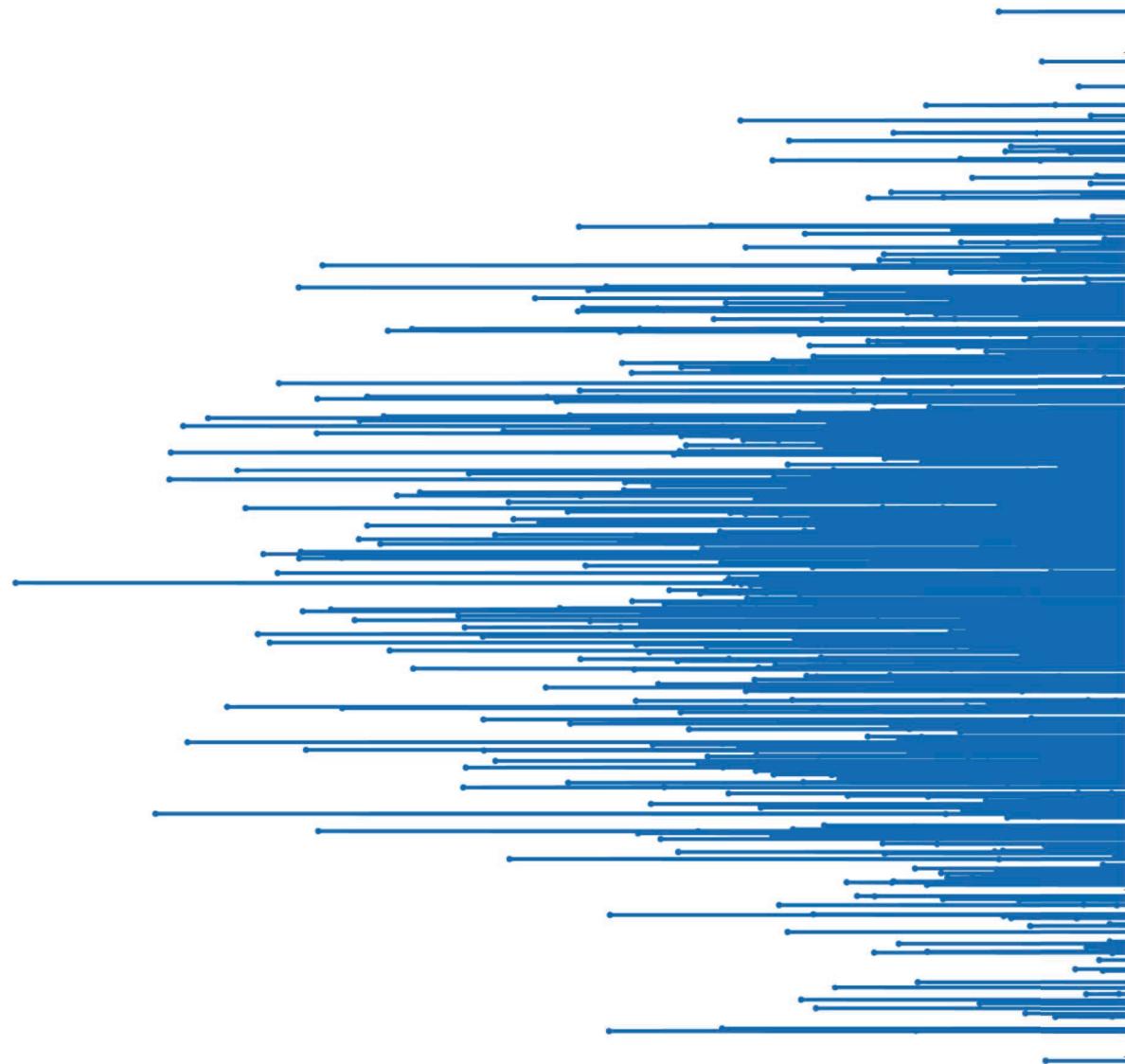


# PHYS 2010 (W20)

## Classical Mechanics



2020.03.05

Relevant reading:

Knudsen & Hjorth: X

Christopher Bergevin

York University, Dept. of Physics & Astronomy

Office: Petrie 240 Lab: Farq 103

cberge@yorku.ca

Ref.s:

Knudsen & Hjorth (2000), Fowles & Cassidy (2005)

## 239. Tangled Stars

A



B



C



D



Three of the four tangled stars are identical.  
Which one is different?

A

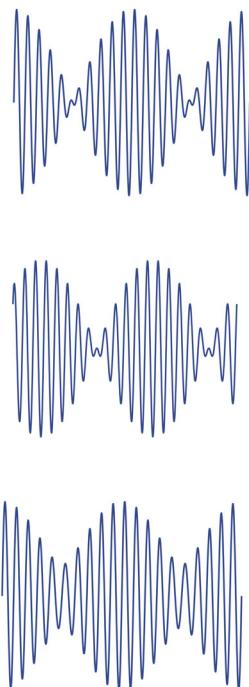
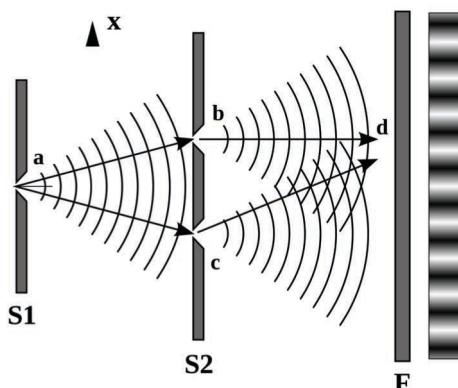
B

C

D

## Looking Ahead...

### Superposition



DDHO is a linear system

$$m\ddot{x} + c\dot{x} + kx = \sum_n (m\ddot{x}_n + c\dot{x}_n + kx_n) = \sum_n F_n(t) = F_{ext}$$

Periodic forcing & superposition



$$F_{ext} = \sum_n F_n(t)$$



Fourier analysis

$$f(t) = \frac{1}{2}a_0 + \sum_{n=1}^{\infty} [a_n \cos(n\omega t) + b_n \sin(n\omega t)]$$

$$a_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \cos(n\omega t) dt \quad n = 0, 1, 2, \dots$$

$$b_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \sin(n\omega t) dt \quad n = 1, 2, \dots$$



## Impulse response

- Intuitively defined in two different (but equivalent) ways:

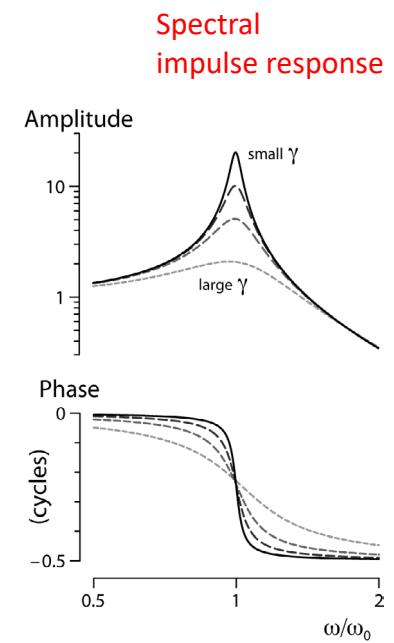
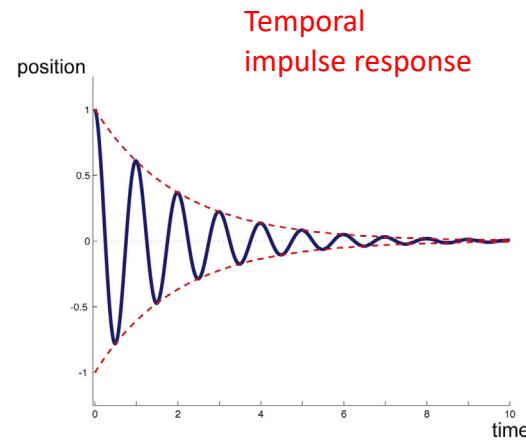
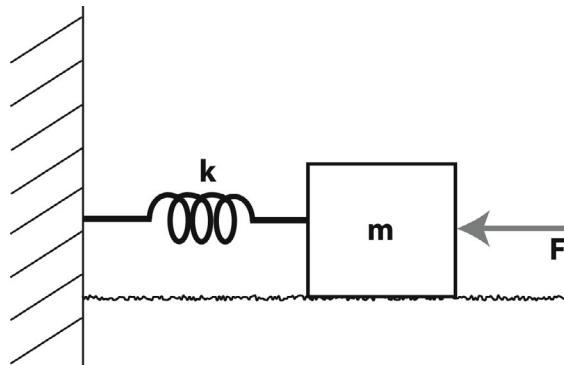
1. Time response of 'system' when subjected to an impulse

(e.g., striking a bell w/ a hammer)

2. Fourier transform of resulting response

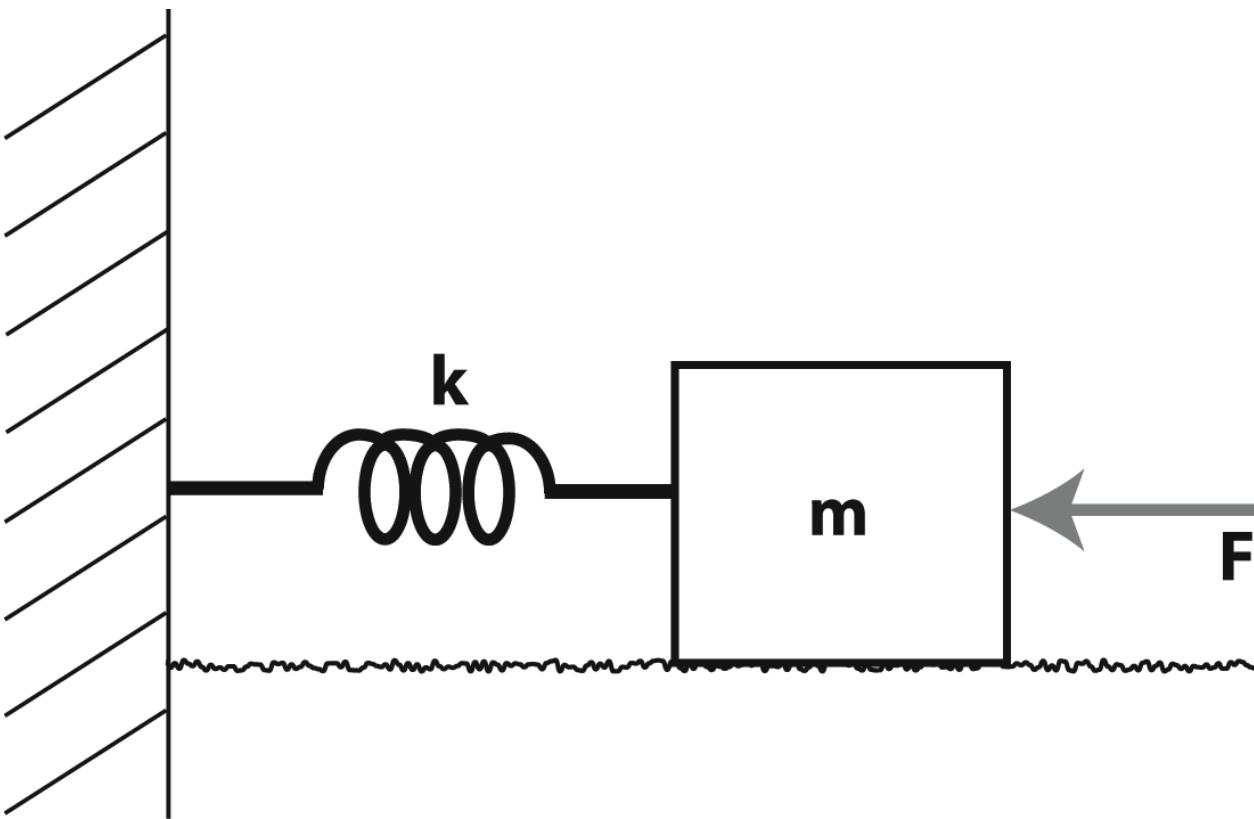
(e.g., spectrum of bell ringing)

ex. Harmonic oscillator



(Important) Note: The Fourier transform of the impulse response is called the *transfer function*

## Superposition & Linearity



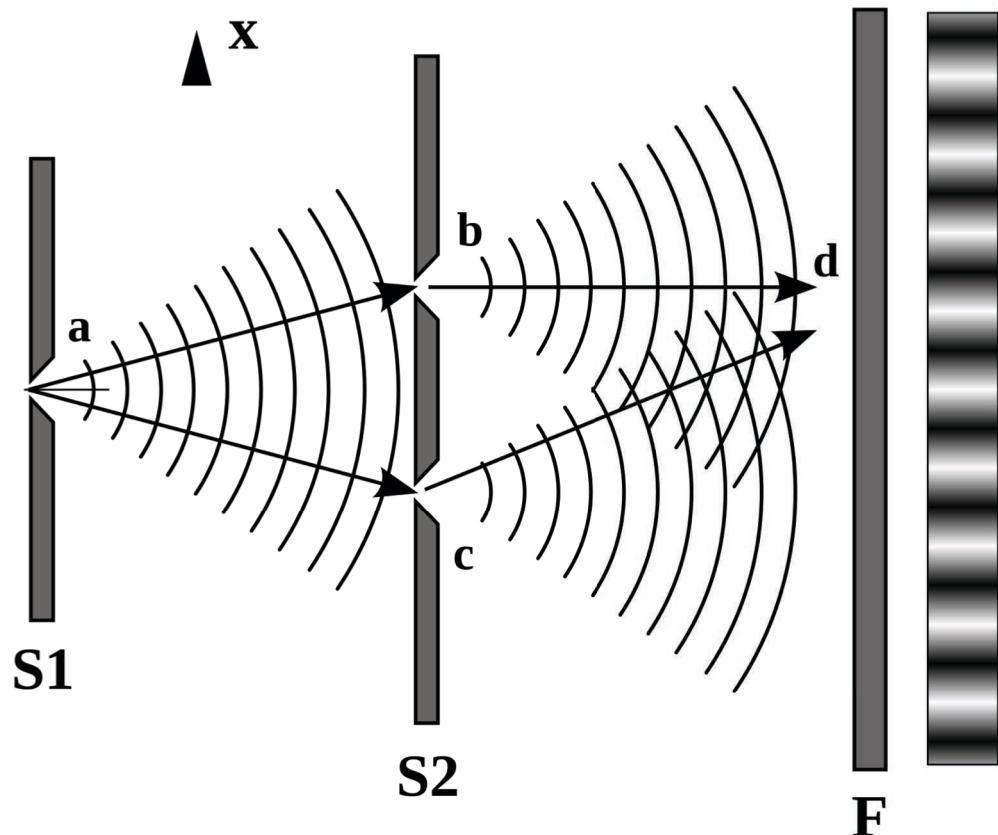
→ When dealing with linear oscillators (or linear systems in general), superposition takes a domineering position in how we approach analysis and modeling

# Superposition



Thomas Young (1773-1829)

"Double-slit experiment"



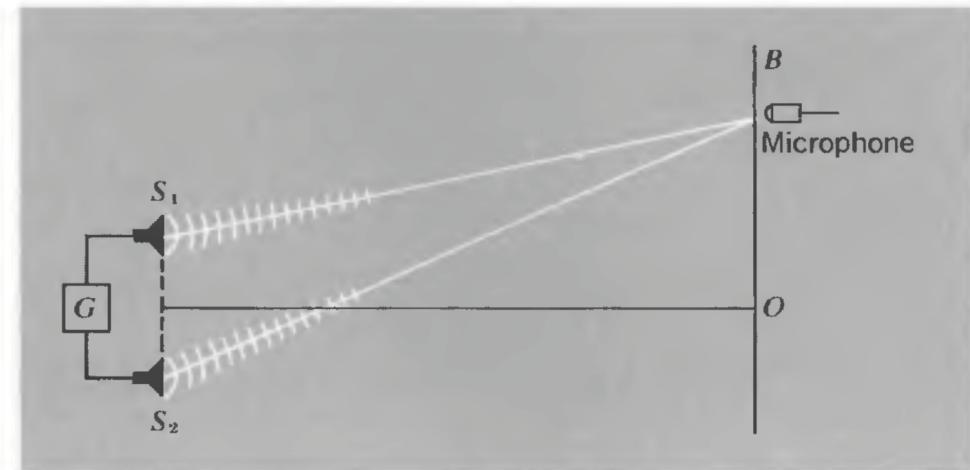
→ Wave theory of light  
(via *phase interference*)

## Superposition

Each source:

$$x_1 = A_1 \cos(\omega t + \alpha_1)$$

$$x_2 = A_2 \cos(\omega t + \alpha_2)$$



Their sum (at the mic)

$$x = x_1 + x_2 = A_1 \cos(\omega t + \alpha_1) + A_2 \cos(\omega t + \alpha_2)$$

$$= A \cos(\omega t + \alpha)$$

A tad messy to solve for the constants there....

