Digital Signal Processing

Class 14 03/05/2025

- Class Overview
 - Frequency Analysis of Discrete Signals
 - Discussion of Lab 2
 - Classification using Frequency Domain Features
- Assignments
 - Reading:
 - Chapter 4: Frequency Analysis of Signals
 - Chapter 5: Frequency-Domain Analysis of LTI Systems

- Exam
 - Take home exam
 - Will post Monday, March 17
 - Due March 24

- Lab 2
 - Classification using Frequency Domain Features
 - Due Mar 22

Questions about HW4?

- Key concept behind the action of LTI systems on signals:
 - Signals can be decomposed into superposition of frequency components
 - Basis function for this decomposition are sines and cosines (and complex exponential)
 - Frequency components of signal are unchanged when passed through Linear Time Invariant systems
 - Only amplitude and phase change

Changes magnitude and phase $H(s) \qquad \qquad M \cdot A \cdot \sin(\omega t + \varphi + \theta)$

- Frequency response of an LTI system
 - The response of an LTI system to any input is:

$$y(n) = \sum_{k=-\infty}^{\infty} h(k)x(n-k)$$

- Considering a complex exponential input $x(n) = Ae^{j\omega n}$

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

$$H(\omega) = \sum_{k=-\infty}^{\infty} h(k)e^{-j\omega k}$$

$$y(n) = AH(\omega)e^{j\omega n}$$

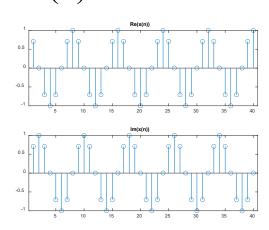
- This shows that complex exponentials are the eigenfunctions and $H(\omega)$ are the eigenvalues of an LTI system.
- Since any signal can be decomposed into complex exponentials, $H(\omega)$ completely characterizes the LTI system.
- Example of how the system modifies the amplitude and phase of a sinusoidal input but not the frequency:
 - Impulse response of system is

$$h(n) = \left(\frac{1}{2}\right)^{n} u(n)$$

$$H(\omega) = \sum_{n = -\infty}^{\infty} h(n)e^{-j\omega n} = \sum_{n = -\infty}^{\infty} \left(\frac{1}{2}\right)^{n} u(n)e^{-j\omega n} = \sum_{k = 0}^{\infty} \left(\frac{1}{2}\right)^{n} e^{-j\omega n} = \sum_{k = 0}^{\infty} \left(\frac{1}{2}e^{-j\omega}\right)^{n} = \frac{1}{1 - \frac{1}{2}e^{-j\omega}}$$

- What does the system do to an complex exponential input
 (i.e. and input at some particular frequency)
 - Consider an input with a frequency of $\pi/4$

$$x(n) = Ae^{j\omega n} = Ae^{jn\pi/4}$$



$$H(\omega) = \frac{1}{1 - \frac{1}{2}e^{-j\omega}}, \quad H(\pi/4) = \frac{1}{1 - \frac{1}{2}e^{-\frac{j\pi}{4}}}$$

$$|H(\pi/4)| = 1.3572, \quad \phi = -28.68^{\circ}$$

• Example in book shows:

$$|H(\pi/2)| = 0.8944$$
 $\phi = -26.6^{\circ}$
 $|H(\pi)| = 0.6667$ $\phi = 0^{\circ}$

- If you want to see the affect of this LTI system on the amplitude and frequency at all frequencies, use Matlab freqz

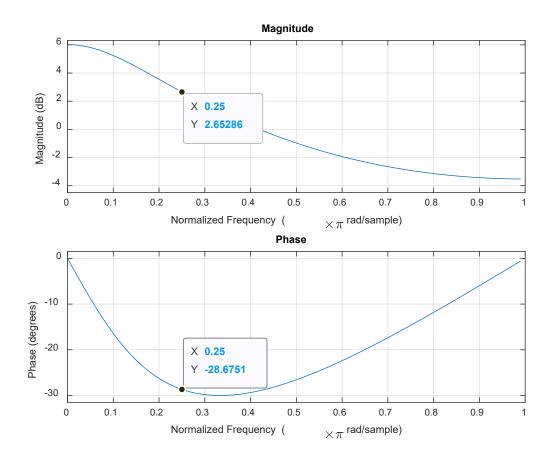
$$H(w) = \frac{1}{1 - \frac{1}{2}e^{-j\omega}}$$

