Why You Hear What You Hear

The Problem Book

Eric J. Heller
Jean-François Charles

The artwork heading the chapters in this problem book was created by Eric J. Heller. These images and others can be found on the website ericjhellergallery.com. All of the images come from some aspect of Heller's research. For the most part, they are high-resolution intended for large format prints. Explanations of the science behind images can be found on the web gallery.

Why You Hear What You Hear: The Problem Book.
First edition. Last updated January 8^{th}, 2013.

If you find any error in the text, thank you for your feedback: jfc@jeanfrancoischarles.com

This document is a combination of problems, short questions, and quiz questions, strongly integrated with *Why You Hear What You Hear*, a book published by Eric J. Heller and Princeton University Press. Although many of the ideas for projects and problems came from various teaching fellows and Prof. Heller over the years of teaching “The physics of music and sound” and now “Why you hear what you hear: the science of music and sound” at Harvard University, the lion’s share of credit for this problem book goes to Jean-François Charles. Hugh Churchill and Kate Jensen were also key among the teaching Fellows contributing to the material. Hugh Churchill was the first contributor for the problems *Tsunami*, *What’s not on you iPod*, *A Hard Day’s missing fundamental?*, *Four Helmholtz resonators*, *Missing fundamental with Max Partials*, and *Smooth & sharp sax*.

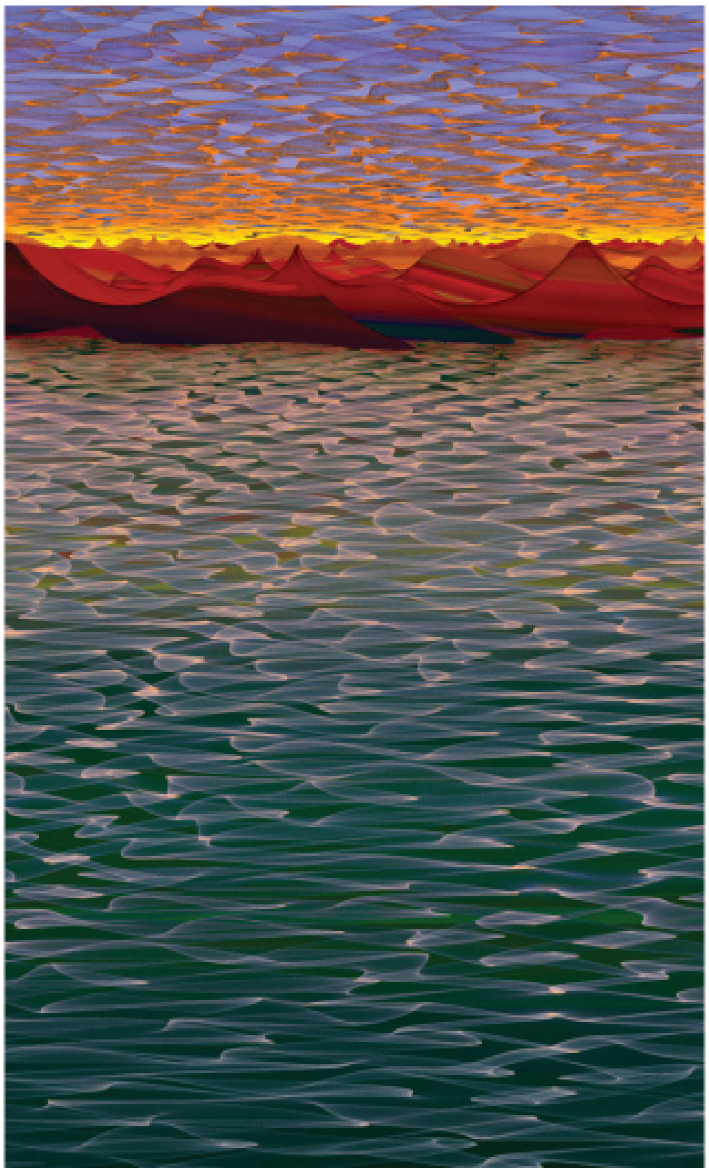
In addition to organizing and setting the format for the book, Jean-François refined, corrected, and greatly enhanced the content. Additional projects can be found scattered among the chapter pages on the books website, whyyouhearwhatyouhear.com; they are easily identifiable with the red heading “Project:” Other projects are in fact commonplace on these pages and in the book, in the form of creative variations on the ideas and manipulations using readily available tools mentioned so many times in the book. Often these tools were used to create the figure seen on the book pages and the pages of the website.