If instead we used complex exponentials:

$$z_1 = A_1 e^{j(\omega t + \alpha_1)}$$

$$z_2 = A_2 e^{j(\omega t + \alpha_2)}$$

Then:

$$z = z_1 + z_2 = A_1 e^{j(\omega t + \alpha_1)} + A_2 e^{j(\omega t + \alpha_2)}$$

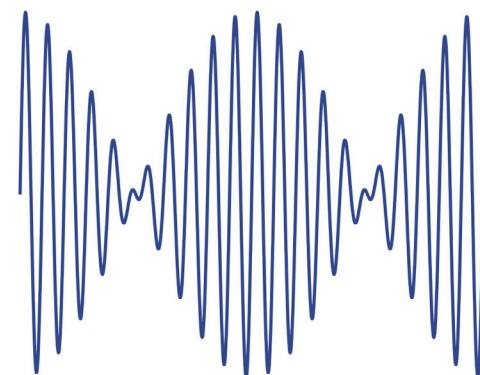
$$= e^{j(\omega t + \alpha_1)} [A_1 + A_2 e^{j(\alpha_2 - \alpha_1)}]$$

→ This latter equation, while seemingly intimidating, tells us a lot!

## Superposition: "Beats" (adding two different frequencies)

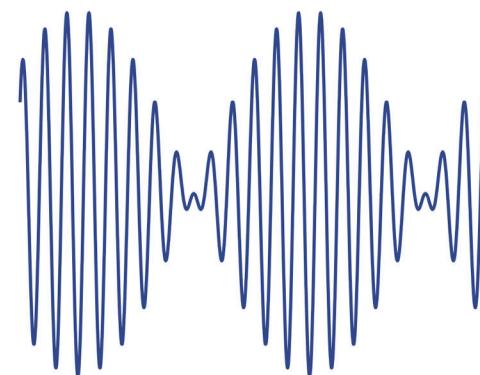
$$x(t) = A_1 \sin(2\pi f_1 t + \phi_1) + A_2 \sin(2\pi f_2 t + \phi_2)$$

$$\begin{aligned}f_1 &= 1, f_2 = 1.1 \\A_1 &= 1, A_2 = 1 \\\phi_1 &= 0, \phi_2 = 0\end{aligned}$$



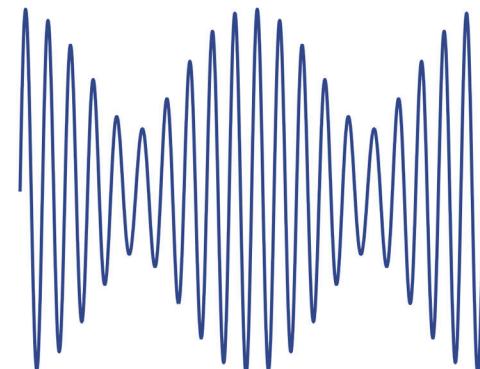
→ Changing (relative) phase affects summation

$$\begin{aligned}f_1 &= 1, f_2 = 1.1 \\A_1 &= 1, A_2 = 1 \\\phi_1 &= \pi/2, \phi_2 = 0\end{aligned}$$



→ Changing (relative) amplitudes affects summation

$$\begin{aligned}f_1 &= 1, f_2 = 1.1 \\A_1 &= 2, A_2 = 1 \\\phi_1 &= 0, \phi_2 = 0\end{aligned}$$

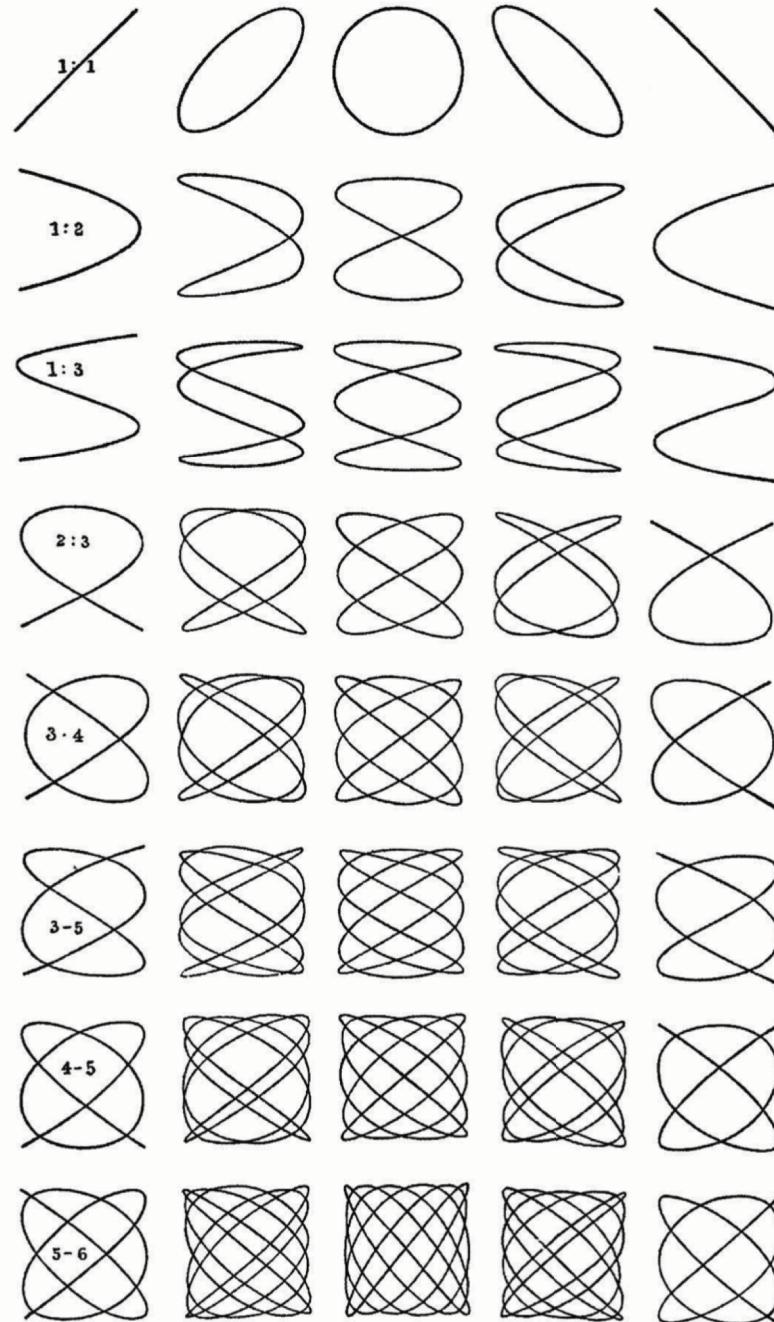


## Recall: Lissajou figures

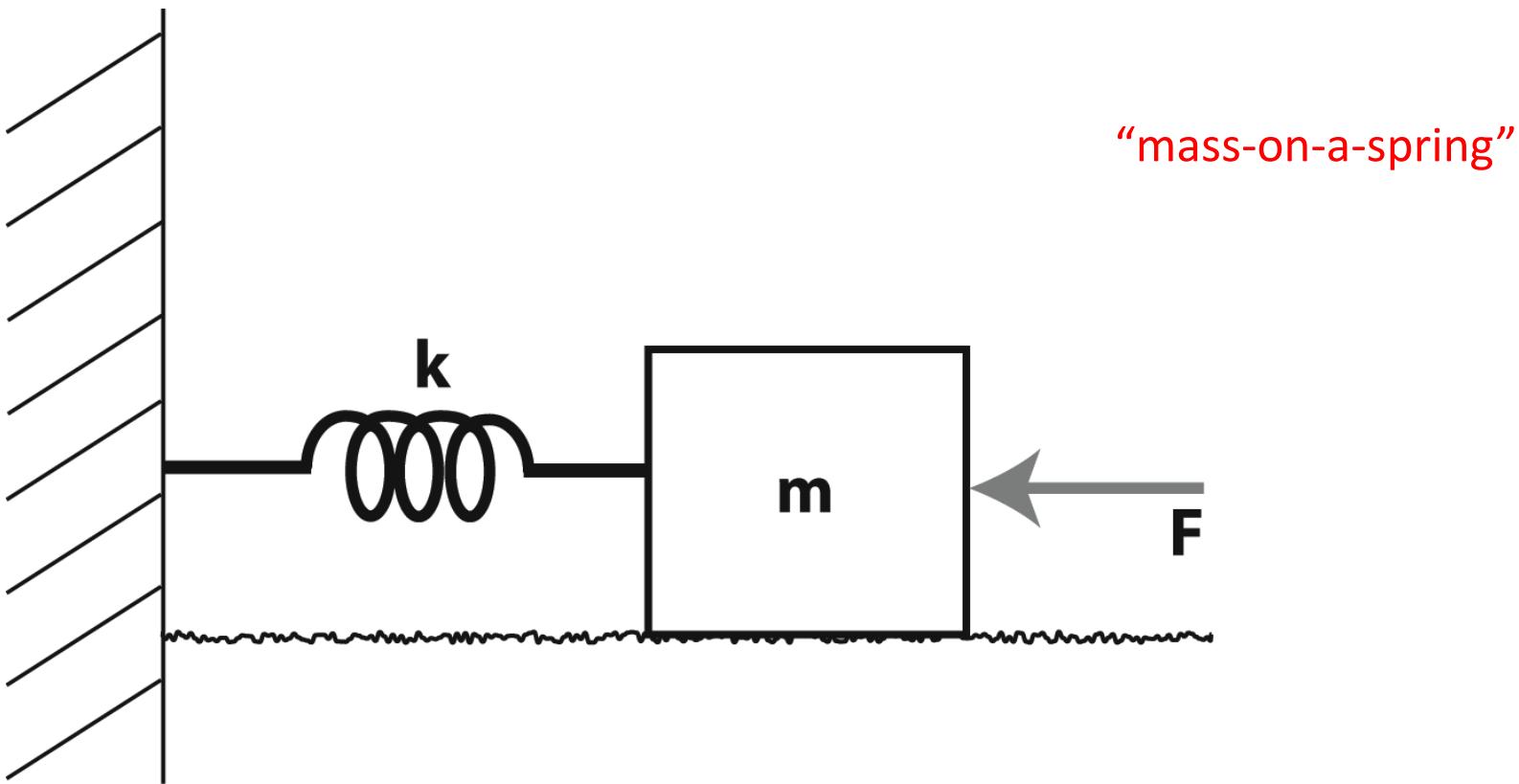


Jules Antoine Lissajous (1822-1880)

### *Sound Vibrations*



Tying this all back together....



- One of the more fundamental/canonical problems in all areas of physics...

## Fourier & the DDHO

Consider the driven case

$$m\ddot{x} + c\dot{x} + kx = F_{ext}$$

where the driving force is a superposition of individual forces

$$F_{ext} = \sum_n F_n(t)$$

Linearity allows us consider each separately...

$$m\ddot{x}_n + c\dot{x}_n + kx_n = F_n(t)$$

... and simply add them all back up!

$$x(t) = \sum_n x_n(t)$$

$$m\ddot{x} + c\dot{x} + kx = \sum_n (m\ddot{x}_n + c\dot{x}_n + kx_n) = \sum_n F_n(t) = F_{ext}$$

For linear systems, the "door" of superposition swings both ways!

## Fourier & the DDHO

Now, can we make a smart choice for what all those "individual" forces are?

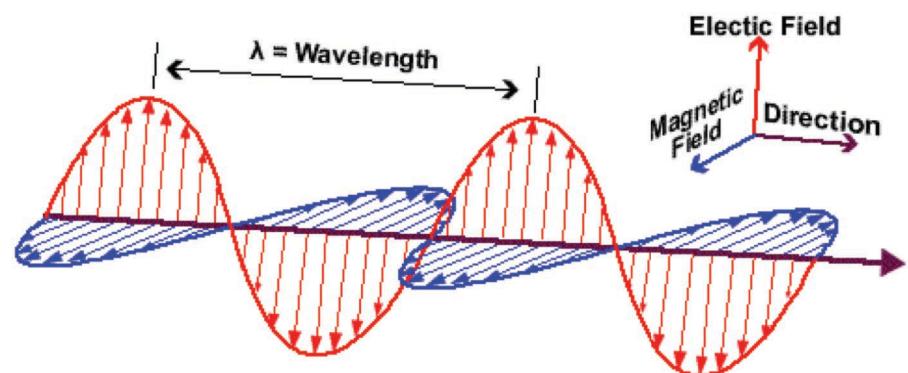
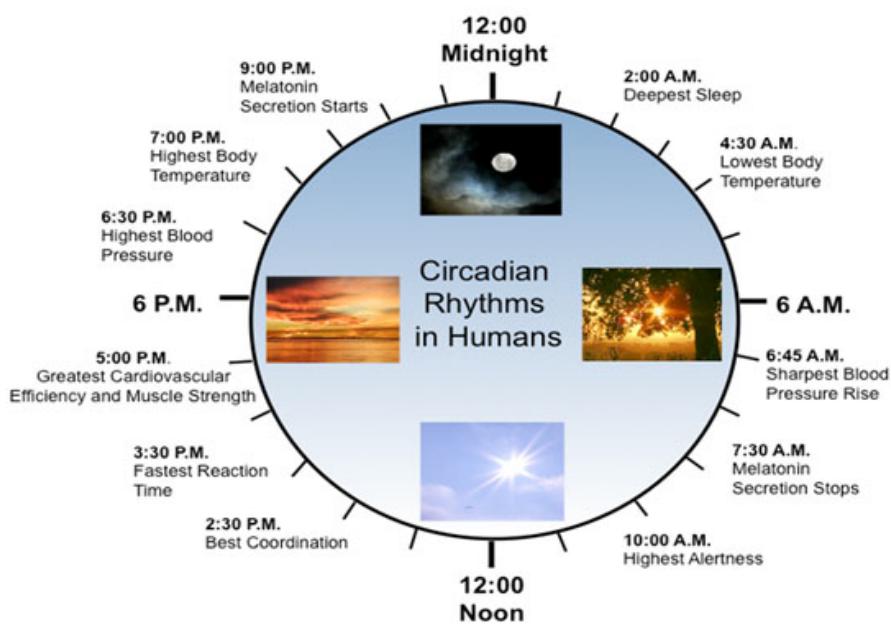
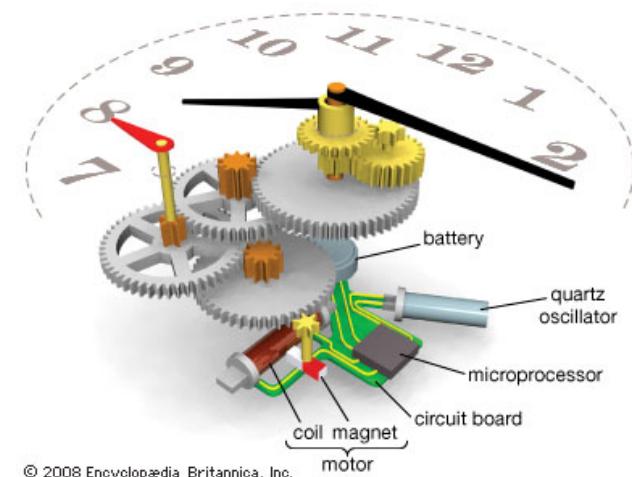
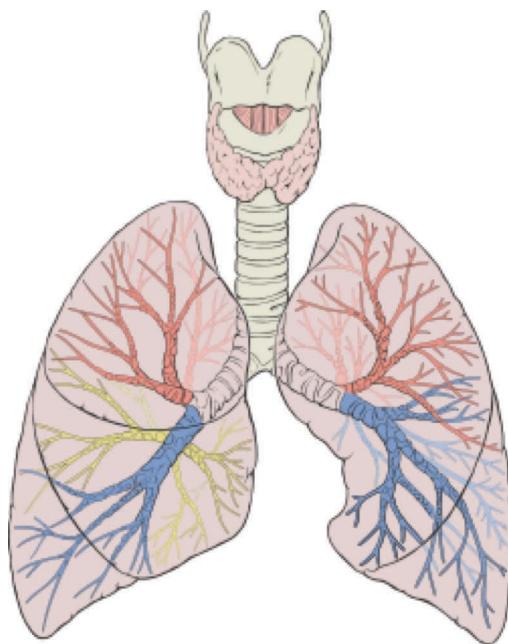
$$F_{ext} = \sum_n F_n(t)$$

In many instances, the driving force is something that eventually repeats itself\*

$$F_{ext}(t) = F_{ext}(t + T)$$

\* - Note that much of what we discuss here also usefully applies even when a given "signal" is not strictly periodic (or even remotely close to such!)