Numerator and denominator coefficients are:

$$b = 1$$

$$a = \begin{bmatrix} 1, & -1/2 \end{bmatrix}$$

freqz(1, [1, -1/2, 100)



$$|H(\pi/4)| = 1.3572, \quad \phi = -28.68^{\circ}$$

 $|H(\pi/4)| = 20\log_{10}(1.3572) = 2.6529$

- If an LTI system changes the magnitude and phase of an input
 - You can begin to see how filtering works
 - Consider what the LTI does to each frequency component

Example of a moving average filter:

$$y(n) = \frac{1}{M+1} \sum_{k=1}^{M} x(n-k)$$

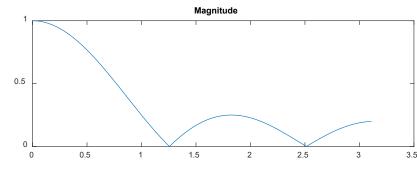
Frequency response is (using the finite geometric series sum)

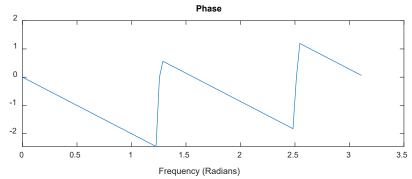
$$H(\omega) = \frac{1}{M+1} \sum_{k=0}^{M} e^{-j\omega k} = \frac{1}{M+1} \frac{1 - e^{-j\omega(M+1)}}{1 - e^{-j\omega}}$$

$$H(\omega) = \frac{1}{M+1} \frac{\sin(\omega(M+1/2))}{\sin(\omega/2)} e^{-j\omega/2}$$

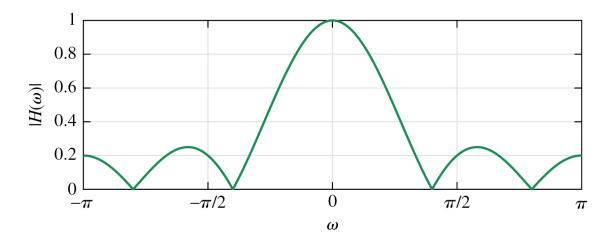
• Moving average (M=4):

$$H(z) = \frac{1}{M+1} \sum_{k=0}^{M} e^{-j\omega k} = \frac{1}{5} \left[1 + z^{-1} + z^{-2} + z^{-3} + z^{-4} \right]$$

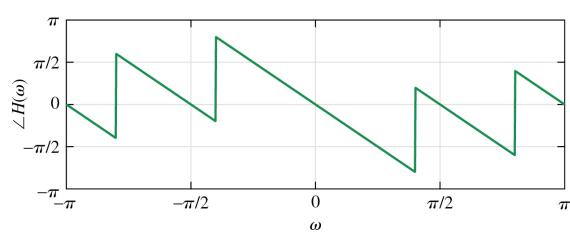




```
a = 5;
b = [1, 1, 1, 1, 1];
[h,w] = freqz(b,a,100);
mag = abs(h);
phase = angle(h);
figure(1)
subplot(2,1,1)
hold off
plot(w, mag)
title('Magnitude');
subplot(2,1,2)
plot(w,phase)
title('Phase');
xlabel('Frequency (Radians)')
```



This is a low-pass filter



M=4

Example with Infinite impulse response

$$y(n) = ay(n-1) + bx(n)$$

We have found the impulse response for this system a few times:

$$H(z) = \frac{b}{1 - az^{-1}}$$

$$H(\omega) = \frac{b}{1 - ae^{-j\omega}}$$

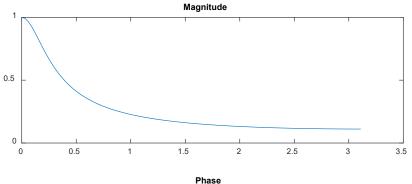
$$|H(\omega)| = \frac{1-a}{\sqrt{1-2a\cos\omega + a^2}}$$