Indeed many of these projects and investigations can be taken anywhere from the level of particle homework set to an entire semester project, by extending the care, tools, and questions surrounding the project.

Answers for selected problems and questions are available for download by instructors from Princeton University Press.

Contents

Sound Itself



1 – Problem: Narrow escape

Read in Why You Hear What You Hear

- Chapter 1: How Sound Propagates
- Chapter 2: Wave Phenomenology

Tools

- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)

What happens when a plane wave hits a narrow hole in a wall?

1.1 On a two-dimensional model of air cells, make one to three drawings to show what happens as the pressure emerges from the gap in the wall. Show how the pressure applies from air cells to air cells.
For instance, your starting point for your drawings could look like this:

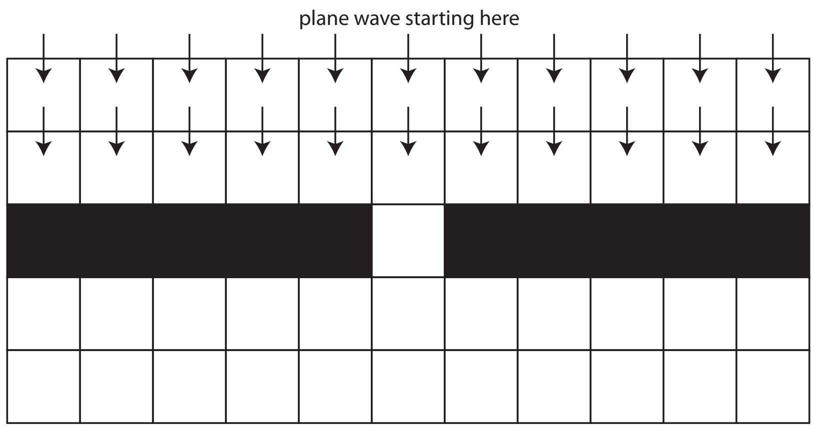


Figure 1: Walls & air cells

1.2 Simulate this in **Ripple**. Take a screenshot.
With white representing background pressure, and the little cells shown here corresponding to pixels in Ripple, the black pixels represent a wall you have drawn, with a one-pixel gap. Load up two rows of pixels with pressure or use a plane source; make sure to turn **Fixed Edges** off, slow the app down as much as possible, and try to catch the excess pressure in the early stages of leaking out.

1.3 Which phenomenon is showed here (select all that apply)?
☐ Interference
☐ Reflection
☐ Refraction
☐ Diffraction

2 – Problem: Measure the speed of sound

Read in Why You Hear What You Hear

- Chapter 2: Wave Phenomenology – Speed of sound

Tools

- **Audacity** (details on http://audacity.sourceforge.net/) (or other sound analysis program, like Amadeus, details on http://www.hairersoft.com/)

In this problem, you are going to make a rough measure of the speed of sound.
Make a recording of a short percussive sound, like a hand clap. You will have to identify the first reflection. Think about ways to get a good recording for this exercise: relative position of the source, microphone (computer), and reflective surface (wall). We suggest you use a set-up similar to the following:

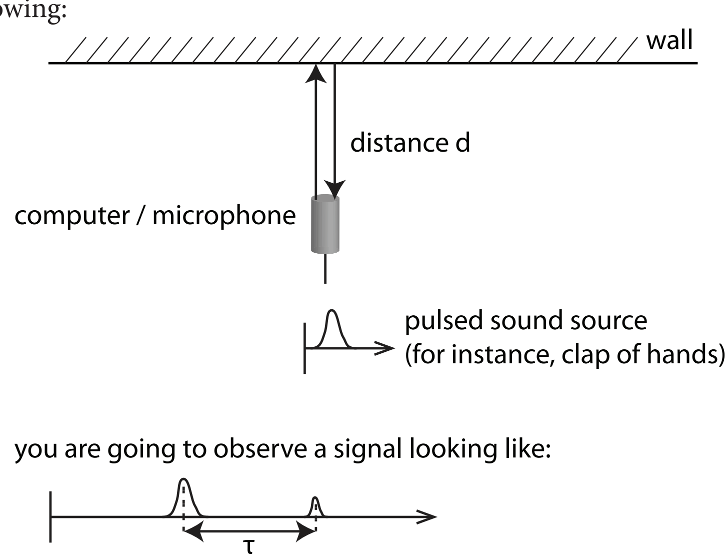


Figure 2: Reflection recording set-up

Take note of the direct distance *d* from your microphone to the reflective surface.
Save your original recording as a **wav** format sound file.
Slow the sound down (in **Audacity**, use the command **Change Speed** in the **Effects** menu) and zoom in on the waveform display to see the first reflection. Try different settings. Find a good slowing down factor that enables you to visualize and hear as clearly as possible the first reflection of the sound. Beware of other closeby objects which may give false, early signals. Save the slow version of your sound in **wav** format, and keep note of the slowing down factor you used.

2.1 Print the waveform of the sound (slowed down). Indicate on the plot the slowing down factor that you used. Label the direct sound and the first reflection. Measure the time delay on the zoomed in waveform display with the cursor.
What is the time delay τ between the direct sound and the reflection? (Don't forget to take into account the slowing down factor you used.)

2.2 Express the speed of sound c as a function of d and τ.

2.3 Using your measures, what speed of sound do you get? Compare to the theoretical speed of sound. Is your result realistic given the uncertainty on the distance measurement? Describe the geometry you used, distances, etc. The tabulated speed depends on temperature. Measure temperature and look up the speed of sound at that temperature in air, and compare your result. Hint: use as flat a reflecting wall surface as you can, and make sure the wall is perpendicular to the line of sight from microphone to wall.

2.4 If you can increase the distance from microphone to wall, would that be likely to increase your accuracy of measurement, and why? Discuss possible reasons for any differences between your results and the tabulated values.

3 – Problem: Tsunami

Read in Why You Hear What You Hear

- Chapter 2: Wave Phenomenology

You may use the units you find the most convenient, such as miles for length, and miles per minute for speed.

Read this excerpt from a warning issued for Hawaii about a tsunami created by the large earthquake in Chile on February 27th (Feb. 26th Hawaiian time, HST):

```
BULLETIN
NWS PACIFIC TSUNAMI WARNING CENTER EWA BEACH HI
1133 AM HST SAT FEB 27 2010

TO - CIVIL DEFENSE IN THE STATE OF HAWAII

SUBJECT - TSUNAMI WARNING SUPPLEMENT

AN EARTHQUAKE HAS OCCURRED WITH THESE PRELIMINARY PARAM-
ETERS
     ORIGIN TIME - 0834 PM HST FRI 26 FEB 2010
     COORDINATES - 36.1 SOUTH   72.6 WEST
     LOCATION    - NEAR COAST OF CENTRAL CHILE
     MAGNITUDE   - 8.8  MOMENT

EVALUATION

A TSUNAMI HAS BEEN GENERATED THAT COULD CAUSE DAMAGE
ALONG COASTLINES OF ALL ISLANDS IN THE STATE OF HAWAII.
URGENT ACTION SHOULD BE TAKEN TO PROTECT LIVES AND PROP-
ERTY.

A TSUNAMI IS A SERIES OF … OCEAN WAVES. EACH INDIVIDUAL
WAVE CREST CAN LAST 5 TO 15 MINUTES OR MORE AND EXTEN-
SIVELY FLOOD COASTAL AREAS. … TSUNAMI WAVES EFFICIENTLY
WRAP AROUND ISLANDS.

ALL SHORES ARE AT RISK NO MATTER WHICH DIRECTION THEY
FACE. THE TROUGH OF A TSUNAMI WAVE MAY TEMPORARILY EX-
POSE THE SEAFLOOR BUT THE AREA WILL QUICKLY FLOOD AGAIN.
EXTREMELY STRONG AND UNUSUAL NEARSHORE CURRENTS CAN AC-
COMPANY A TSUNAMI.

THE ESTIMATED ARRIVAL TIME IN HAWAII OF THE FIRST TSU-
NAMI WAVE IS
     1105 AM HST SAT 27 FEB 2010

MESSAGES WILL BE ISSUED HOURLY OR SOONER AS CONDITIONS
WARRANT.
```

Figure 3: Hawaii tsunami bulletin excerpt

3.1 The warning states that tsunami waves efficiently wrap around islands. What can you say about the wavelength of a tsunami wave relative to the diameter of the islands? Explain your reasoning (with a picture, if helpful).