## Periodic forcing....



## Fourier & the DDHO

Now, can we make a smart choice for what all those "individual" forces are?

$$F_{ext} = \sum_n F_n(t)$$

In many instances, the driving force is something that eventually repeats itself

$$F_{ext}(t) = F_{ext}(t + T)$$

Then it is fairly natural to represent such a function via an infinite series of sinusoids (i.e., a **Fourier series**)

$$f(t) = \frac{1}{2}a_0 + \sum_{n=1}^{\infty} [a_n \cos(n\omega t) + b_n \sin(n\omega t)]$$

## Fourier & the DDHO

$$f(t) = \frac{1}{2}a_0 + \sum_{n=1}^{\infty} [a_n \cos(n\omega t) + b_n \sin(n\omega t)]$$

Just like a Taylor series expansion, there are a unique set of coefficients ( $a_n$  and  $b_n$ ) that determine the appropriate amplitude and phase of a given frequency term (i.e.,  $n\omega$ )

There is a reasonably clear "recipe" for these coefficients...

$$a_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \cos(n\omega t) dt \quad n = 0, 1, 2, \dots$$

$$b_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \sin(n\omega t) dt \quad n = 1, 2, \dots$$

We can also use complex numbers to manage "bookkeeping" more efficiently....

$$f(t) = \sum_n c_n e^{in\omega t} \quad n = 0, \pm 1, \pm 2, \dots$$

$$c_n = \frac{1}{T} \int_{-T/2}^{T/2} f(t) e^{-in\omega t} dt$$

## Fourier analysis

- Deep history throughout mathematics, physics, engineering, biology, .....
- Backbone of modern signal processing and linear systems theory
- Lays at foundation of many modern methodologies in medical imaging (e.g., MRI, CT scans)
- Builds off the basic idea of a *Taylor series* (which posits we can describe a function as an infinite series of polynomials)



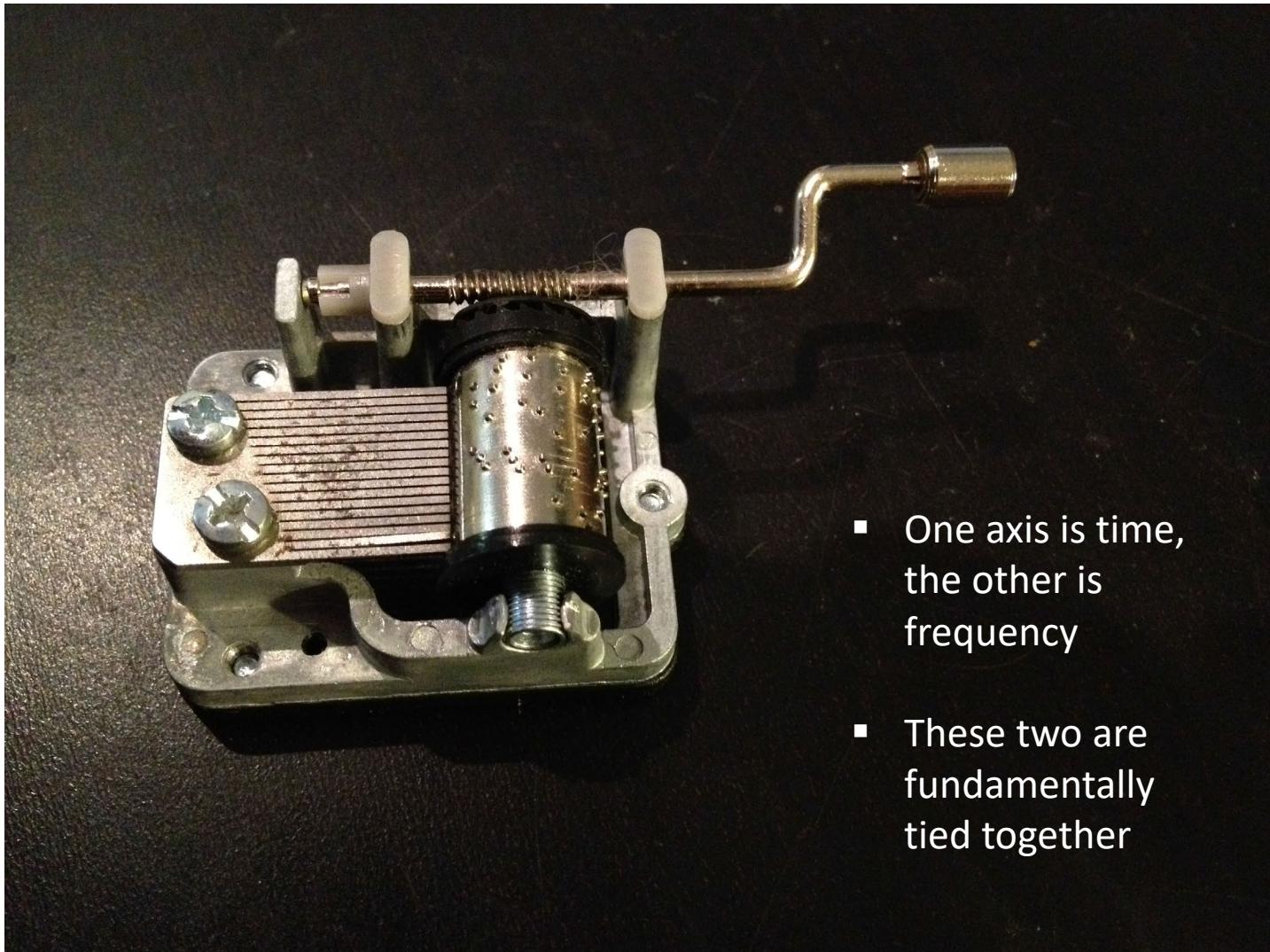
Joseph Fourier (1768-1830)

Basic idea: Represent ‘signal’ as a sum of sinusoids

Note: We focus on 1-D here for clarity, but these ideas generalize to higher dimensions (e.g., 2-D for images)

## Key idea: Fourier transform

- Allows one to go from a time domain description (e.g., recorded signal) to a spectral description (i.e., what frequency components make up that signal)



- One axis is time, the other is frequency
- These two are fundamentally tied together

## Fourier series

Intuitive connection back to Taylor series:

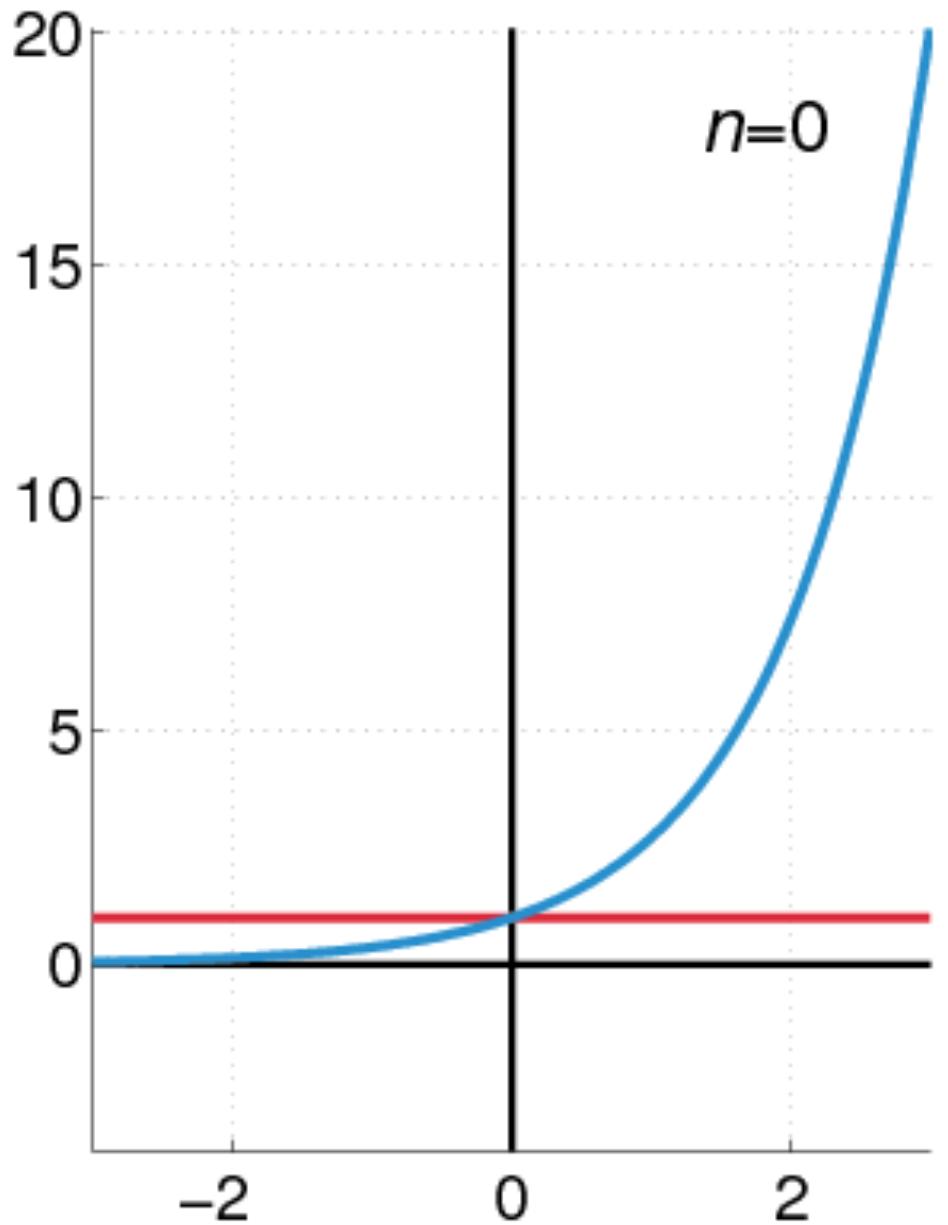
$$y(x_1 + \Delta x) \approx y(x_1) + \sum_{n=1}^N \frac{1}{n!} \left. \frac{d^n y}{dx^n} \right|_{x_1} (\Delta x)^n. \quad (\text{D.2})$$

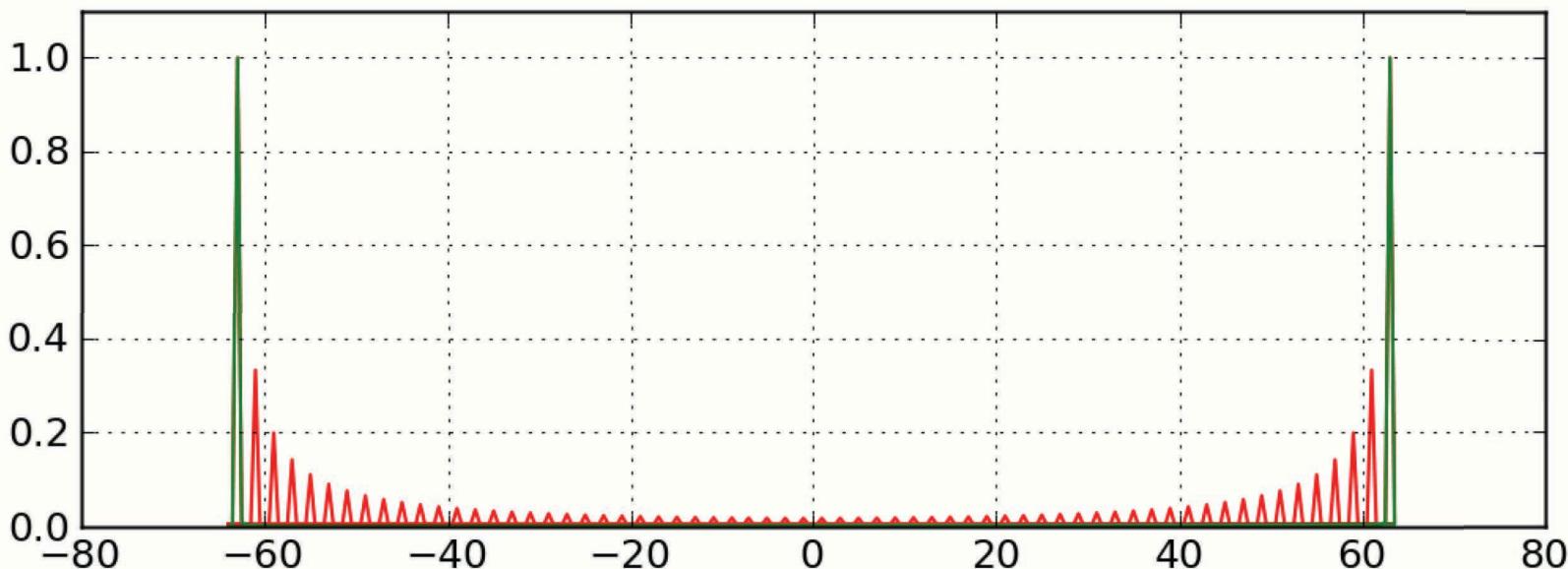
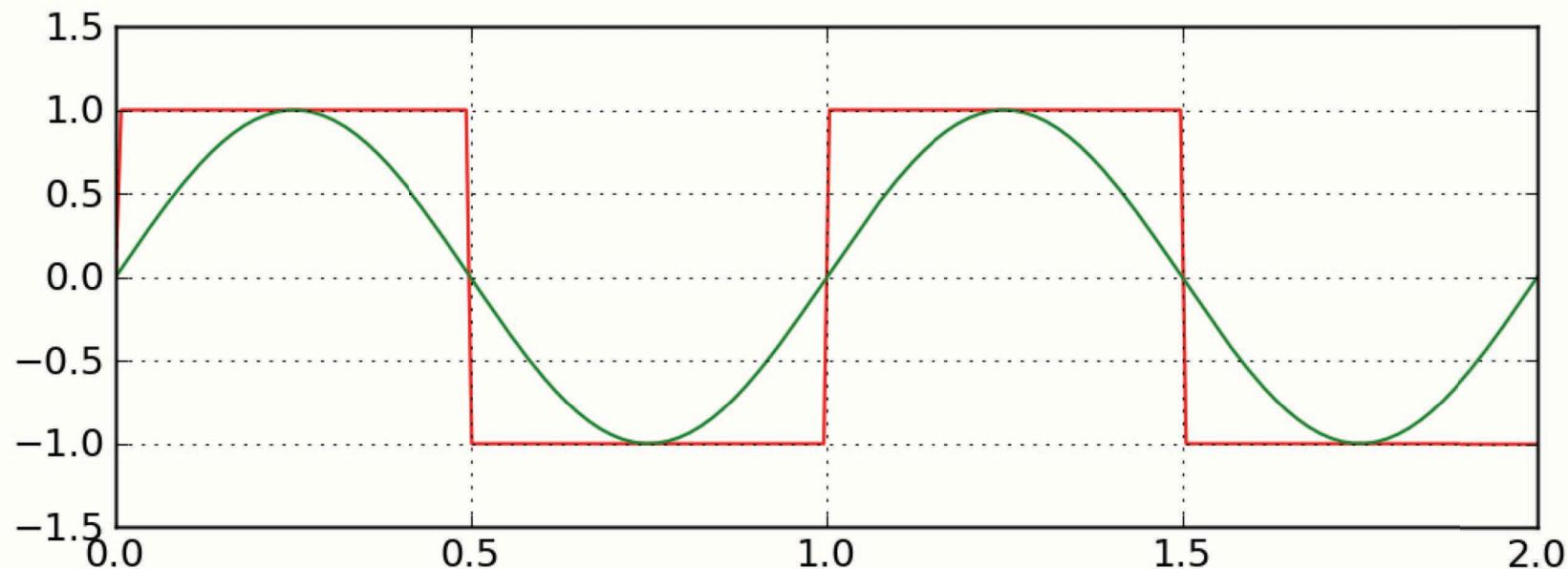
$$\begin{aligned} f(x) &= f(x_o) + f'(x_o)(x - x_o) + \frac{f''(x_o)}{2!}(x - x_o)^2 + \cdots + \frac{f^{(n)}(x_o)}{n!}(x - x_o)^n + \cdots \\ &= \sum_{n=0}^{\infty} \frac{f^{(n)}(x_o)}{n!}(x - x_o)^n \end{aligned}$$

Taylor series → Expand as a (infinite) sum of polynomials

Different Idea: Fourier series → Expand as a (infinite) sum of sinusoids

“The exponential function  $e^x$  (in blue),  
and the sum of the first  $n+1$  terms of  
its Taylor series at 0 (in red).”