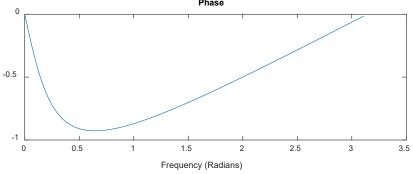
$$\phi = -\tan^{-1}\left(\frac{a\sin\omega}{1 - a\cos\omega}\right)$$

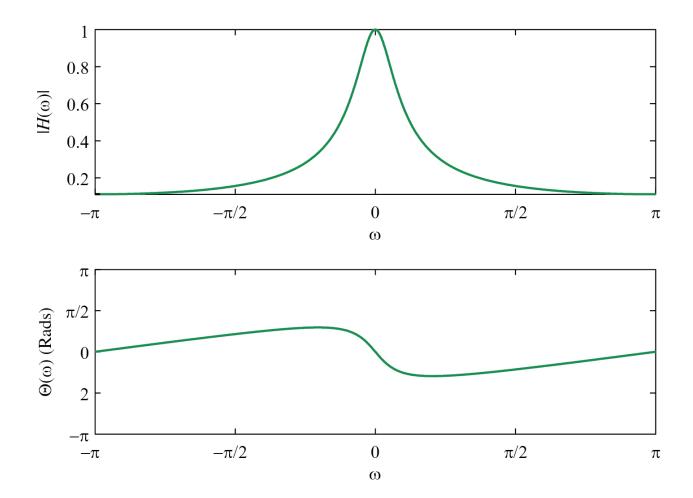
- Example with Infinite impulse response

$$y(n) = ay(n-1) + bx(n)$$

$$H(z) = \frac{1-a}{1-az^{-1}}, \quad a = 0.8$$







This is a also low-pass filter

a = 0.8

- Transient and steady-state response of system
 - Example

$$y(n) = ay(n-1) + x(n), \quad y(-1)$$

$$y(0) = ay(-1) + x(0)$$

$$y(1) = a[ay(-1) + x(0)] + x(1)$$

$$y(2) = a[a[ay(-1) + x(0)] + x(1)] + x(2)$$

$$\vdots$$

$$y(n) = a^{n+1}y(-1) + \sum_{k=0}^{n} a^k x(n-k)$$

- Transient and steady-state response of system
 - If the input is a complex exponential: $x(n) = Ae^{j\omega n}$

$$y(n) = a^{n+1}y(-1) + \sum_{k=0}^{n} a^k A e^{j\omega(n-k)} = a^{n+1}y(-1) + A \left[\sum_{k=0}^{n} a^k e^{-j\omega k}\right] e^{j\omega n}$$

$$y(n) = a^{n+1}y(-1) + A\left[\sum_{k=0}^{n} \left(ae^{-j\omega}\right)^{k}\right]e^{j\omega n}$$

Using sum of finite geometric series

$$y(n) = a^{n+1}y(-1) + A \left[\frac{1 - \left(ae^{-j\omega}\right)^{n+1}}{1 - ae^{-j\omega}} \right] e^{j\omega n} = a^{n+1}y(-1) - \frac{Aa^{n+1}e^{-j\omega(n+1)}}{1 - ae^{-j\omega}} e^{j\omega n} + \frac{A}{1 - ae^{-j\omega}} e^{j\omega n}$$
Steady-state

These die off as n increases

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- Steady-state for periodic input
 - Discrete Fourier Series of input:

$$x(n) = \sum_{k=0}^{N-1} c_k e^{j2\pi k/N}$$

Output of each harmonic gets modified by: $H\left(\frac{2\pi k}{N}\right)$

$$y(n) = \sum_{k=0}^{N-1} c_k H\left(\frac{2\pi k}{N}\right) e^{j2\pi k/N}$$

so also periodic with modified Fourier Series coefficients

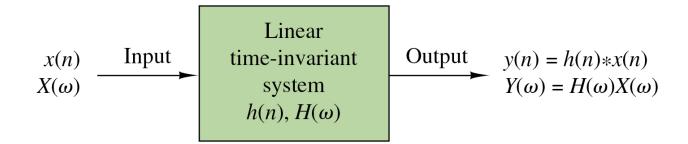
- Steady-state for aperiodic input
 - Use convolution to find output:

$$Y(\omega) = H(\omega)X(\omega)$$
$$|Y(\omega)| = |H(\omega)||X(\omega)|$$
$$\angle Y(\omega) = \angle H(\omega) + \angle X(\omega)$$

