

# ESE 150 – Lab 05: PsychoAcoustics

## LAB 05

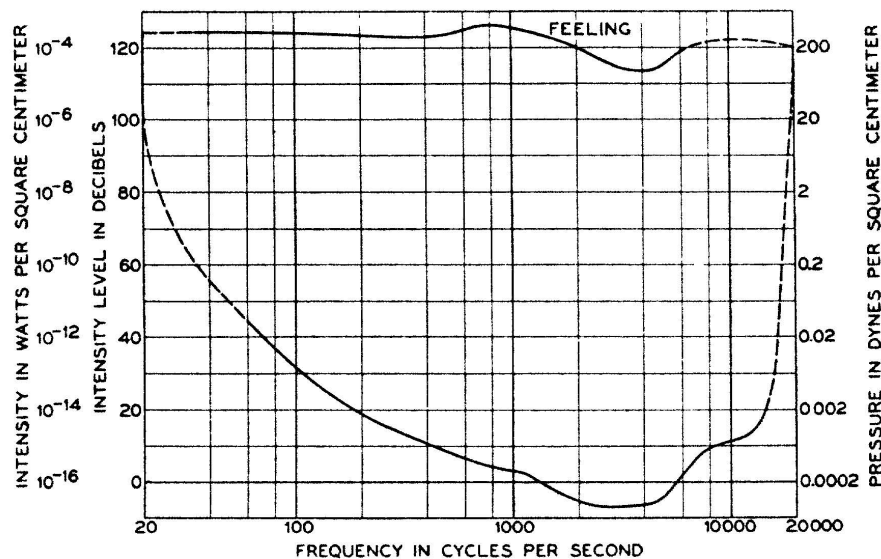
In this lab we will do the following:

1. Examine and characterize hearing thresholds in terms of frequency and loudness
2. Investigate frequency masking and critical bands
3. Investigate Off-frequency masking

### Background:

In lecture we discussed Psychoacoustics, which is a branch of psychophysics that focuses on the study of the psychological and physiological responses associated with sound. In other words, it is a study of the way our brains interpret sound. This branch of science is used heavily in the MP3 algorithm to implement a lossy compression algorithm that makes MP3 compression ratios reach nearly 12:1!

In lab today you'll experiment with some sound files to investigate several interesting psychoacoustic effects. You'll work with your lab partner to see how your brain interprets sound. The first experiment you'll perform is seeing how your brain interprets the loudness of sounds at different frequencies. Recall from lecture the "Fletcher Curve":



This curve above represents the way most humans interpret the loudness of different frequencies vs how "loud" or how large the amplitude of the signals they hear really were. The two biggest things to notice are that we have to make the amplitude of low frequency and high frequency signals VERY high, in order for us to feel they are at the same volume as frequencies in the range of human conversation.

In the first section of lab today, you'll measure and generate a fletcher curve for yourself. In the later sections of lab, you'll investigate frequency masking. This is the situation when we have a loud signal blocking out our ability to hear other sounds in similar frequencies.

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For both of these, you'll want to think about the critical bands in the Cochlea:

Number	Center Freq. (Hz)	Cut-off Freq. (Hz)	Bandwidth (Hz)
		20	
1	50	100	80
2	150	200	100
3	250	300	100
4	350	400	100
5	450	510	110
6	570	630	120
7	700	770	140
8	840	920	150
9	1000	1080	160
10	1170	1270	190
11	1370	1480	210
12	1600	1720	240
13	1850	2000	280
14	2150	2320	320
15	2500	2700	380
16	2900	3150	450
17	3400	3700	550
18	4000	4400	700
19	4800	5300	900
20	5800	6400	1100
21	7000	7700	1300
22	8500	9500	1800
23	10500	12000	2500
24	13500	15500	3500

