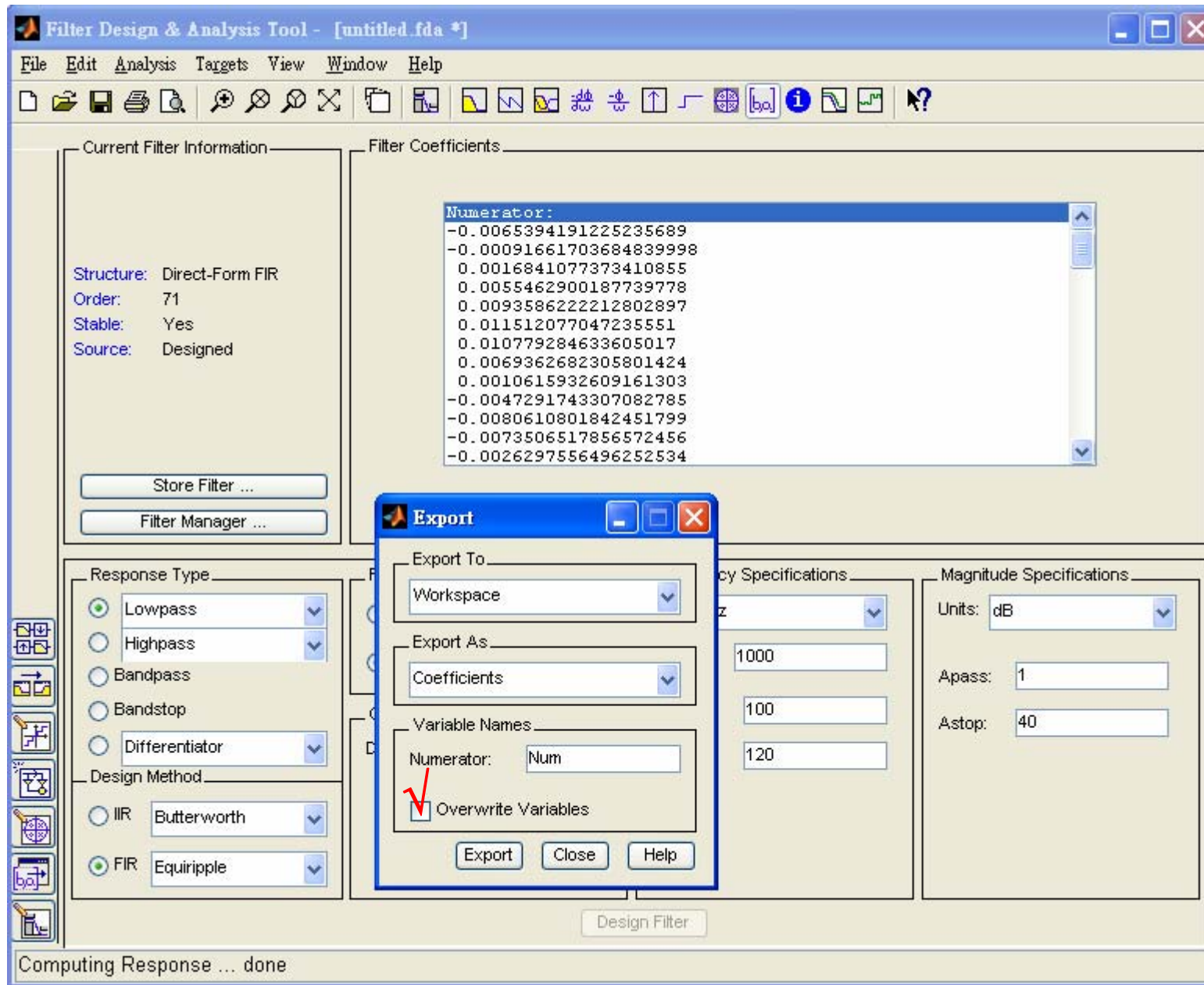
$$y(n) = \sum_{k=0}^M b_k x(n-k) \quad H(z) = \frac{Y(z)}{X(z)} = b_0 + b_1 z^{-1} + b_2 z^{-2} \cdots + b_M z^{-M}$$