3.2 The island of Hawaii has a diameter of roughly 100 miles. What must the wavelength of the tsunami wave be, expressed as an inequality?

3.3 Based on the facts given above (including the bulletin), what is the approximate speed of propagation of a tsunami wave?

3.4 Again using the facts given above, about how far is Hawaii from Chile?

3.5 The largest tsunami ever recorded, over 500 meters high*, occurred in Lituya Bay, Alaska when an earthquake caused a massive landslide into the bay. The bay is about 10 miles long and 2 miles wide, with a narrow opening to the sea. Why would these conditions lead to an unusually large tsunami? According to one survivor on a boat that rode out the quake, the wave may in the bay have been traveling up to 600 mph.
* 520 meters - 1720 feet - was the "runup" onto the steep shoreline walls. The wave may have been 200 feet high in the center of the bay, it is estimated.

4 – Problem: Loudspeaker phase

Read in Why You Hear What You Hear

- Chapter 2: Wave Phenomenology – Interference and Superposition (enough to get us started for this problem)
- Chapter 7: Sources of sound (also relevant)

Tools

- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)

Let's consider two distant speakers fed with the same sinusoidal source. When a listener walks around in the space, the amplitude of the sound heard varies greatly. You are going to replicate this experiment using **Harvard Falstad Ripple applet**.
Setting up: you can set the simulation speed higher or lower to make some things easier to measure, but leave it fixed after that. Some experimentation to optimize your conditions may be needed.

✓ Check **Stopped**
✓ Select **Clear waves** under **Actions…**
✓ Uncheck **Fixed Edges**
✓ Set **Setup: Dipole Source** (menu **Setup** at the top)
✓ Set 2 probes
✓ Drawing mode **normal**
✓ Set the resolution to about 200
✓ Set **Mouse = Draw Walls**, and draw walls as required. It's usually easier to draw closed shapes and open them up with **erase walls** later.
✓ Drag probes around to where to want them.
✓ When ready, unchecked **Stopped**
✓ Check **Stopped** to take screen shots.

Create a "loudspeaker" by drawing a box, setting a sound source inside of it, and making a small hole in the box for the sound to get out. Now, make a pair of loudspeakers oriented the same way. You are going to demonstrate the effects of changing the distance between the speakers, and/or the frequency of the sound, using sound of various wavelengths, from large to about 1/4 of the spacing between the speakers.