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### **Prelab:**

1. Read lab.
2. Where does the critical band centered at 1000Hz begin and end?
3. Given strong signals at 1200Hz and 1700Hz, which of the following frequencies are likely to be masked, and why?
  - a. 1070Hz
  - b. 1260Hz
  - c. 1400Hz
  - d. 1620Hz
  - e. 1760Hz
  - f. 4120Hz
4. Write MATLAB code to read the first 2048 samples of a song, convert them to frequencies, and identify the strongest frequency (largest magnitude) in each of the 24 critical bands.
  - a. Download lab5\_song.wav from the course syllabus.
  - b. In MATLAB's command window type:

```
[samples_time, samp_rate]=audioread('lab5_song.wav')
```
  - c. This has imported the PCM quality .WAV file into a matrix called: samples\_time
  - d. It has also determined the sampling rate with which the .WAV file was created.
  - e. Extract 2048 samples starting 60 seconds into the song. (Hint: How do you tell MATLAB to extract a range of values from a vector?)
  - f. Use the provided a plot\_dft function (similar to the one you wrote in Lab 4) to convert the samples to frequencies.

```
function [freq_half, samples_freq, freq_mag] = plot_dft(samples_time,
    samp_rate, figure_num)

samples_freq = fft(samples_time);
samples = length(samples_time); % # of samples
samples_freq_new = abs(samples_freq / samples);
samples_freq_new = samples_freq_new(1:round(length(samples_freq) /
    2.0), :) * 2;
freqs = (0:(samples - 1));
freqs = freqs * (samp_rate/(samples - 1));
freq_half = freqs(1:round(length(freqs) / 2, 0));
figure(figure_num);
freq_mag = samples_freq_new;
stem(freq_half, samples_freq_new);
title('Frequency Domain');
xlabel('Frequency (Hertz)');
ylabel('Amplitude (Volts)');
end
```

- i. Note that this plot\_dft is returning an array of frequencies (freq\_half) to go with the magnitude of each frequency (freq\_mag).

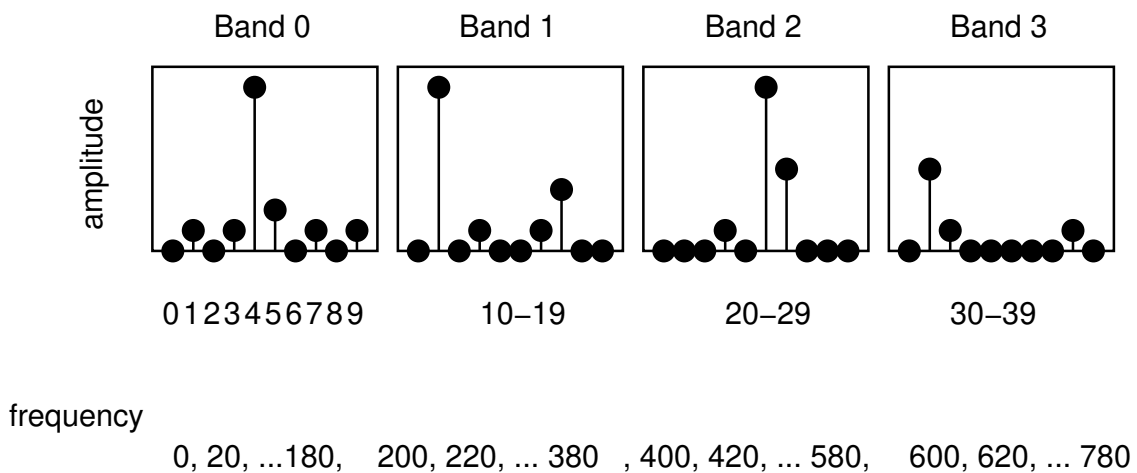
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- ii. Note that these frequencies range from 0 up to half the sample frequency (samp\_rate) and are evenly distributed over the range.
  
- g. Write a MATLAB function:
 

```
function [critical_band_max] = max_freq_per_band(samples_freq,freq_half)
```

 The return value should be a vector of 24 frequencies (in Hz), one for each critical band.
  - i. Each frequency returned is the frequency in the associated band that has the highest amplitude in the band.
  - ii. Function should only take about 10 lines of Matlab code
  - iii. Hint: Define two vectors of 24x1, one for start\_band and one for stop\_band, and assign the cut-off band frequencies to the vector entries. Use a for loop to iterate over each of the bands (i.e. 24 times) and finds the maximum frequency in the band.
  - iv. Hint: find() Matlab function will likely be helpful:  
<https://www.mathworks.com/help/matlab/ref/find.html>
  - v. Include your MATLAB function in your report.
  - vi. Save your code. You will need to perform an operation much like this in Lab 6.
  
- h. Use your function to report the strongest frequencies and include the frequencies in your report.

To illustrate, here's a smaller example. This has only 40 frequencies between 0 and 780 divided into 4 bands.



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samples\_freq will be a 40 element vector with a value:

```
[0,0.1,0,0.1,0.95,0.2,0,0.1,0,0.1,0,0.95,0,0.1,0,0,0.1,0.3,0,0,0,0,0,0.1,0,0.95,0.4,0,0,0,0,0.4,0.1,0,0,0,0,0,0.1,0]
```

freq\_half also has 40 elements=[0,20,40,...,780]

For this simple case, there are 4 bands, so the start\_band and stop\_band are only 4x1.

```
start_band=[0,200,400,600]
```

```
stop_band=[199,399,599,799]
```