## Fourier series

$$\begin{aligned}f(t) &= a_0 + a_1 \sin(\omega t) + b_1 \cos(\omega t) + \\&\quad + a_2 \sin(2\omega t) + b_2 \cos(2\omega t) + \\&\quad + a_3 \sin(3\omega t) + b_3 \cos(3\omega t) + \dots\end{aligned}$$

$$\begin{aligned}&= A_0 + A_1 \sin(\omega t + \phi_1) \\&\quad + A_2 \sin(2\omega t + \phi_2) \\&\quad + A_3 \sin(3\omega t + \phi_3) + \dots\end{aligned}$$

$$= \sum_{n=0}^{\infty} A_n \sin(n\omega t + \phi_n)$$

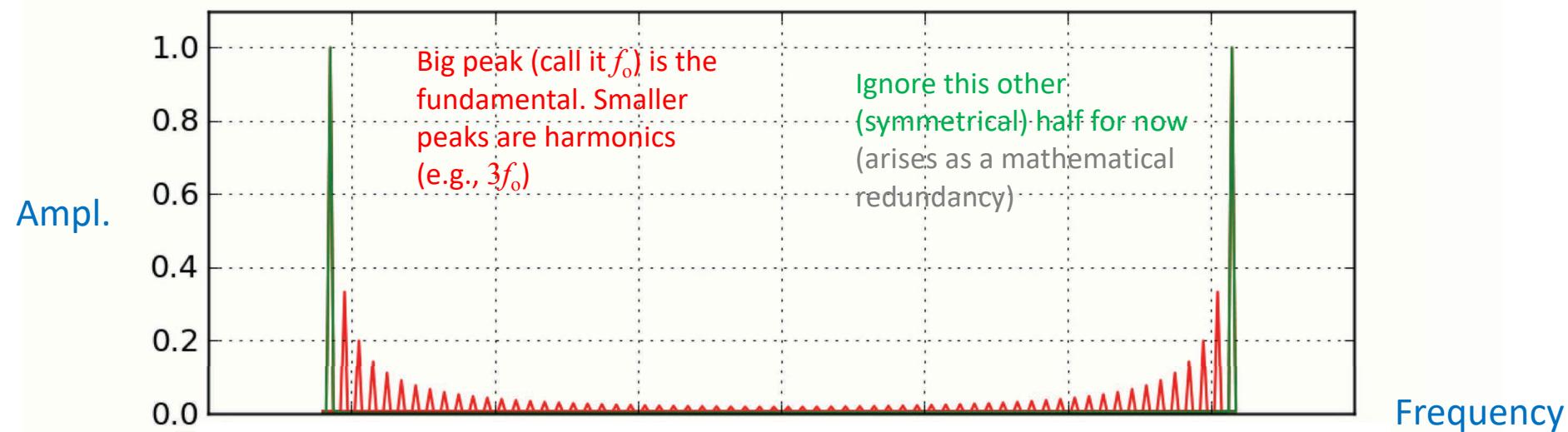
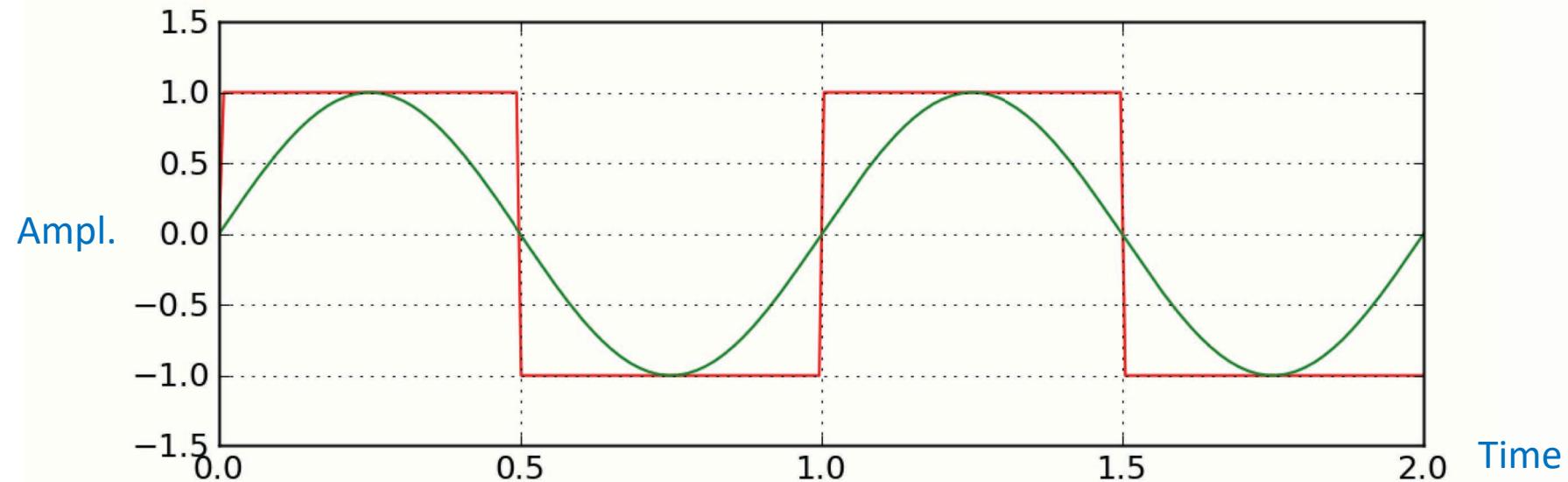
$$a_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \cos(n\omega t) dt \quad n = 0, 1, 2, \dots$$

$$b_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \sin(n\omega t) dt \quad n = 1, 2, \dots$$

$$= \sum_{n=0}^{\infty} B_n e^{in\omega t} \quad \text{where } B_n \in \mathbb{C}, \quad i = \sqrt{-1}$$

Complex #'s are much  
more compact and  
easier to deal with

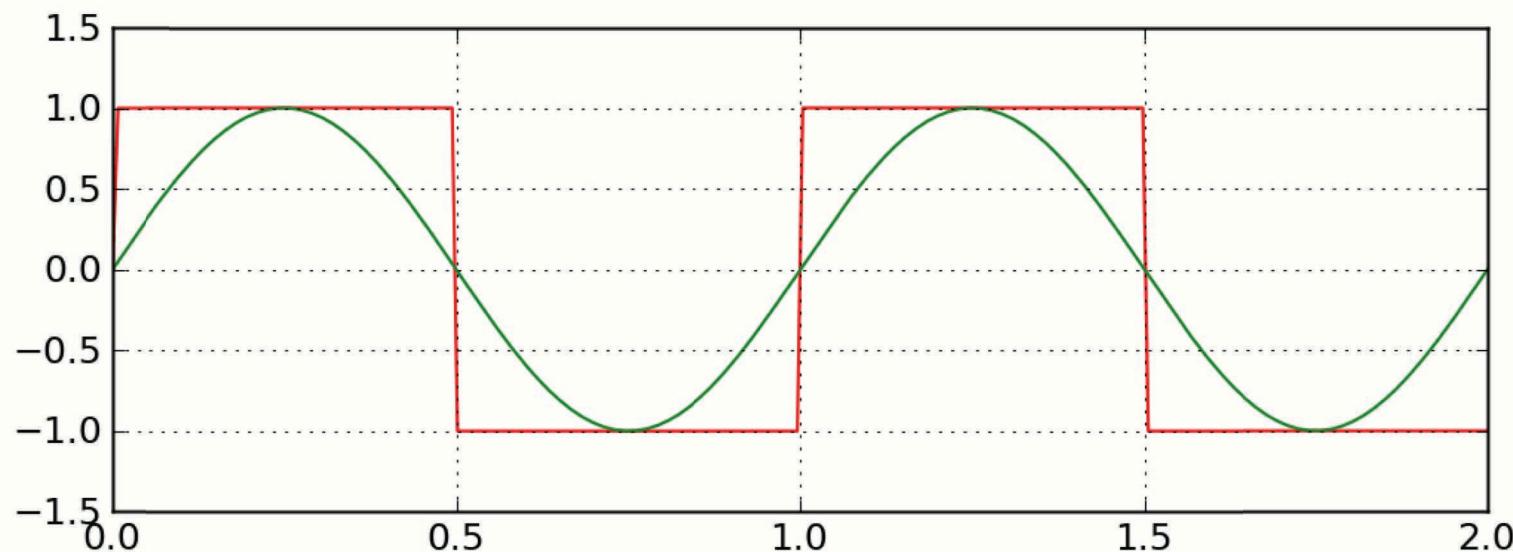
## e.g., Square "Waves"



```
% ### EXsquareW.m ###      2017.02.19 C.Bergevin
```

```
% Visually demonstrate the build-up of a square wave  
% by adding successive (user-specified) terms of the Fourier series  
% expansion; also quantifies the Gibbs phenomenon
```

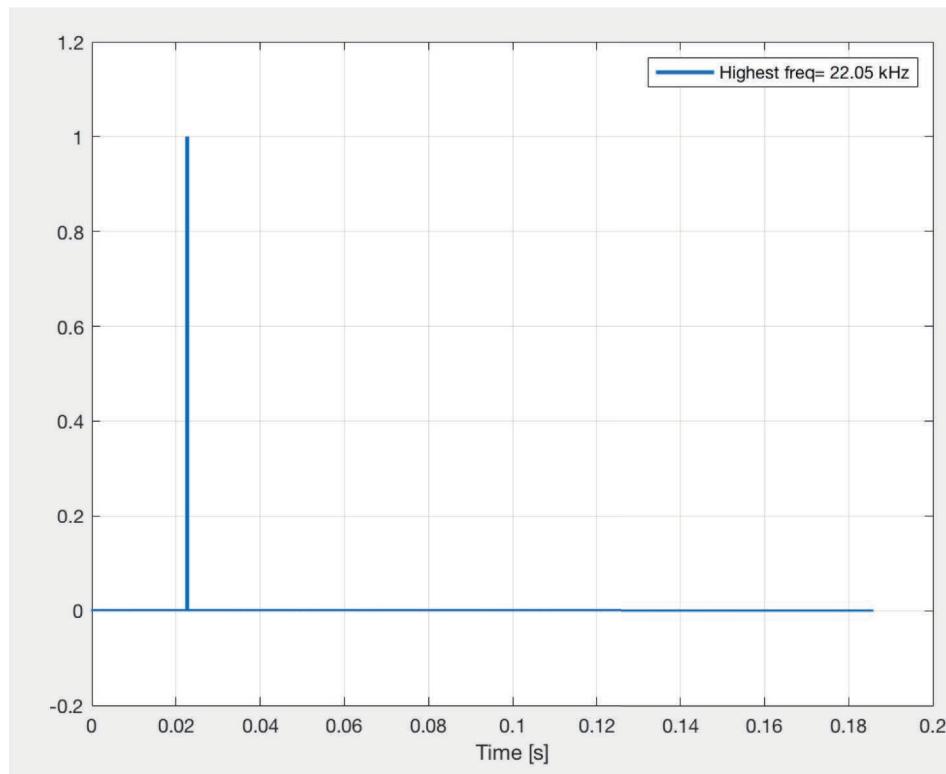
```
clear  
% =====  
P.order= [1 2 4 8 15 25 100 500];    % array of # of terms to compute {[1 2 4 8 15 25 100 500]}  
P.tau= 1;    % period {1}  
P.A= 1;      % peak-to-peak amplitude {1}  
P.M= 10000;   % total # of points per interval (must be even?) {1000}  
P.pause= 0.5;  % time to pause between displaying new iterates [s] {0.5}  
% =====  
t= linspace(-1.5*P.tau,1.5*P.tau,3*P.M); % time array  
squareT= repmat(P.A*([zeros(P.M/2,1);ones(P.M/2,1)]-0.5),3,1)';    % create sawtooth baseline  
% --- use a loop to add in the terms  
for mm=1:numel(P.order)  
    tempN= P.order(mm);  
    squareF= 0;      % dummy initial indexer  
    for nn= 1:tempN  
        nextTerm= (2*P.A/pi)*(1/(2*nn-1))*sin((2*nn-1)*(2*pi/P.tau)*t);    % create next term in  
series  
        squareF= squareF+ nextTerm;  
    end  
    % --- estimate "overshoot"  
    [M,indx]= max(squareF);  
    disp(['Overshoot ratio ~',num2str(M/max(squareT))]);  
    % --- visualize  
    figure(1); clf;  
    h1= plot(t,squareT,'b-','LineWidth',2); hold on; grid on; xlabel('x'); ylabel('y');  
    ylim(0.65*P.A*[-1 1]);  
    h2= plot(t,squareF,'r--','LineWidth',2); legend([h1 h2], 'square', 'Fourier  
series', 'Location', 'NorthWest');  
    title(['(truncated) Fourier reconstruction of square wave w/ ',num2str(tempN), ' terms']);  
    pause(P.pause);  
end
```