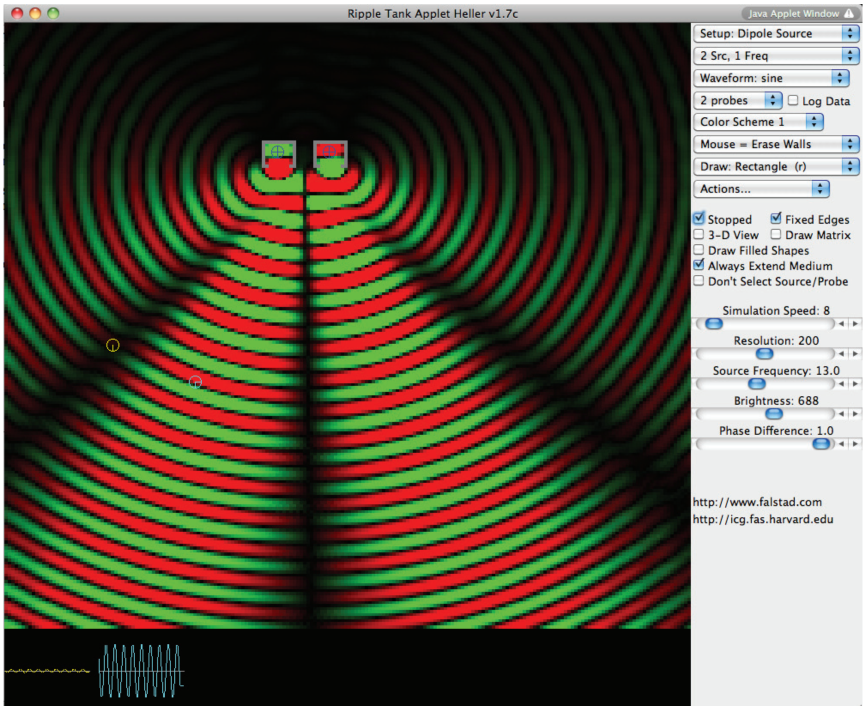


Figure 4: Two loudspeakers in **Ripple**, here out-of-phase

4.1 Set the phase to be perfectly in-phase (phase slider at zero) or perfectly out-of-phase (phase slider all the way to the right). Put the speakers at different distances. Try a wide range of frequencies, and comment on the effect of that. Can you get the sound from the two speakers to cancel out in places? Write up a summary of your conclusions about what happens, including some screenshots. Include the concepts of constructive and destructive interference.

4.2 Find another experiment you like and set up your own system to investigate. Make a screenshot and explain what you discovered. Be creative! You might want to try out variations of some of the things you have seen in *Why You Hear What You Hear*.

5 – Problem: Horn simulation

Read in Why You Hear What You Hear

- Chapter 3: Sources of Sound

Tools

- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)

A reasonable conical horn can be made by rolling up a piece of stiff cardboard, leaving approximately a 2-inch hole at the throat of the horn, and perhaps a seven or 10-inch mouth. The effect on the voice or other sound sources injected into the throat of the horn is significant. Try it! Here, you will do a numerical simulation of the effects of adding a horn to a source. Compare two identical sources, one placed inside the throat of the horn, and one placed far outside the horn in an open region. In the figure below, probes "A" and "B" from sources at "a" and "b" were compared.

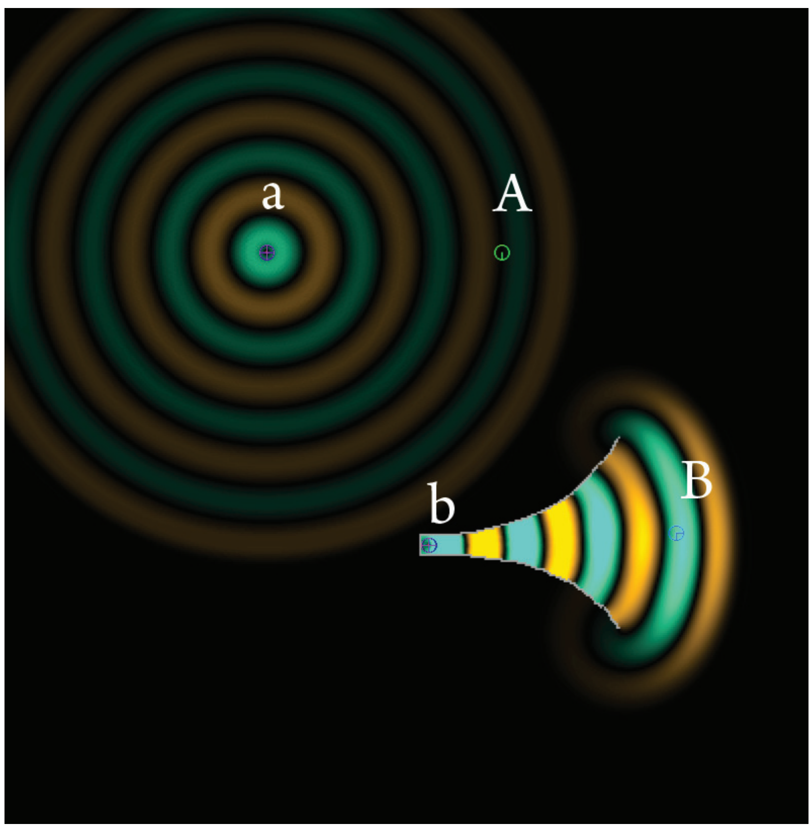


Figure 5: Simulation of a horn in **Ripple**.

Setting up: before drawing the horn in Ripple, we recommend to set the Resolution to the maximum: 400 pixels.