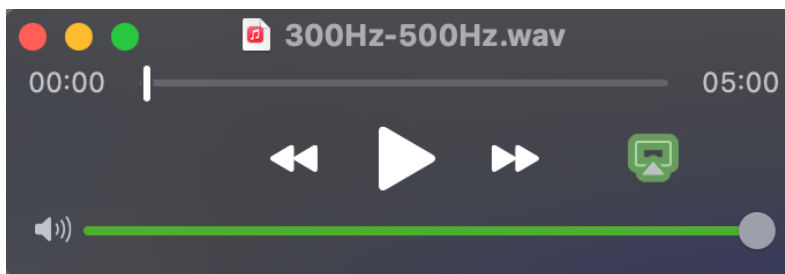
The return value (critical\_band\_max) for this example is 4x1 = [80,220,500,620]

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### Lab Procedure:

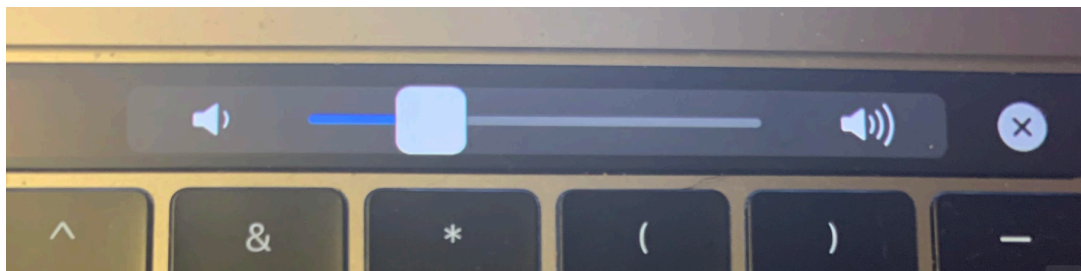
#### Lab – Section 1: Measuring Hearing Thresholds

- In this section you'll generate your own Fletcher Curve
1. Download the first set of tones from the Course Webpage.
  2. When opening the .wav files use an application where you can see the time as the tone is playing (e.g. QuickTime Player for Mac users, Groove Music for Windows). **Don't use Audacity or any other audio editing/recording software. Only use an audio player application. This is very important for running calculations.**



Caption

3. Plug in your headphones before listening to the files. **Note: Make sure you set your computer's sound output very low as these tones will get loud. Don't skip this step.** Once you set the sound output, do not change it for the rest of this section. Here's a reference.



Caption

4. Start with the 20Hz.wav file
  - a. Press play and listen to the file.
  - b. It's likely you won't be able to hear it. Why is that the case?
  - c. Repeat with the 50Hz.wav file. Can you hear anything?
5. Now move on to the remaining files - you should definitely be able to hear these.
  - a. Open the 100Hz file. Listen to it up until the halfway point and keep track of how loud it is

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to you (This is very subjective). Use this perceived loudness as the baseline for the subsequent files.

b. For the remaining files, press play and pause as soon as you perceive the tone to be of equal loudness to the previous tone.

c. Take note of the time and use the following equation to calculate the amplitude of the frequency:

Amplitude = time (at pause)/ total time of audio file

All the files are 30 seconds long, so the denominator should always be 30.

d. Keep a record of the frequency and amplitude in a spreadsheet—you'll be using these data points to graph your Fletcher curve. (The 100Hz should be paired with an amplitude of 0.5)

e. If you can't hear a tone even after the audio file has finished playing, record that frequency in your spreadsheet with an amplitude of 1 (**This will most likely apply to the two tones from step 4 in addition to very high frequencies (Hint: you most likely won't be able to hear the 22kHz frequency)**).

f. Before plotting the points, convert the amplitudes into decibels using the following equation: # of dB =  $(-54 * (1 - \text{Amplitude})) + 90$

6. When you've gone through all of the files, plot your data points (Frequency on the x-axis and Amplitude on the y-axis) and show a TA your Fletcher curve. (Include this plot in your writeup)

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## Lab – Section 2: Frequency Masking and Critical Bands

- In this section you'll investigate **frequency masking**

### 1. Masking Demo:

- In this section, you'll see the effects of on-frequency masking
  - i. Download the maskingDemo.wav file from the CourseWebpage.
  - ii. Open it in Audacity and play the tone.
  - iii. Notice the second signal “masks” the first as it is at the same exact frequency and it is much louder! Your ear cannot differentiate between the two signals in the duration of time that they're playing simultaneously.
  - iv. **Take a screenshot of the two signals in Audacity to include in your writeup.**