## Recall: Impulses

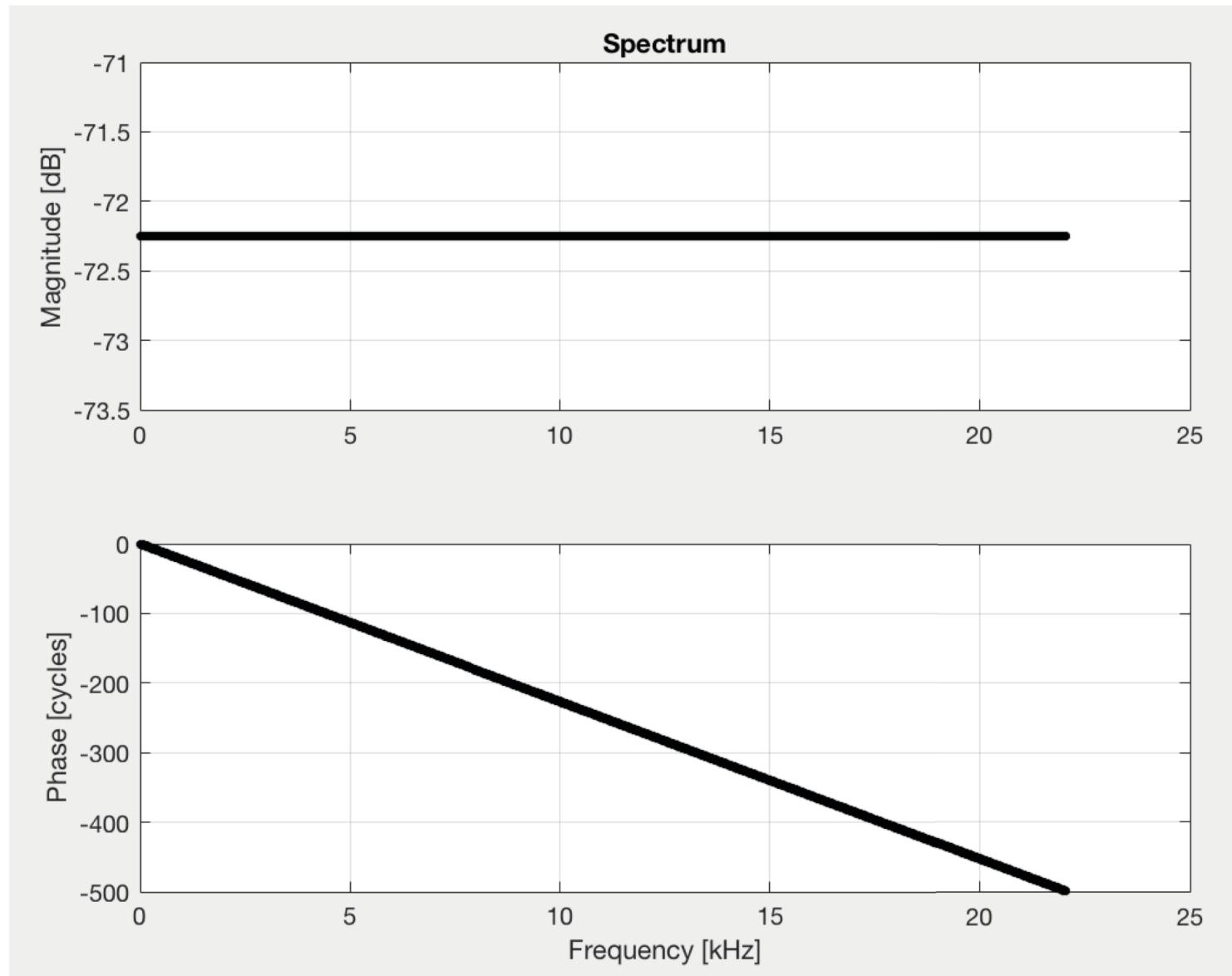
$$\delta(x) = \begin{cases} +\infty, & x = 0 \\ 0, & x \neq 0 \end{cases}$$

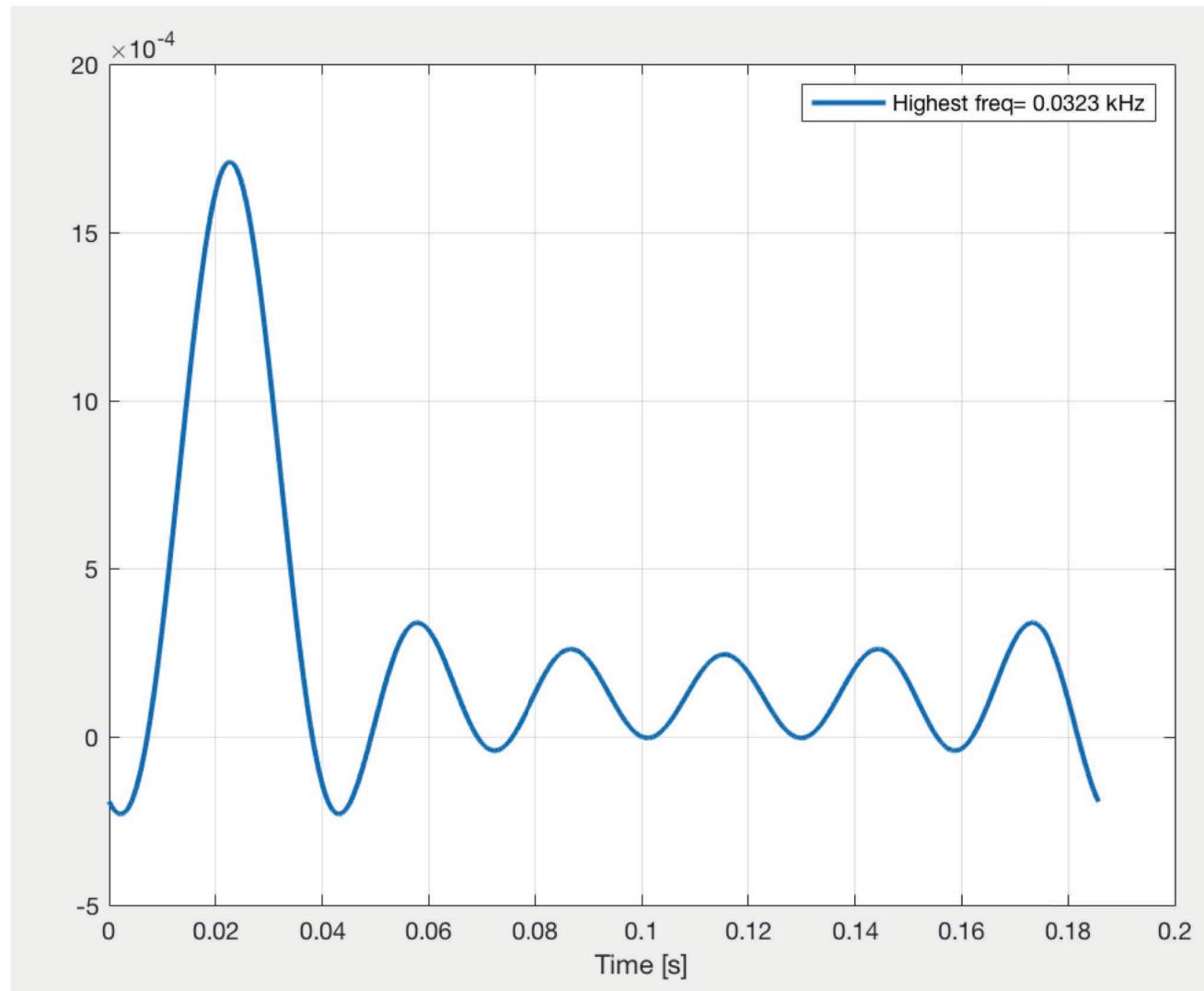
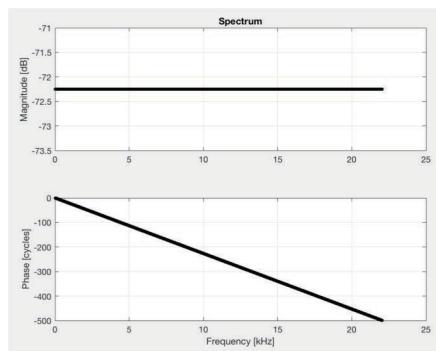
$$\int_{-\infty}^{\infty} \delta(x) dx = 1.$$

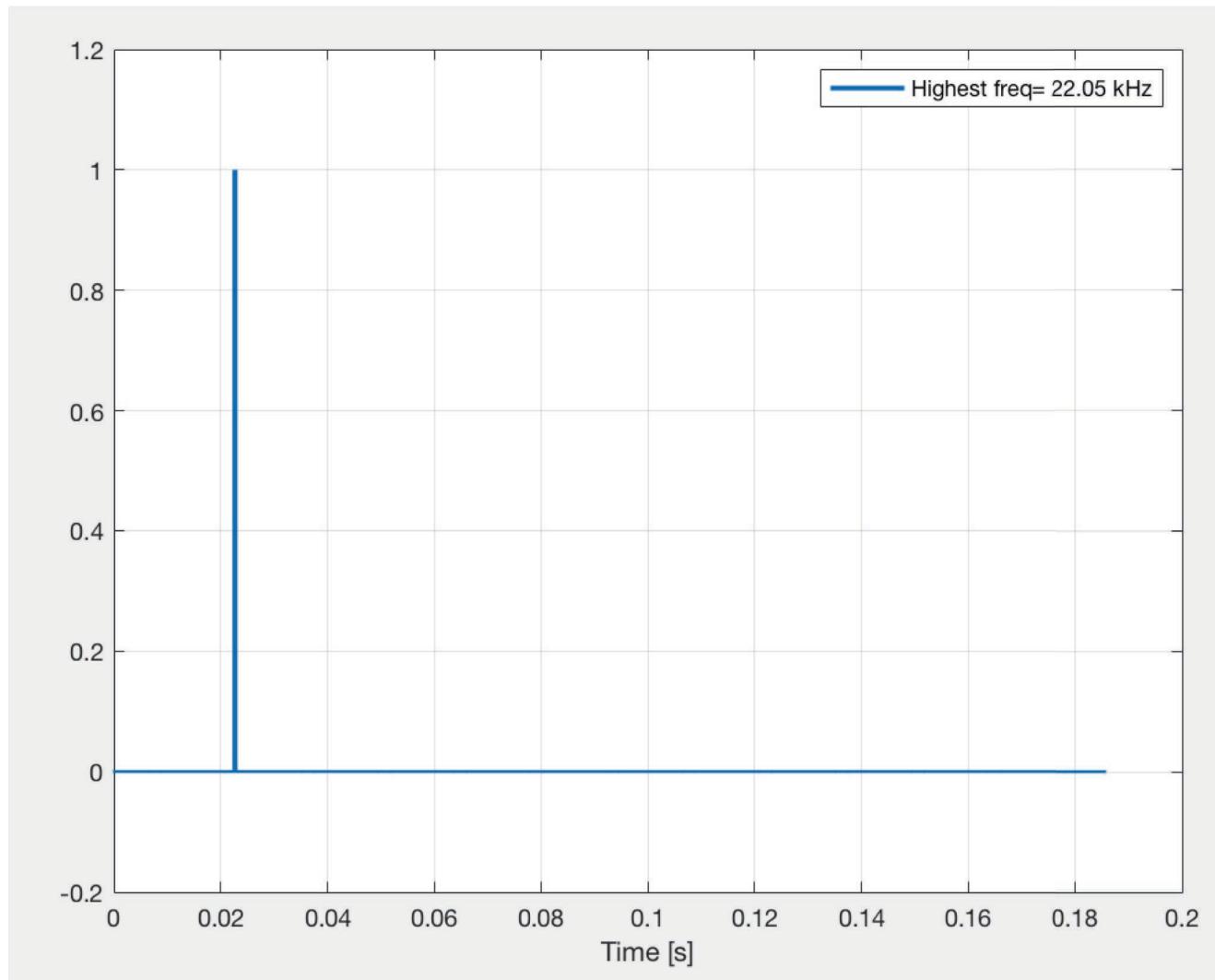


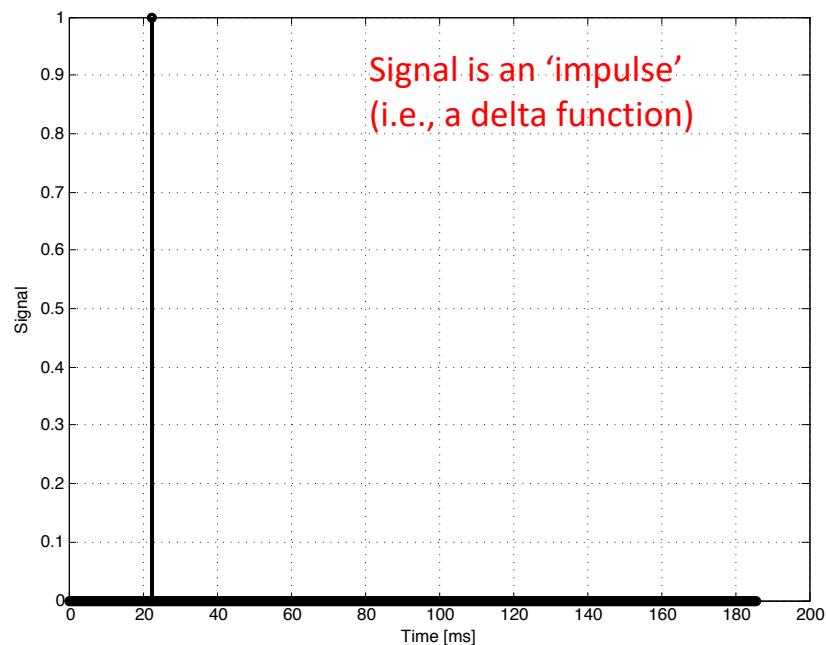
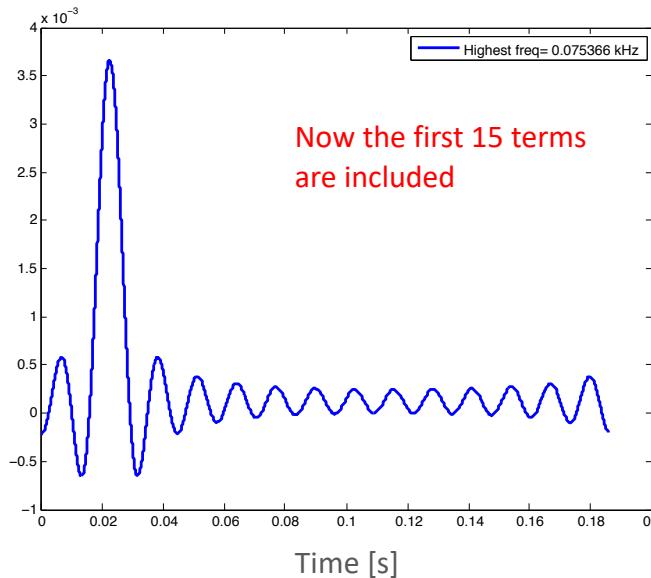
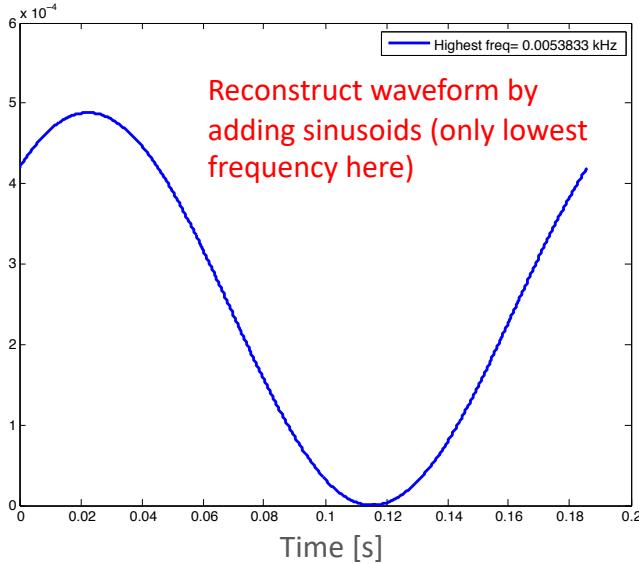
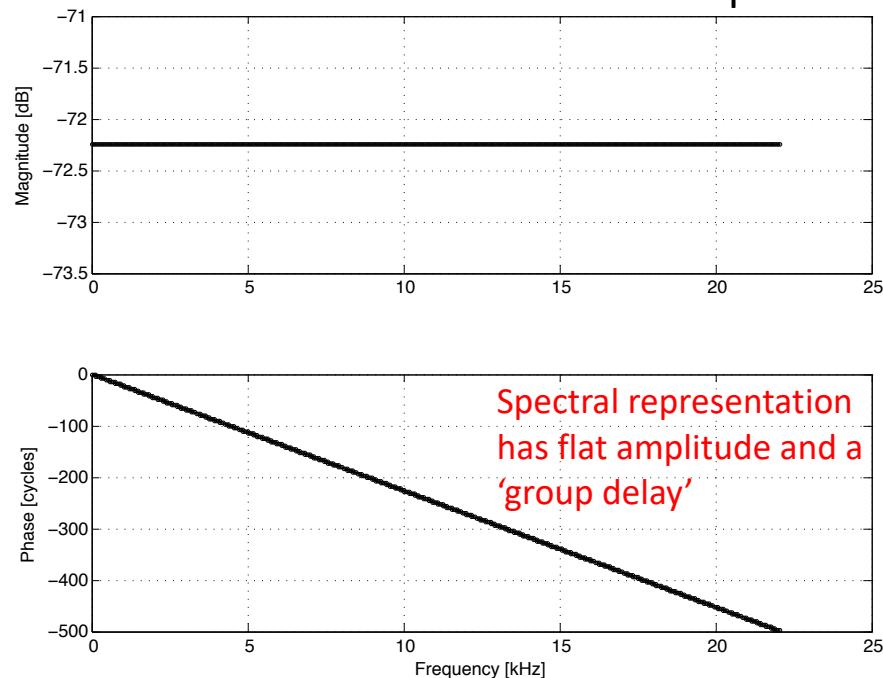
Delta functions are much more ubiquitous in your lives  
(and science) than you might think/realize.....

```
% ### EXbuildImpulse2.m ###           11.03.14
clear; clf;
%
SR= 44100;          % sample rate [Hz]
Npoints= 8192;      % length of fft window (# of points) [should ideally be 2^N]
% [time window will be the same length]
INDXon= 1000;       % index at which click turns 'on' (i.e., go from 0 to 1)
INDXoff= 1001;      % index at which click turns 'off' (i.e., go from 1 to 0)
%
dt= 1/SR;   % spacing of time steps
freq= [0:Npoints/2];    % create a freq. array (for FFT bin labeling)
freq= SR*freq./Npoints;
t=[0:1/SR:(Npoints-1)/SR]; % create an appropriate array of time points
%
% build signal
clktemp1= zeros(1,Npoints); clktemp2= ones(1,INDXoff-INDXon);
signal= [clktemp1(1:INDXon-1) clktemp2 clktemp1(INDXoff:end)];
%
% *****
% plot "final" time waveform of signal
if 1==1
    figure(3); clf; plot(t*1000,signal,'ko-','MarkerSize',5)
    grid on; hold on; xlabel('Time [ms]'); ylabel('Signal'); title('Time Waveform')
end
%
% *****
% now compute/plot FFT of the signal
sigSPEC= rfft(signal);
%
% MAGNITUDE
figure(1); clf;
subplot(211); plot(freq/1000,db(sigSPEC),'ko-','MarkerSize',3)
hold on; grid on; ylabel('Magnitude [dB]'); title('Spectrum (or "Look Up Table")')
%
% PHASE
subplot(212); plot(freq/1000,cycts(sigSPEC),'ko-','MarkerSize',3)
xlabel('Frequency [kHz]'); ylabel('Phase [cycles]'); grid on;
%
% *****
% now make animation of click getting built up, using the info from the FFT
sum= zeros(1,numel(t)); % (initial) array for reconstructed waveform
inclV=[1:30,floor(linspace(31,floor(0.9*numel(freq)),100)),...
        floor(linspace(0.9*numel(freq),numel(freq),20))];
%
figure(2); clf; grid on;
for nn=1:numel(freq)
    sum= sum+ abs(sigSPEC(nn))*cos(2*pi*freq(nn)*t + angle(sigSPEC(nn)));
    if ismember(nn,inclV), plot(t,sum,'LineWidth',2); grid on; xlabel('Time [s]');
    legend(['Highest freq= ',num2str(freq(nn)/1000), ' kHz'])
    pause(3/(nn));
end
```