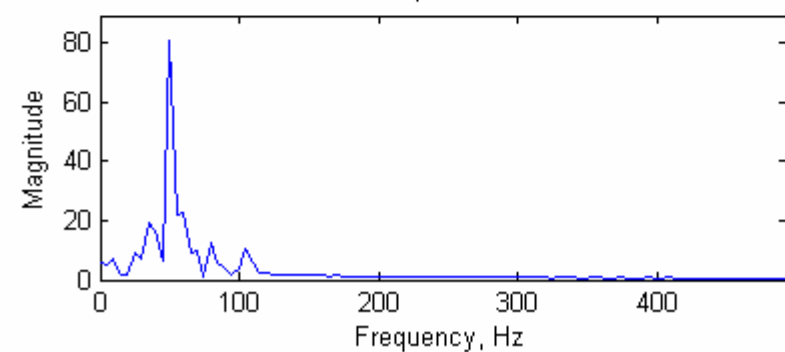
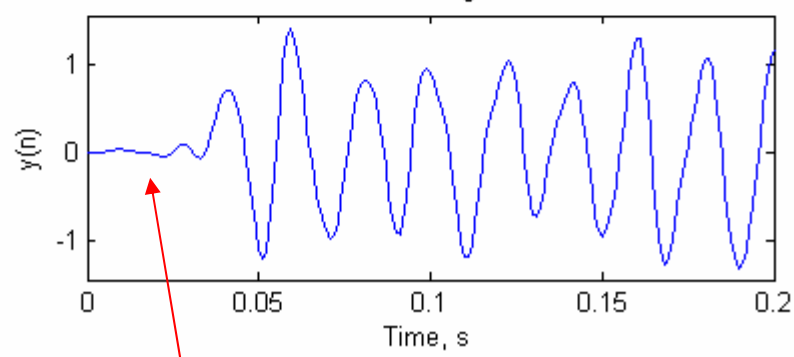
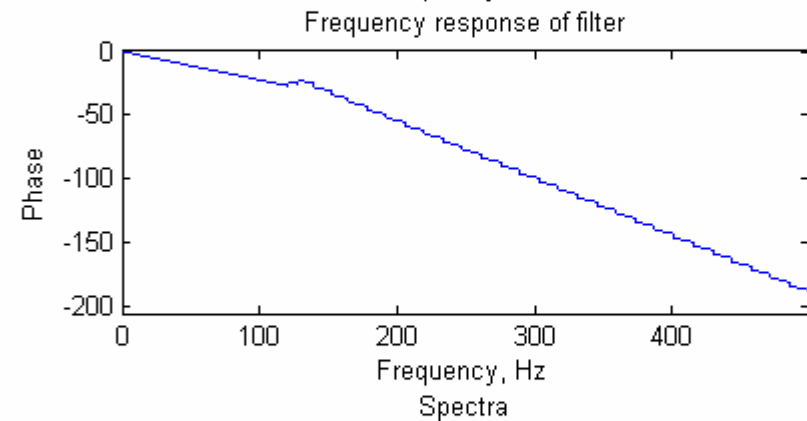