### 2. Simultaneous Masking – On Frequency Masking

- Simultaneous masking occurs when a sound is made inaudible by a noise or unwanted sound of the same duration as the original sound.
  - i. There are four basic parts to simultaneous masking:
    1. Critical Bandwidth, similar frequencies, lower frequencies, and the effects of intensity.
  - ii. This is called “on-frequency masking”
    1. This is used heavily in the MP3 algorithm
- Recall from lecture the 24 Critical bands of the Cochlea (in background section).
- We think of the 24 bands as a bank of auditory filters that our ear has.
- If a “masker” signal is loud enough it can mask other frequencies that are in the *same* auditory filter bank and make it so you cannot hear them.
  - ii. This is called “on-frequency masking”
    1. This is used heavily in the MP3 algorithm
- In this section you'll calculate the amplitude of a masker signal once it's finally able to mask another signal
  - a. Download the second set of tones from the Course Webpage
    - i. Open the 1000-1005Hz.wav file in an application where you can see the time as the tone is playing (e.g. QuickTime Player for Mac users, Groove Music for



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Windows). **Don't use Audacity or any other audio editing/recording software. Only use an audio player application. This is very important for running calculations.**

- ii. Press play on the file. As soon as you're able to hear a singular pure tone, press pause. Take note of the time and use the following equation to calculate the amplitude of the frequency:  
Amplitude = time (at pause)/ total time of audio file  
All the files are 30 seconds long, so the denominator should always be 30.
- iii. Now repeat with the remaining files.
- iv. Keep a record of the frequency and amplitude in a spreadsheet. You'll be plotting these points to see how masking changes with distance from frequency. We're using 1000Hz as the center frequency, so it's the second frequency in the title of the file that you'll be plotting. (Ex: For the 1000-1005Hz.wav file, use 1005Hz as the frequency to plot).
- v. Before plotting the points, convert the amplitudes into decibels using the following equation: # of dB = (-54 \* (1 - Amplitude)) + 90
- vi. **Include this plot in your writeup.**

b. Next, you'll calculate the amplitude of a maskee signal once it's finally been masked by another signal.

- i. Download the third set of tones from the Course Webpage.
- ii. Open the 701-725Hz.wav file in an application where you can see the time as the tone is playing (e.g. QuickTime Player for Mac users, Groove Music for Windows). **Don't use Audacity or any other audio editing/recording software. Only use an audio player application. This is very important for running calculations.**
- iii. Press play on the file. As soon as you're able to hear a singular pure tone, press pause. Take note of the time and use the following equation to calculate the amplitude of the frequency:  
Amplitude = 1 - (time (at pause)/ total time of audio file)  
**(Be careful: This equation is slightly different from that of part a)**  
All the files are 30 seconds long, so the denominator should always be 30.
- vii. **Record this amplitude along with its associated file and include in your writeup.**

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viii. Now repeat with the 760-710Hz.wav and 300-1000Hz.wav files.

iv. Your data is largely subjective, but that's the way auditory experiments go!

### 3. Simultaneous Masking – Frequency Resolution

- Another interesting auditory experiment is to determine one's ability to discern between two similar frequencies.
- If two sounds of two different frequencies are played at the same time, two separate sounds can often be heard rather than a combination tone.
  - i. The ability to hear frequencies separately is known as *frequency resolution* or frequency selectivity.
- But, when signals of two different frequencies are played at the same time and are perceived as a combination tone, they are said to reside in the same critical band.
  - a. Download the fourth set of tones from the Course Webpage.
  - b. Start by playing the first audio file 1000-1001Hz.wav
    - i. Can you differentiate between the two signals? Record Yes or No.
    - ii. What do you actually hear? Describe this.
    - iii. Is what you're hearing a combination tone or two separate tones?
  - c. Repeat for the remaining tones, recording your answers to the above questions for each of them (you'll be submitting these in your writeup).
  - d. Before leaving, call over a TA. For which audio files do you hear separate tones? Can you explain why? This is your exit check off.

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### **Postlab:**

1. A typical siren is not a pure tone but will shift through a large range of frequencies. With what you now know about masking, explain why a pure tone might make a poor siren and how a siren that spans a large frequency range would address the problem.
2. Consider someone with hearing damage such that they cannot hear any tones above 8kHz. How could you create a hearing aid that would allow them to “hear” sounds above that occur between 8kHz and 22kHz. (a similar question, how you could create a “hearing aid” that would allow you to “hear” sounds between 20kHz and 50kHz.)
3. What masking phenomena might you need to consider in your “hearing-aid” solution?

### **HOW TO TURN IN THE LAB**

- Upload a PDF document to canvas containing:
  - Prelab answers (2, 3, 4h)
  - MATLAB from prelab (4g)
  - All images (Fletcher curve, Audacity screenshot, etc)
  - Answers to questions in Section 2.2 and 2.3
  - Explanation of the Fletcher curve (e.g.: observations and what it means)
  - Postlab answers