Energy Density:

$$|Y(\omega)|^2 = |H(\omega)|^2 |X(\omega)|^2$$

$$S_{yy} = |H(\omega)|^2 S_{xx}$$



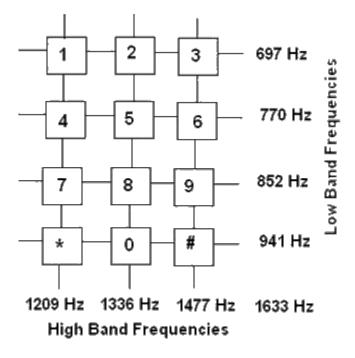
Lab 2 -

- Classification using Frequency Domain Features
 - -Part A: Decoding phone numbers from touchtone frequencies
 - -Part B: Classification of simple words from audio

- Phone tones
 - Tools to use:
 - Fourier transform to see if you can identify discrete frequency that can be associated with numbers on the "dial"
 - Segment tones for numbers "dialed" and try to map to numbers



• Phone tones

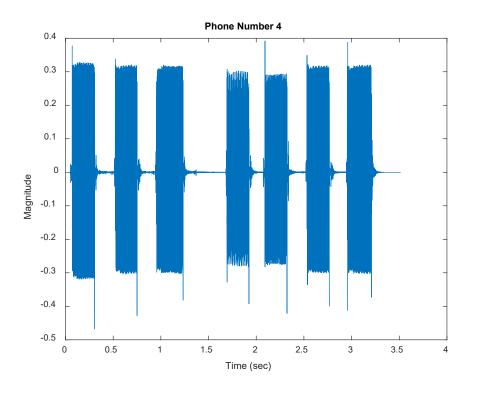


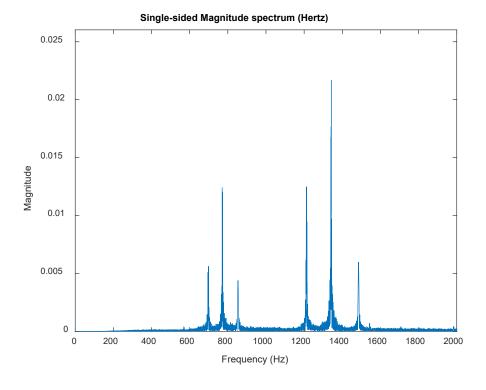
- Part A: Decoding phone numbers from touchtone frequencies
 - Code to read files and plot time domain signals:

```
phone number = 'phone number 4.mp3';
[phn,fs] = audioread(phone number);
phone_part1 = split(phone number,'.');
phone call = split(phone part1{1},' ');
callnum = phone call{3};
phn1 = phn(:,1);
nsamp = length(phn1);
tm = (1/fs) * [1:nsamp];
figure(1)
plot(tm,phn1);
xlabel('Time (sec)')
ylabel('Magnitude')
title(['Phone Number ',callnum])
```

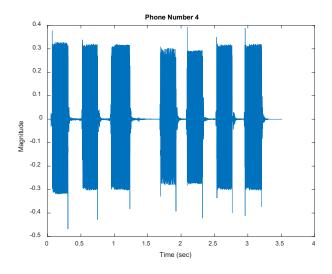
• Code to plot magnitude of Fourier transform