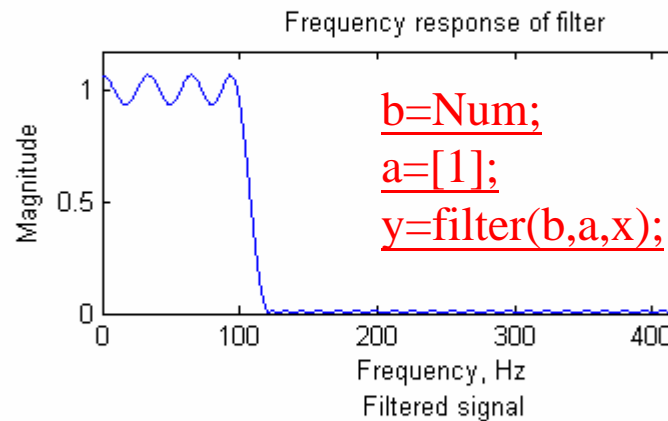
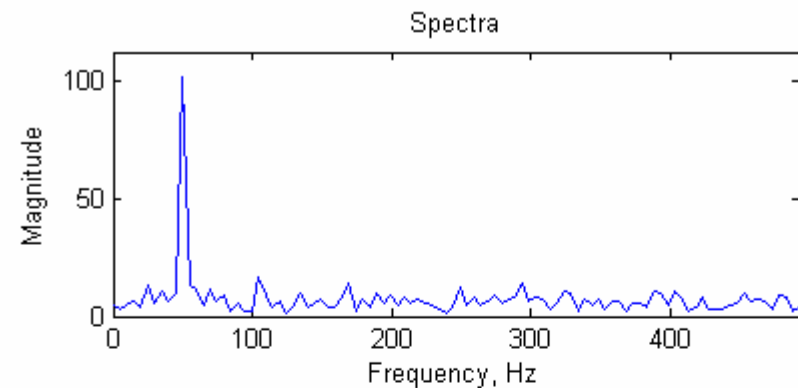
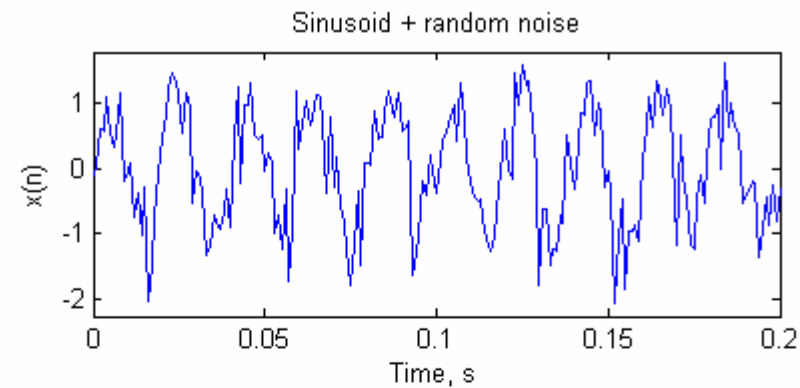