✓ Check **Stopped**
✓ Select **Clear waves** under **Actions…**
✓ Uncheck **Fixed Edges**

- ✓ Set **Setup: Dipole Source** (menu **Setup** at the top)
- ✓ Set 2 probes
- ✓ Drawing mode **normal**
- ✓ Set the resolution to the maximum, 400 pixels
- ✓ Set **Mouse = Draw Walls**, and draw the horn walls as required. For instance, use the ellipse tool, then set **Mouse = Erase Walls** to erase the parts not needed for the horn. Seal off the horn tube at the left with a vertical line.
- ✓ Drag probes around to where to want them.
- ✓ When ready, unchecked **Stopped**
- ✓ Check **Stopped** to take screen shots.

5.1 Log data for the two probes A and B (you might log data only after signal reaches the probes). Compare the amplitude of the signals: for instance plot both signals using **Microsoft Excel**. Write a short report after your experiments. Include a screen shot and any plot that you find relevant. Is the overall power increased when using a horn? (Estimate the angle subtended by the beam exiting the horn, and use that and the relative amplitudes to estimate the total power. You might want to experiment with differently shaped chambers for confining the source to see if you can increase the power.)

absolute time	relative time (here, time minus 2523)	probe 1 (no horn)	probe 2 (with horn)
2523	0	0.0295	0.3415
2525	2	0.0104	0.3292
2527	4	-0.004	0.3033
2529	6	-0.0194	0.255
2531	8	-0.0365	0.1946
2533	10	-0.0501	0.1385
2535	12	-0.0615	0.0792
2537	14	-0.0683	0.0066
2539	16	-0.0722	-0.0656
2541	18	-0.0742	-0.1335
2543	20	-0.0692	-0.2012
2545	22	-0.0618	-0.257
2547	24	-0.0555	-0.2933
2549	26	-0.0438	-0.3189
2551	28	-0.03	-0.3432
2553	30	-0.014	-0.3477
2555	32	0.0046	-0.3197
2557	34	0.0203	-0.2806
2559	36	0.0357	-0.2323
2561	38	0.0507	-0.1689
2563	40	0.0633	-0.1107
2565	42	0.074	-0.0547
2567	44	0.0809	0.0232
2569	46	0.0827	0.1052
2571	48	0.0803	0.1674
2573	50	0.0742	0.2267
2575	52	0.0667	0.282
2577	54	0.0593	0.3127
2579	56	0.0457	0.3278
2581	58	0.0284	0.3436

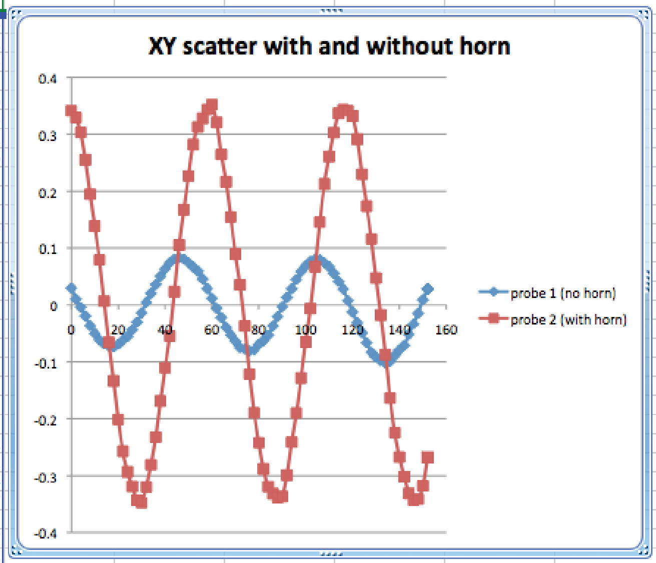


Figure 6: Sound amplitude the same distance from a source vs. time, with and without a horn.

6 – Short questions

Wind turbine

6.1 The Vestas V164-8.0 MW offshore wind turbine (see http://www.vestas.com) has a rotor diameter of 164 meters (538 feet). How many revolutions per minute must the blades be rotating in order for their tips to go supersonic (travel faster than the speed of sound)?

Noisy machine

We have a noisy piece of machinery, and measure a sound intensity of 90 dB.

6.2 What will we measure if we turn on a second identical piece of machinery, situated at the same distance of our measure tool?

6.3 With only the first machine running, what measure will we get if we double the distance between the machine and our measure instrument?