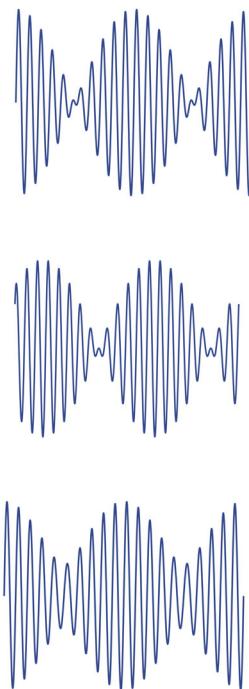
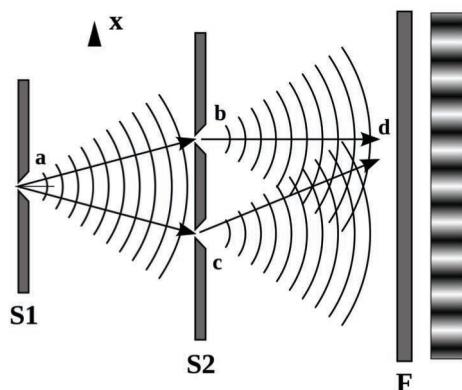


**Temporal****Spectral**

→ Eventually all the sinusoids add up such that things cancel out everywhere except at the point of the impulse!

## Summary

### Superposition



DDHO is a linear system

$$m\ddot{x} + c\dot{x} + kx = \sum_n (m\ddot{x}_n + c\dot{x}_n + kx_n) = \sum_n F_n(t) = F_{ext}$$

Periodic forcing & superposition



$$F_{ext} = \sum_n F_n(t)$$

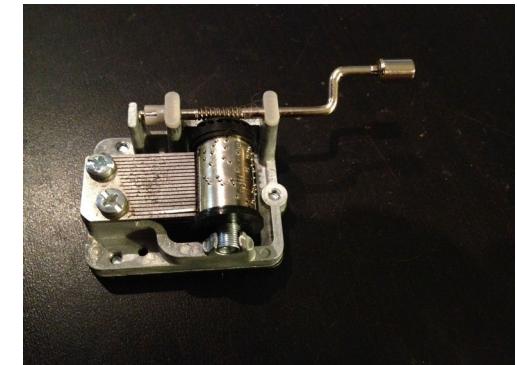


Fourier analysis

$$f(t) = \frac{1}{2}a_0 + \sum_{n=1}^{\infty} [a_n \cos(n\omega t) + b_n \sin(n\omega t)]$$

$$a_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \cos(n\omega t) dt \quad n = 0, 1, 2, \dots$$

$$b_n = \frac{2}{T} \int_{-T/2}^{T/2} f(t) \sin(n\omega t) dt \quad n = 1, 2, \dots$$



```

%%% EXspecREP3.m ###          10.29.14
% Example code to just fiddle with basics of discrete FFTs and connections
% back to common real-valued time waveforms
% --> Demonstrates several useful concepts such as 'quantizing' the frequency
% Requires: rfft.m, irfft.m, cycs.m, db.m, cyc.m
% ----
% Stimulus Type Legend
% stimT= 0 - non-quantized sinusoid
% stimT= 1 - quantized sinusoid
% stimT= 2 - one quantized sinusoid, one un-quantized sinusoid
% stimT= 3 - two quantized sinusoids
% stimT= 4 - click I.e., an impulse)
% stimT= 5 - noise (uniform in time)
% stimT= 6 - chirp (flat mag.)
% stimT= 7 - noise (Gaussian; flat spectrum, random phase)
% stimT= 8 - exponentially decaying sinusoid (i.e., HO impulse response)

clear; clf;
% -----
SR= 44100;           % sample rate [Hz]
Npoints= 8192;        % length of fft window (# of points) [should ideally be 2^N]
                      % [time window will be the same length]
stimT= 8;             % Stimulus Type (see legend above)
f= 2580.0;            % Frequency (for waveforms w/ tones) [Hz]
ratio= 1.22;           % specify f2/f1 ratio (for waveforms w/ two tones)
% Note: Other stimulus parameters can be changed below
% -----
dt= 1/SR;             % spacing of time steps
freq= [0:Npoints/2];   % create a freq. array (for FFT bin labeling)
freq= SR*freq./Npoints;
% quantize the freq. (so to have an integral # of cycles in time window)
df = SR/Npoints;
fQ= ceil(f/df)*df;    % quantized natural freq.
t=[0:1/SR:(Npoints-1)/SR]; % create an array of time points, Npoints long
% ---
% compute stimulus
if stimT==0 % non-quantized sinusoid
    signal= cos(2*pi*f*t);
    disp(sprintf(' \n *Stimulus* - (non-quantized) sinusoid, f = %g Hz \n', f));
    disp(sprintf('specified freq. = %g Hz', f));
elseif stimT==1 % quantized sinusoid
    signal= cos(2*pi*fQ*t);
    disp(sprintf(' \n *Stimulus* - quantized sinusoid, f = %g Hz \n', fQ));
    disp(sprintf('specified freq. = %g Hz', fQ));
    disp(sprintf('quantized freq. = %g Hz', fQ));
elseif stimT==2 % one quantized sinusoid, one un-quantized sinusoid
    signal= cos(2*pi*fQ*t) + cos(2*pi*ratio*fQ*t);
    disp(sprintf(' \n *Stimulus* - two sinusoids (one quantized, one not) \n'));
elseif stimT==3 % two quantized sinusoids
    fQ2= ceil(ratio*f/df)*df;
    signal= cos(2*pi*fQ*t) + cos(2*pi*fQ2*t);
    disp(sprintf(' \n *Stimulus* - two sinusoids (both quantized) \n'));
elseif stimT==4 % click
    CLKon= 1000;           % index at which click turns 'on' (starts at 1)
    CLKoff= 1001;           % index at which click turns 'off'
    clktemp1= zeros(1,Npoints);
    clktemp2= ones(1,CLKoff-CLKon);
    signal= [clktemp1(1:CLKon-1) clktemp2 clktemp1(CLKoff:end)];
    disp(sprintf(' \n *Stimulus* - Click \n'));
elseif stimT==5 % noise (flat)
    signal= rand(1,Npoints);
    disp(sprintf(' \n *Stimulus* - Noise! \n'));
elseif stimT==6 % chirp (flat)
    f1s= 2000.0;            % if a chirp (stimT=2) starting freq. [Hz] [freq. swept linearly w/ time]
    f1E= 4000.0;            % ending freq. (energy usually extends twice this far out)
    f1SQ= ceil(f1s/df)*df;  %quantize the start/end freqs. (necessary?)
    f1EQ= ceil(f1E/df)*df;
    % LINEAR sweep rate
    fSWP= f1SQ + (f1EQ-f1SQ)*(SR/Npoints)*t;
    signal = sin(2*pi*fSWP.*t)';
    disp(sprintf(' \n *Stimulus* - Chirp \n'));
elseif stimT==7 % noise (Gaussian)
    Asize=Npoints/2 +1;
    % create array of complex numbers w/ random phase and unit magnitude
    for n=1:Asize
        theta= rand*2*pi;
        N2(n)= exp(i*theta);
    end
    N2=N2';
    % now take the inverse FFT of that using Chris' irfft.m code
    tNoise=irfft(N2);
    % scale it down so #s are between -1 and 1 (i.e. normalize)
    if (abs(min(tNoise)) > max(tNoise))
        tNoise= tNoise/abs(min(tNoise));
    else
        tNoise= tNoise/max(tNoise);
    end
    signal= tNoise;
    disp(sprintf(' \n *Noise* - Gaussian, flat-spectrum \n'));
elseif stimT==8 % exponentially decaying cos
    alpha= 500;
    signal= exp(-alpha*t).*sin(2*pi*fQ*t);
    disp(sprintf(' \n *Exponentially decaying (quantized) sinusoid* \n'));
end

% -----
% *****
figure(1); clf      % plot time waveform of signal
plot(t*1000,signal,'k_-','MarkerSize',5); grid on; hold on;
xlabel('Time [ms]'); ylabel('Signal'); title('Time Waveform')
% *****
% now plot rfft of the signal
% NOTE: rfft just takes 1/2 of fft.m output and normalizes
sigSPEC= rfft(signal);
figure(2); clf; % MAGNITUDE
subplot(211)
plot(freq/1000,db(sigSPEC),'ko-','MarkerSize',3)
hold on; grid on;
ylabel('Magnitude [dB]')
title('Spectrum')
subplot(212) % PHASE
plot(freq/1000,cycts(sigSPEC),'ko-','MarkerSize',3)
xlabel('Frequency [kHz]'); ylabel('Phase [cycles]'); grid on;
% -----
% play the stimuli as an output sound?
if (1==1), sound(signal,SR); end
% -----
% compute inverse Fourier transform and plot?
if 1==1
    figure(1);
    signalINV= irfft(sigSPEC);
    plot(t*1000,signalINV,'rx','MarkerSize',4)
    legend('Original waveform','Inverse transformed')
end

```

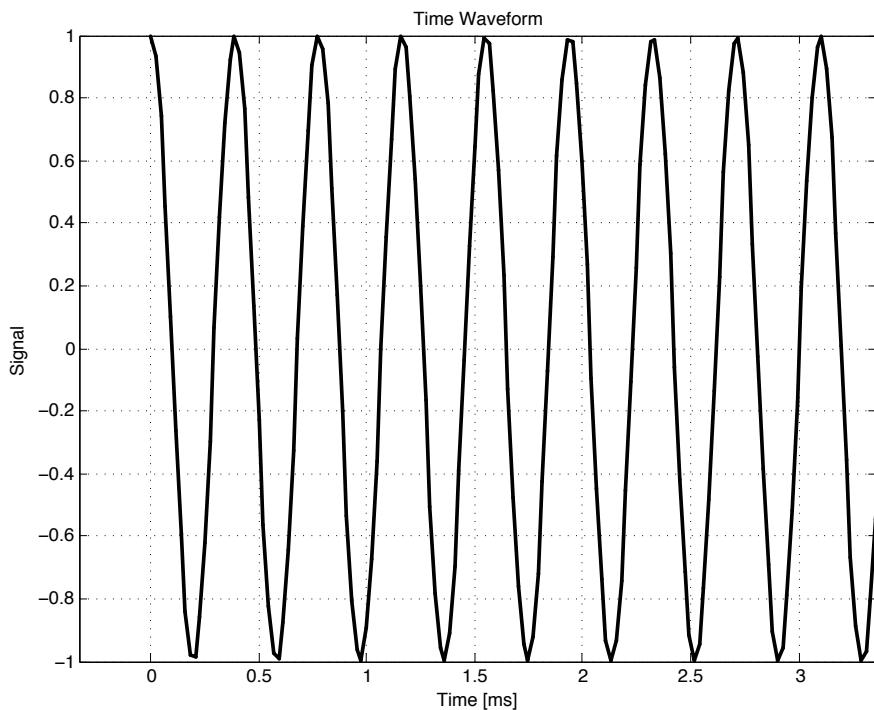
## Fourier transforms of basic (1-D) waveforms