- FIR possesses linear-phase property if filter coefficients are symmetry or anti-symmetry around the center





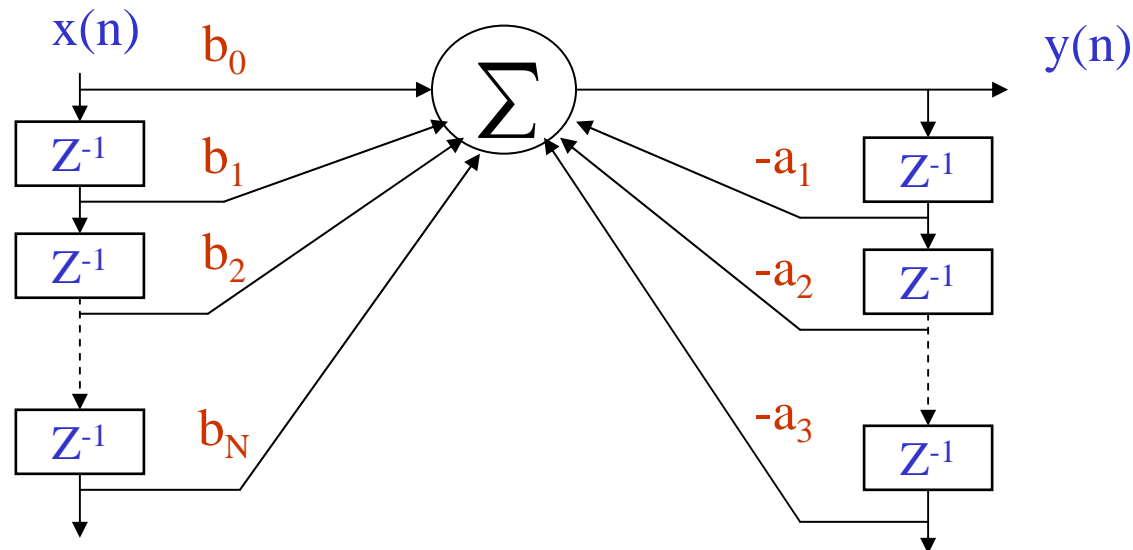


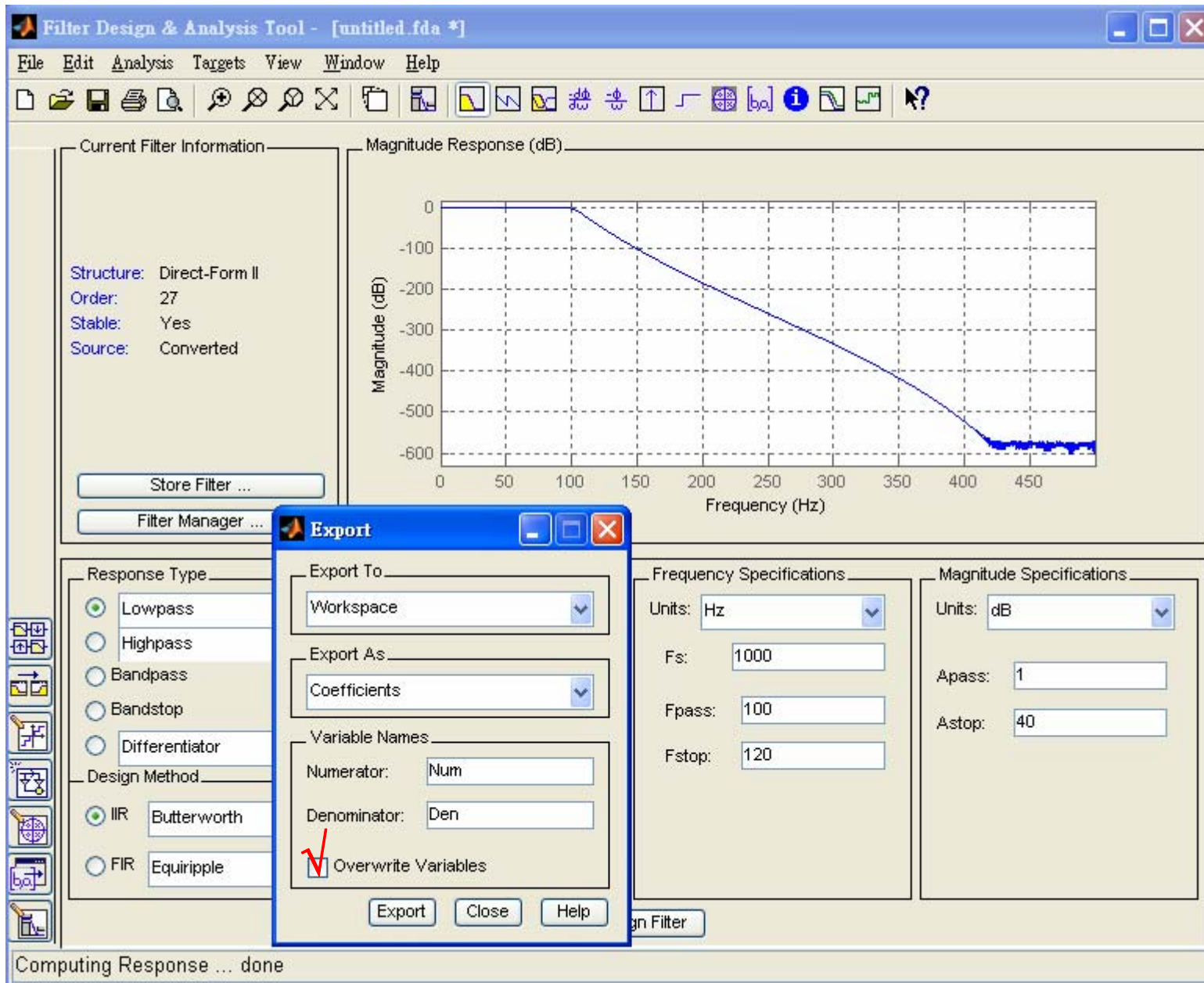