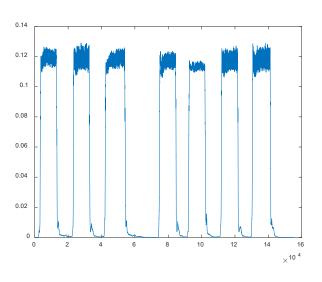
```
fnyquist = fs/2;
x mag = abs(fft(phn1))/nsamp;
bins = [0:nsamp-1];
freq hz = bins*fs/nsamp;
% Plot only positive frequencies
n = ceil(nsamp/2);
figure (2)
plot(freq hz(1:n 2), x mag(1:n 2))
max mag = max(x mag(1:n 2));
axis([0,2000,0,1.2*max mag]);
xlabel('Frequency (Hz)')
ylabel('Magnitude');
title('Single-sided Magnitude spectrum (Hertz)');
```



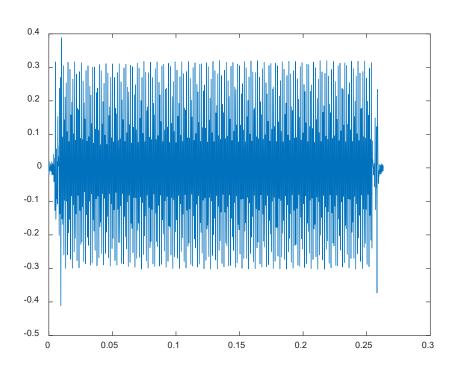


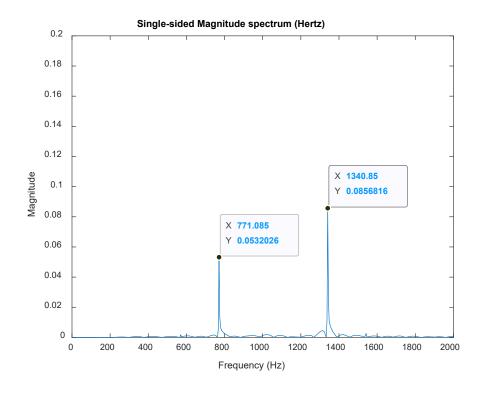
- How to partition key presses?
 - Ideas?
 - Moving average: movmean
 - Moving standard deviation: movstd
 - Moving variance: movvar





• Once you've obtained segments, find peaks



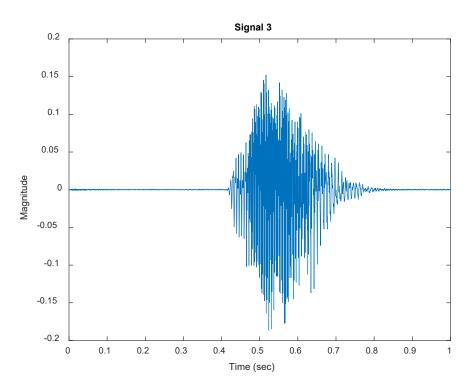


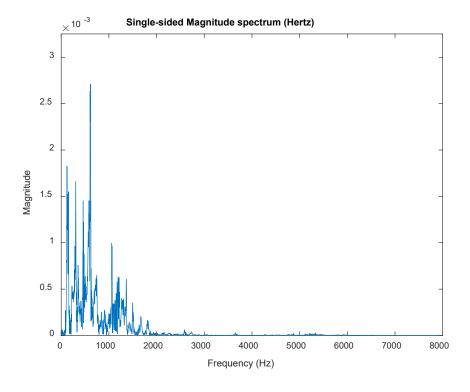
#5: 770 Hz & 1336 Hz

Speech recognition

- There are some very sophisticated methods of speech recognition, which actually seem to work some time.
- We won't be using these.
- Dataset with single words from google.
- We will just try to distinguish two words, like "yes" and "no"
- This will involve obtaining attributes in the frequency domain and using them in a classifier
 - A very neat tool in Matlab called classificationLearner
- A good start for features is finding the power in some set of frequency bands
 - Need to normalize by total power

• Read data:





• Get features:

- You could try looking at spectra for a few yes's and a few no's to see if anything jumps out.
 - Perhaps a frequency band that is strong in one and weak in another
 - Try finding energy in set of frequency bands
 - How many?
 - What are the boundaries?
 - Normalize by total power?
 - Take ratios of bands that look most different?
 - Do some on-line research for good features in speech recognition

• How to use classificationLearner