EXspecREP3.m

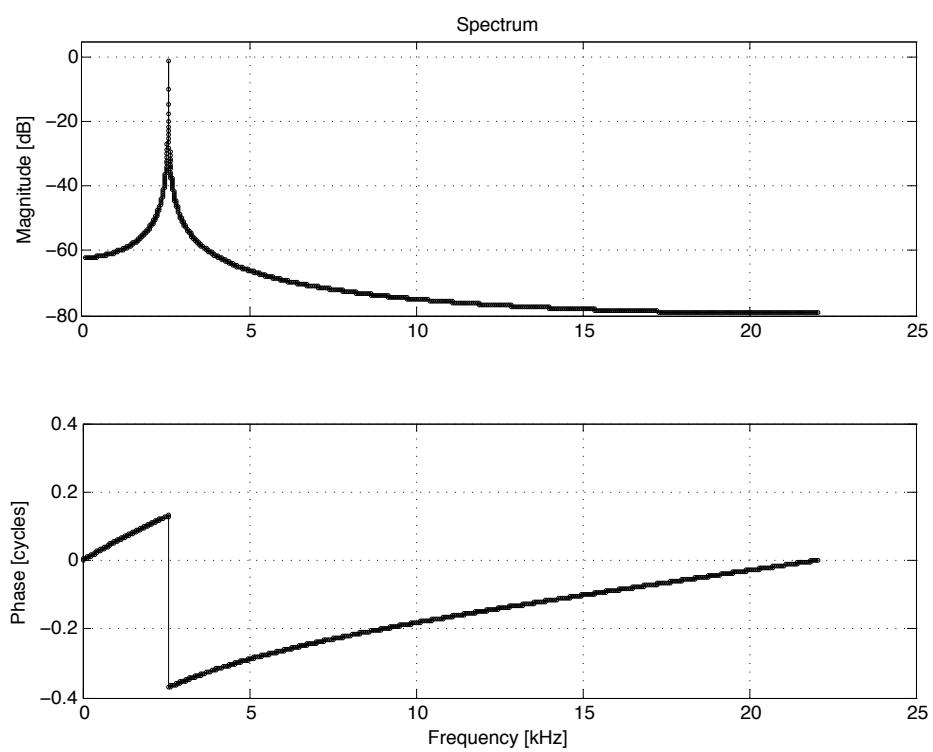
stimT= 0 – non-quantized sinusoid

SR= 44100; % sample rate [Hz]  
Npoints= 8192; % length of fft window

Time domain



Spectral domain



Note: The phase is ‘unwrapped’  
in all the spectral plots

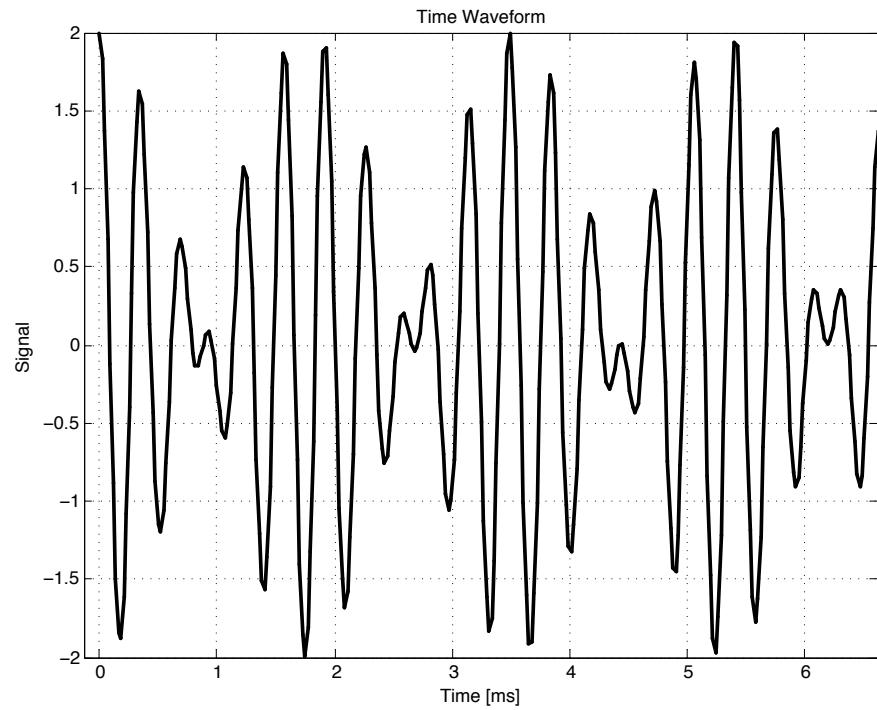
- Magnitude shows a peak at the sinusoid’s frequency

## Fourier transforms of basic (1-D) waveforms

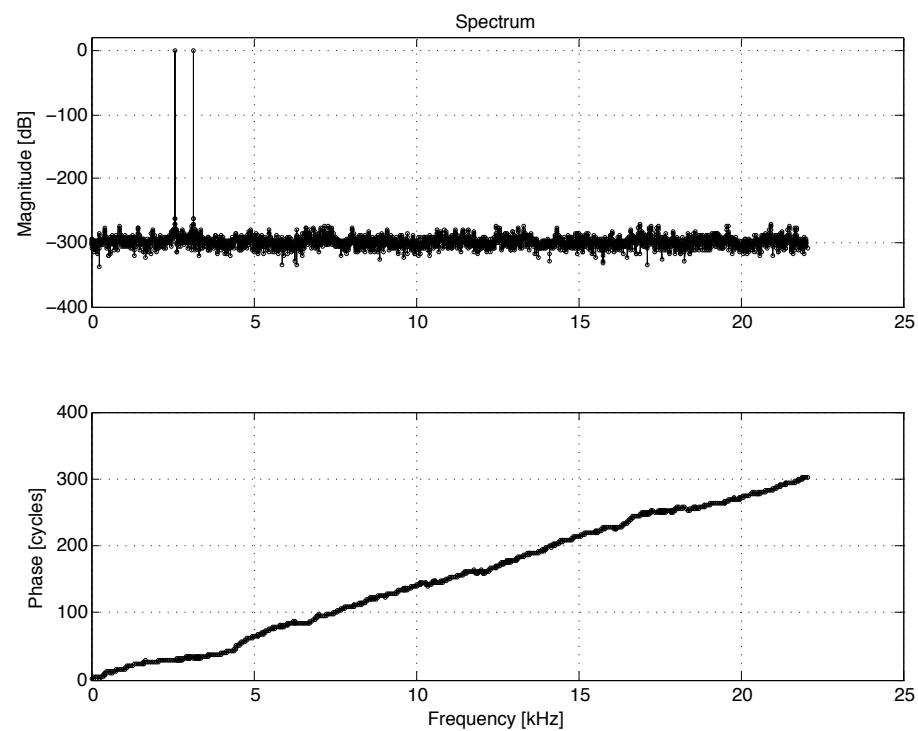
EXspecREP3.m

stimT= 3 – two quantized sinusoids

Time domain



Spectral domain



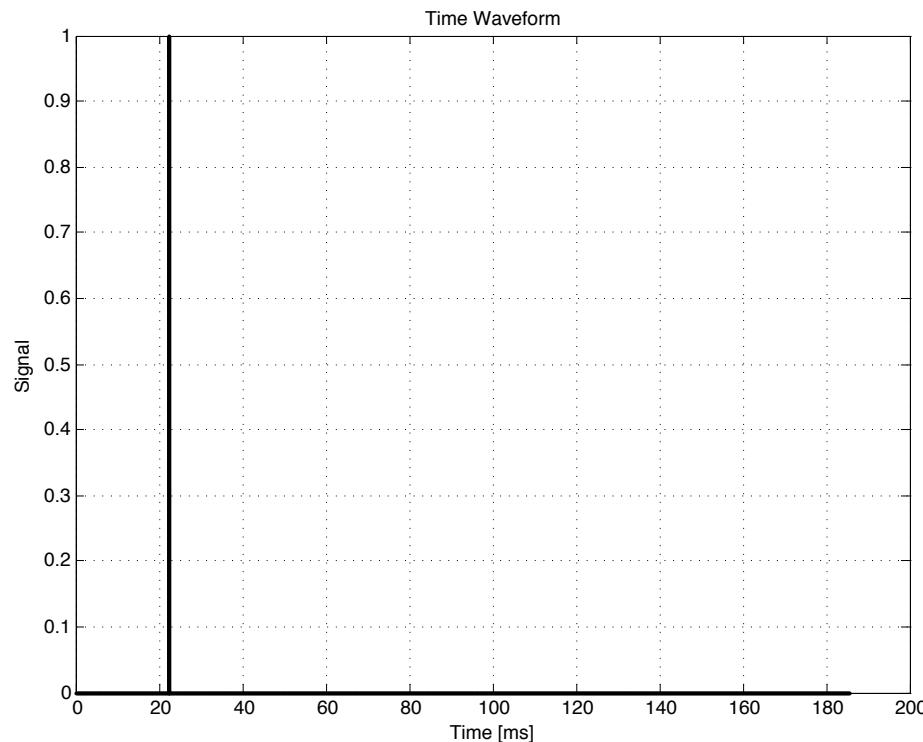
- Magnitude shows two peaks (note the ‘beating’ in the time domain)

## Fourier transforms of basic (1-D) waveforms

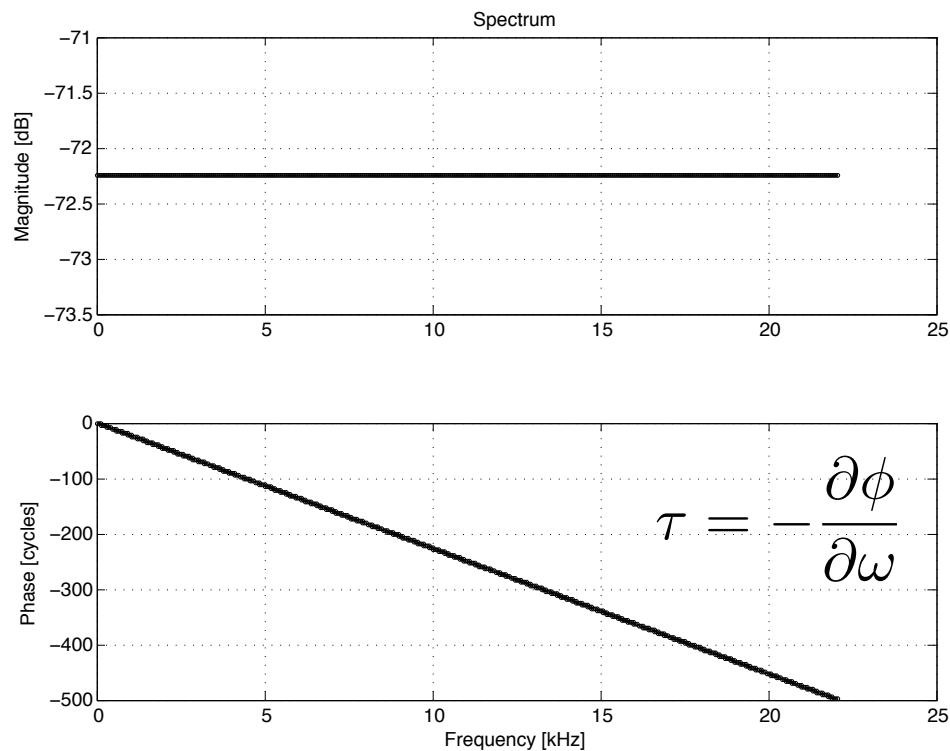
ExspecREP3.m

stimT= 4 - click (i.e., an impulse)

Time domain



Spectral domain



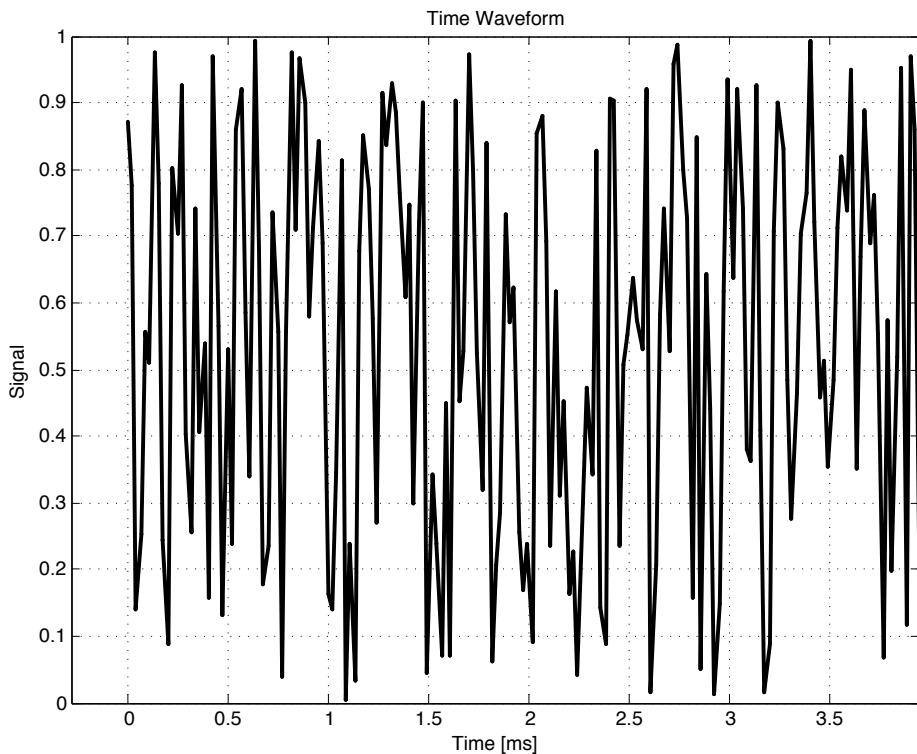
- Click has a flat magnitude (This is also a good place to mention the concept of a '*group delay*'')

## Fourier transforms of basic (1-D) waveforms

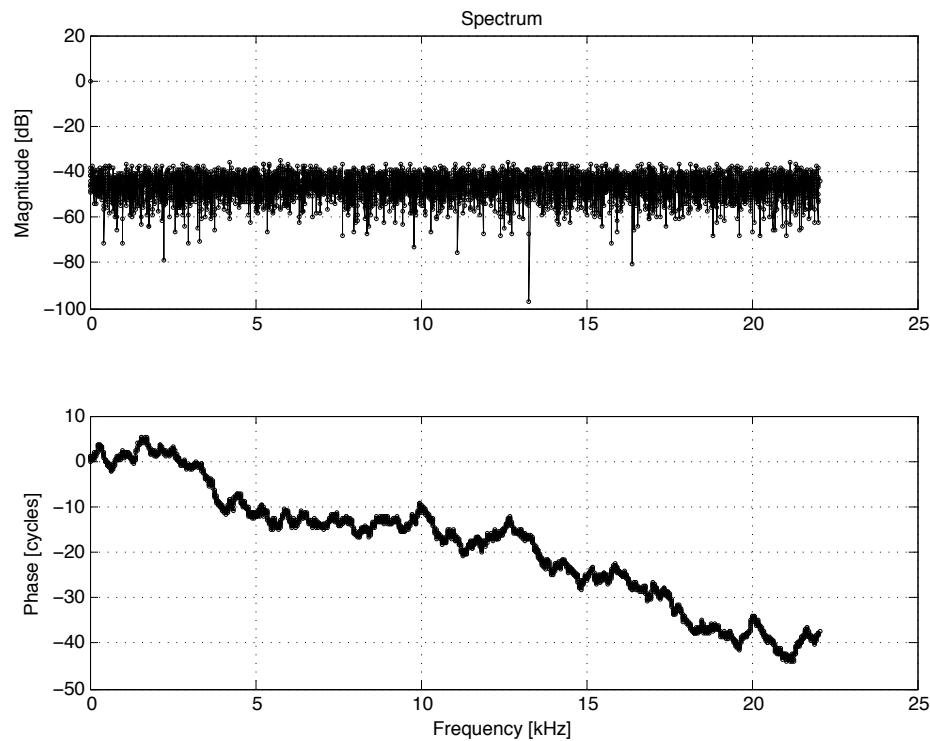
EXspecREP3.m

stimT= 5 – noise (uniform distribution)

Time domain



Spectral domain



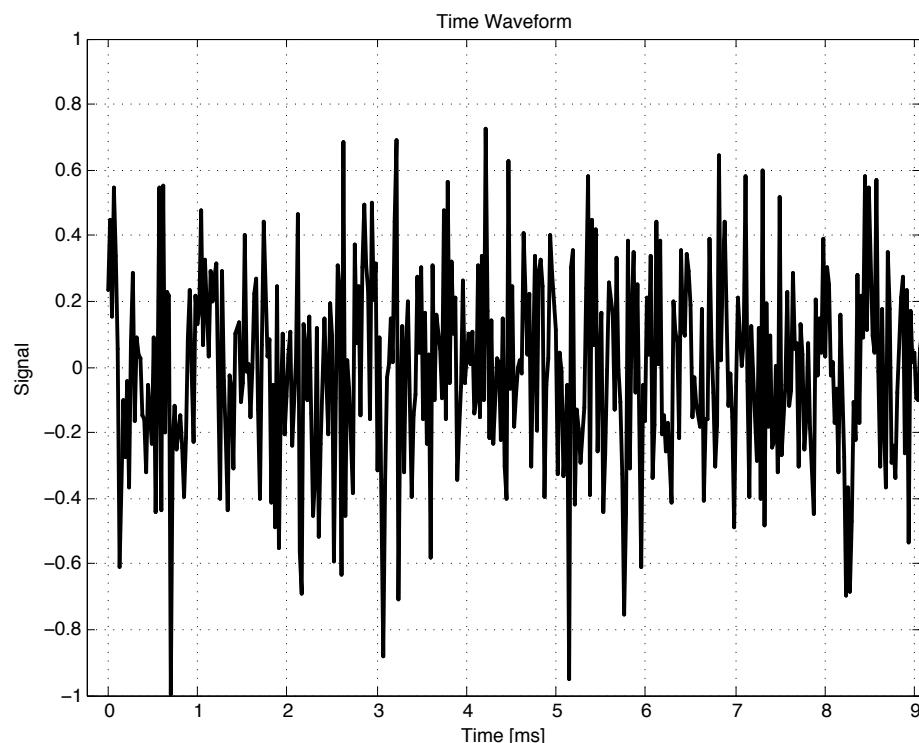
- Magnitude is flat-ish (on log scale), but actually noisy. Phase is noisy too.