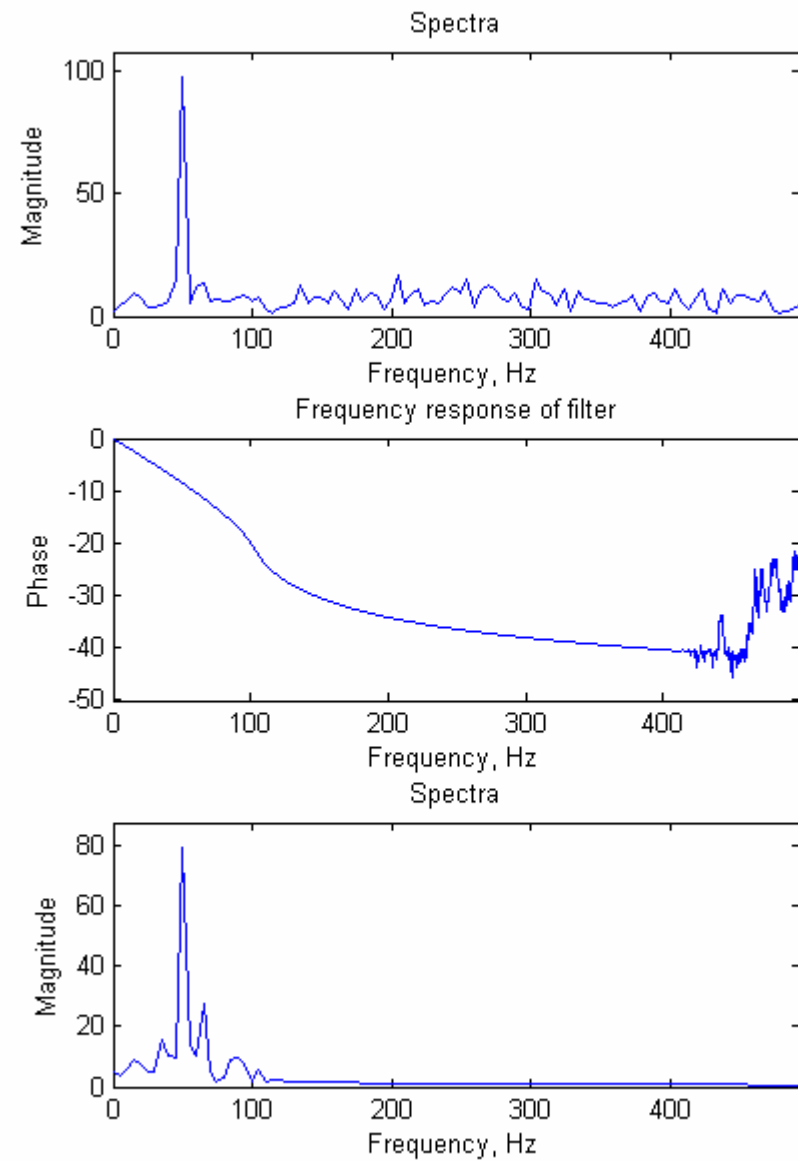
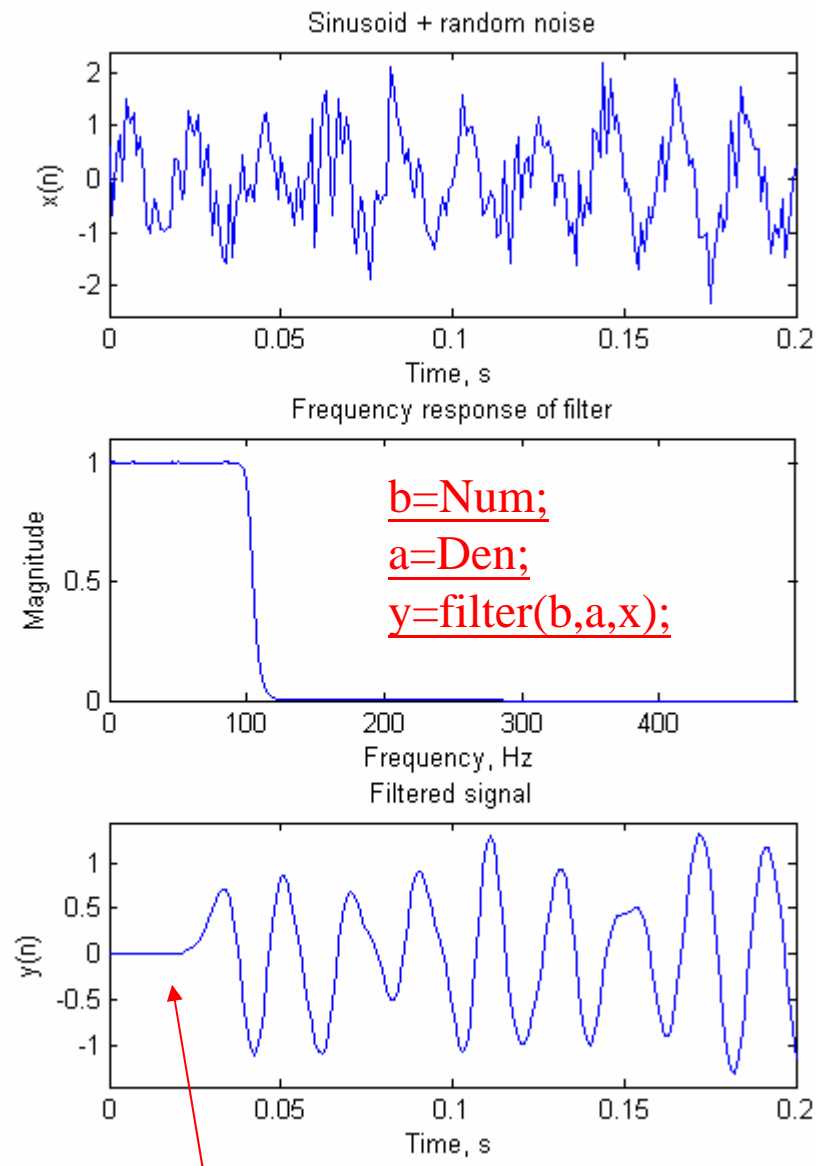
Transient. Group delay = filter length / 2

Infinite impulse response (IIR) filter

$$\sum_{p=0}^N a_p y(n-p) = \sum_{q=0}^M b_q x(n-q) \quad H(z) = \frac{Y(z)}{X(z)} = \frac{\sum_{q=0}^M b_q z^{-q}}{\sum_{p=0}^N a_p z^{-p}}$$

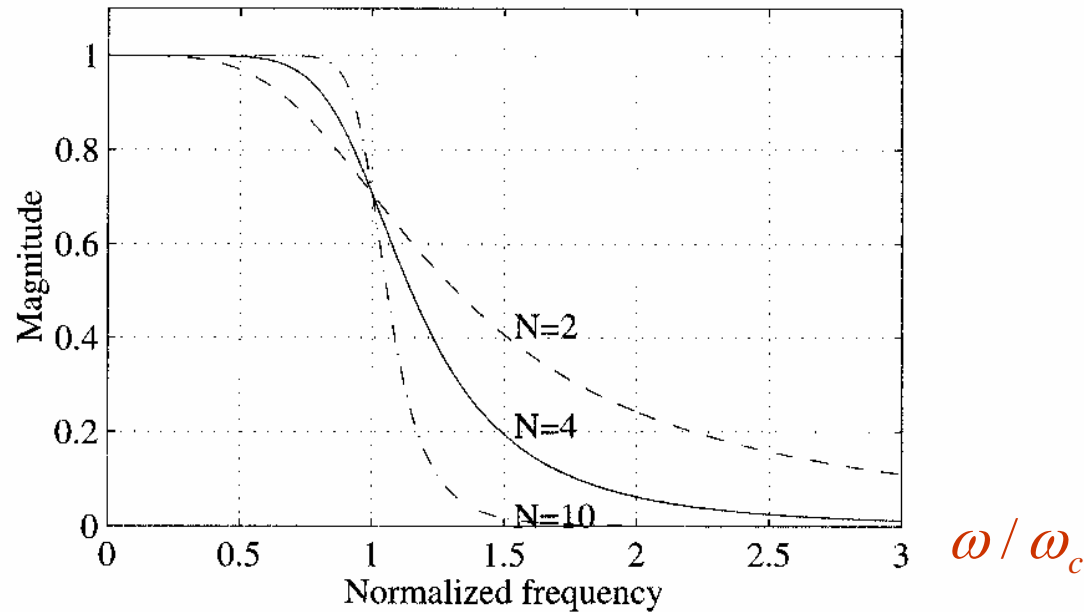






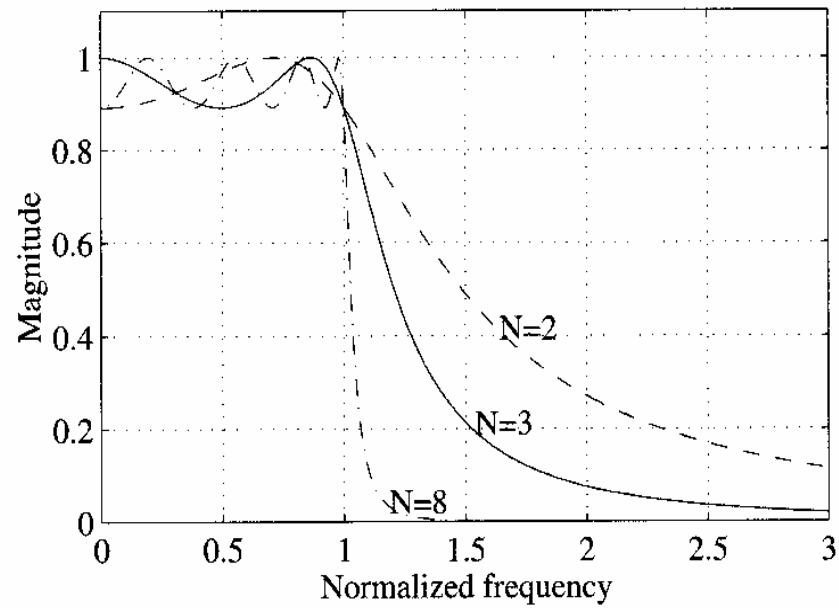
Transient. Group delay is not constant, changed as frequency