## Fourier transforms of basic (1-D) waveforms

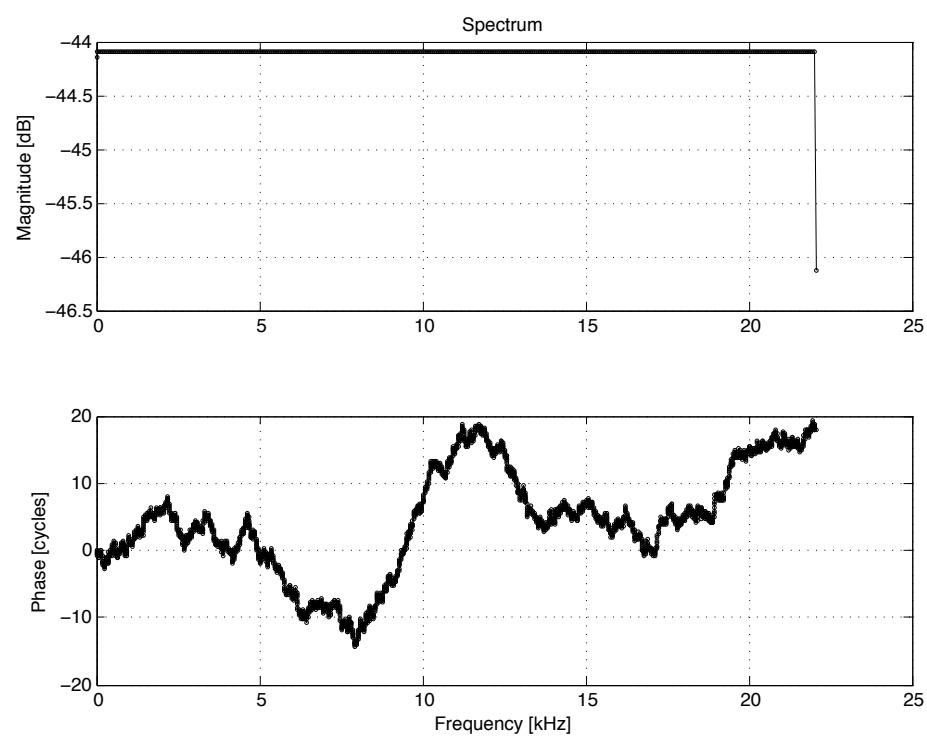
EXspecREP3.m

stimT= 7 - noise (Gaussian distribution)

Time domain



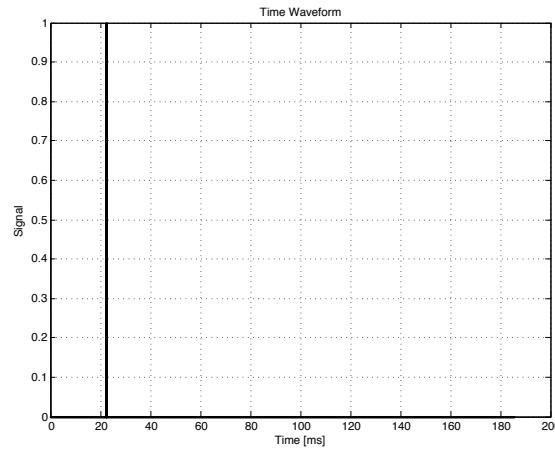
Spectral domain



- Magnitude is flat just like an impulse (i.e., flat), but the phase is random

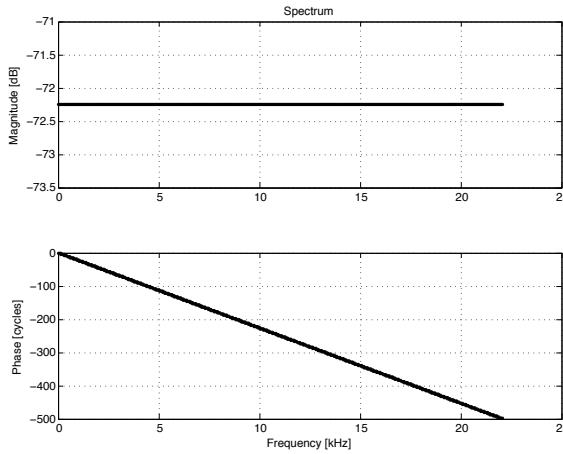
## Fourier transforms of basic (1-D) waveforms

# Impulse

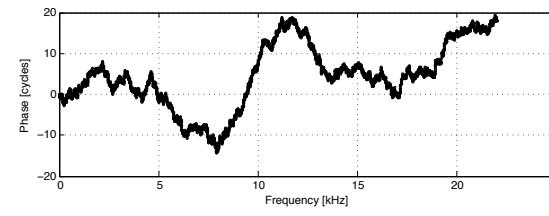
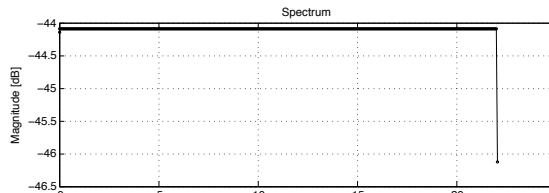
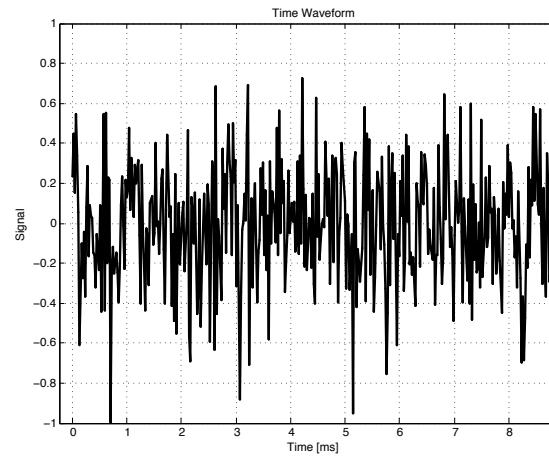


## Time domain

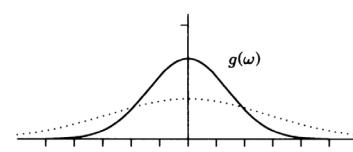
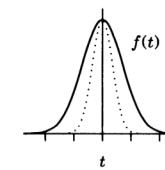
# Spectral domain



# Noise



→ Remarkable that the magnitudes are identical (more or less) between two signals with such different properties. The key difference here is the phase: *Timing is a critical piece of the puzzle!*

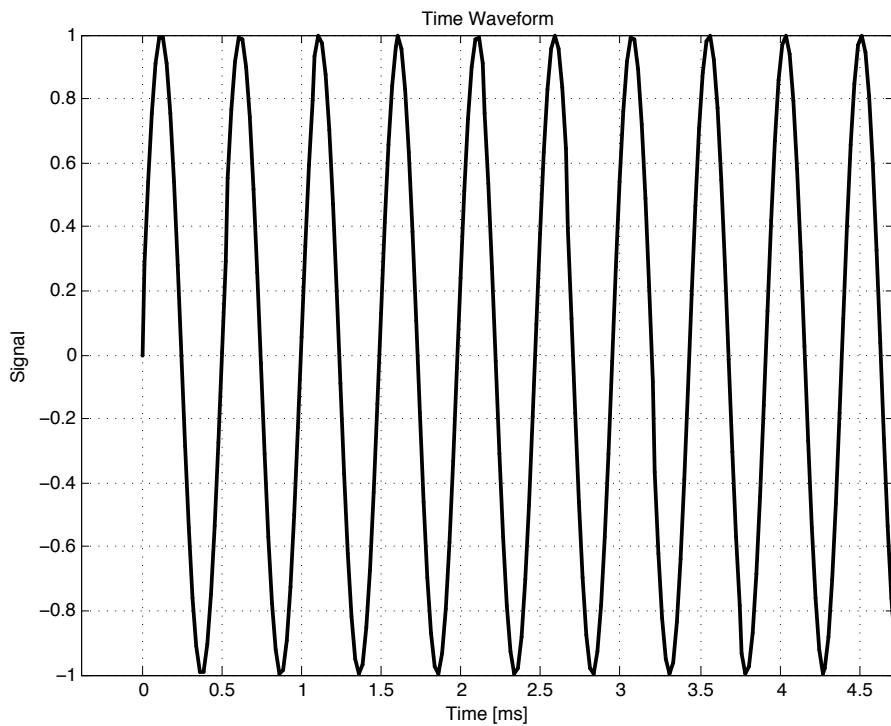


## Fourier transforms of basic (1-D) waveforms

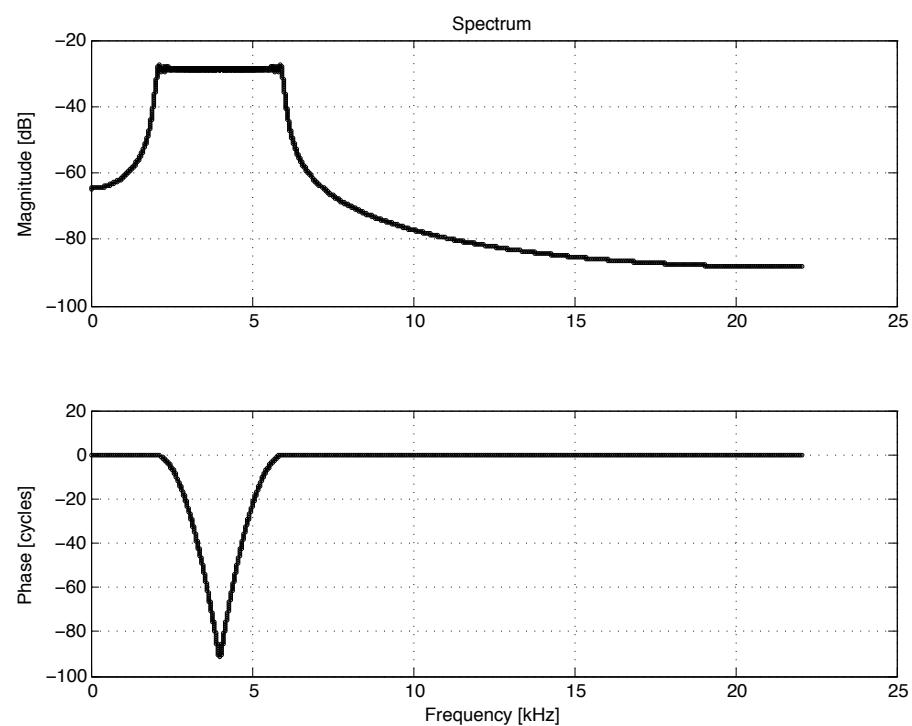
EXspecREP3.m

stimT= 6 - chirp (flat mag.)

Time domain



Spectral domain



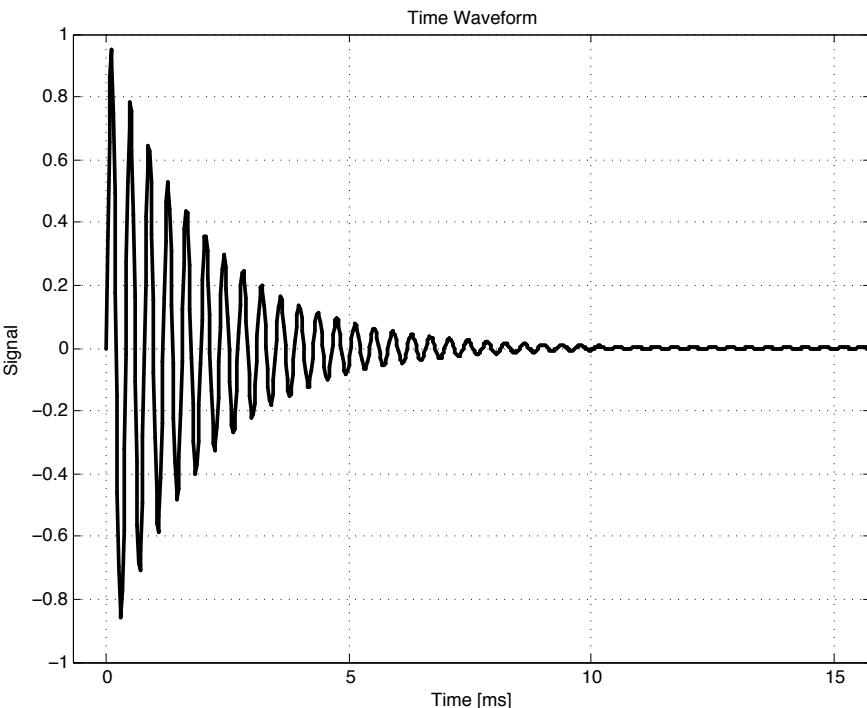
Hard to see on this timescale, but frequency is changing  
(increasing) with time

## Fourier transforms of basic (1-D) waveforms

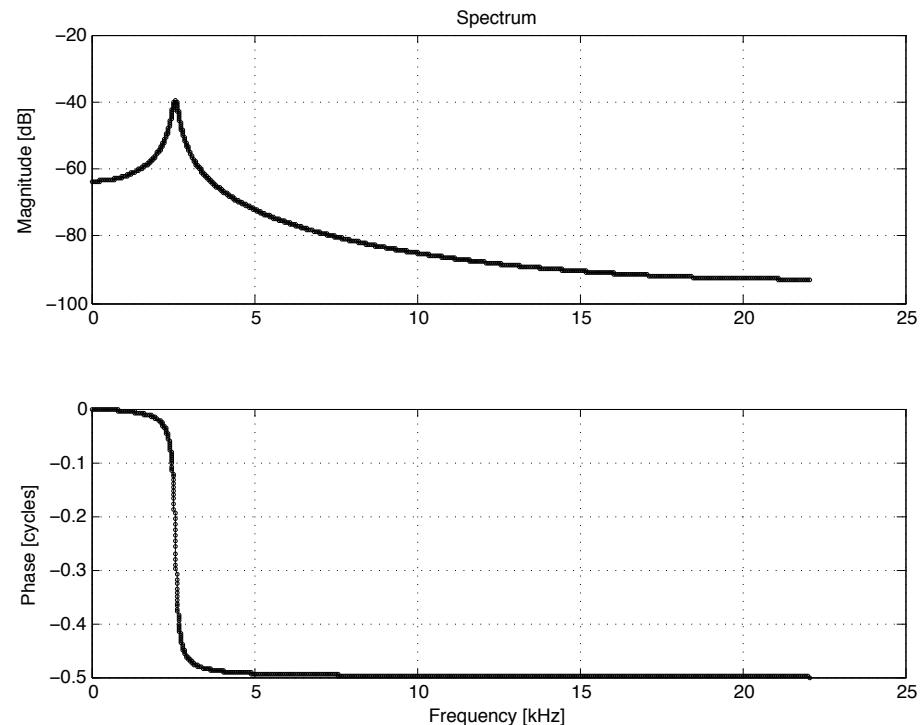
EXspecREP3.m

stimT= 8 – exponentially decaying sinusoid

Time domain



Spectral domain



- This seems to look familiar....