Butterworth approximation

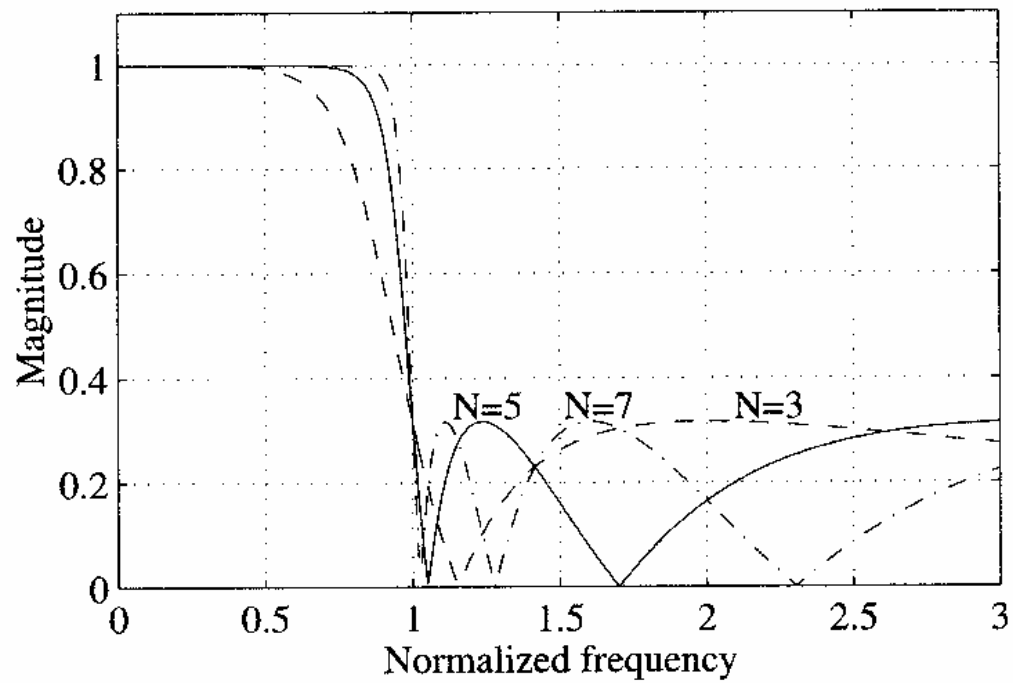


$$|H(j\omega)|^2 = \frac{1}{1 + (\omega / \omega_c)^{2N}}$$

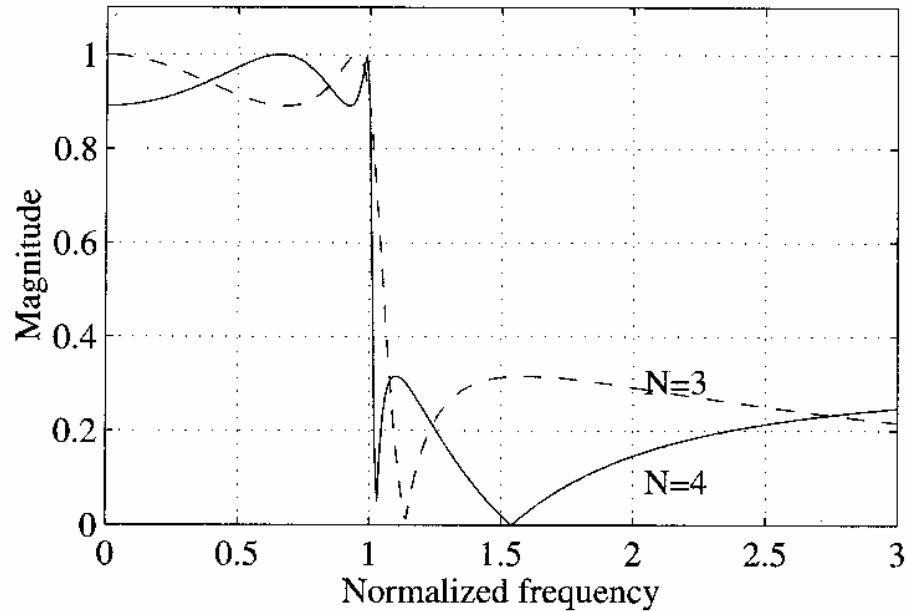
Type I Chebyshev approximation



Type II Chebyshev approximation



Elliptic approximation



- Equiripple in both passband and stopband
- Meet filter requirements with the lowest order

Reference

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- J. Semmlow, Circuits, Signals, and Systems for Bioengineers: A MATLAB-Based Introduction, Academic Press, 2005.
- M.J. Roberts, Signals and Systems: Analysis of Signals Through Linear Systems, McGraw-Hill, 2003.