Golden sound

In Switzerland, a periodic sound with a wavelength of one foot in air enters a block of gold in which the speed of sound is 3240 m/s.

6.4 What is the frequency of the sound?

6.5 Is the wavelength different inside the block of gold? If yes, what's the new wavelength?

Sinus sickness

Tools

- **Mathematica** or other tool to plot functions, such as http://www.fooplot.com or the Java plotter http://math-it.org/Mathematik/Analysis/FunctionPlotter.html

Make sure to clearly caption each plot with the question number.

6.6 Make a plot of $y(t) = \sin(2\pi f t)$ from $t = 0$ (or a little less) to $t = 0.01$ s (or a little more), with $f = 250$ Hz.
✓ Label 0 and 0.01 s on the time axis.
✓ What is the frequency?
✓ What is the period?

6.7 Make a plot of $y(t) = \sin(2\pi f t) + \frac{1}{2}\cos\left(4\pi f t + \frac{\pi}{6}\right)$ from $t = 0$ (or a little less) to $t = 0.01$ s (or a little more), with $f = 200$ Hz.
✓ Label 0 and 0.01 s on the time axis.
✓ What is the frequency?
✓ What is the period?

6.8 Make a power spectrum plot of the function $y(t) = \sin(2\pi f t) + \frac{1}{2}\cos\left(4\pi f t + \frac{\pi}{6}\right)$ (same as in question 6.7).
Remember the power spectrum is not sensitive to phase; the power spectrum of

$y(t) = \sin(2\pi f t)$ is the same as the one of $y(t) = \sin\left(2\pi f t + \frac{\pi}{2}\right) = \cos(2\pi f t)$.

6.9 For each of the following signals S (t), say if it is periodic, and if yes, give the period.

- $S(t) = \sin(2\pi f t) + \sqrt{2}\ \sin(7.2 \times 2\pi f t)$
- $S(t) = \cos\left(2\pi f \sqrt{t}\right)$
- $S(t) = \cos\left(2\pi f t^2\right) + \cos\left(2 \times 2\pi f t^2\right)$
- $S(t) = \cos(2\pi f t) + \sqrt{3}\ \cos(\pi\, 2\pi f t)$
- $S(t) = \cos(2\pi f t) + \sqrt{3}\ \cos\left(3^2\, 2\pi f t\right)$

6.10 Plot the following signals over one period (if periodic) or over several oscillations if not and make power spectrum plots (by hand is fine) of the following functions. Take $f = 200$ Hz.

- $S(t) = \sin(2\pi f t) + 0.5\sin(3 \times 2\pi f t) + 2\cos(2 \times 2\pi f t)$
- $S(t) = \sin(2\pi f t) + 0.5\sin(2 \times 2\pi f t) + 2\cos\left(\sqrt{2}\ 2\pi f t\right)$
- $S(t) = \sin(2\pi f t) + 2\sin(3 \times 2\pi f t) + \cos(2 \times 2\pi f t)$

Money

6.11 Show, either by mathematical arguments or a numerical example, that if your money earns 7 percent every year, the value of your account grows exponentially approximately as $M = M_0\, e^{\alpha t}$, where t is time in seconds and M_0 the initial amount of money, assuming you do not add or subtract anything yourself. Find α.

The exponential horn

6.12 To avoid reflections and maximize energy transfer from a source to the air through a horn, the ideal shape for a horn or speaking trumpet is exponential (see *Why You Hear What You Hear*, chapter 7.) That is, the cross-sectional area of the horn should increase as $A(l) = A_0\, e^{\frac{l-L_0}{L_0}}$, where A_0 is the area of the horn at its mouth, a distance L_0 from its throat. We can build the horn out of n thin discs with areas $A_n = A_0\, e^{\frac{\beta n-L_0}{L_0}}$, where β is the thickness of each disc. Recalling that the impedance of each disc is Z_n, proportional to $1/A_n$, verify that the impedance of the n^{th} segment relative to its neighbors (n -1) and (n+1) obeys the geometric mean rule minimum impedance: $Z_n = \sqrt{Z_{n-1}\, Z_{n+1}}$.

Logs & exponentials

6.13 Solve for x in the following equations. Show your work!

- $6\,y = 10^{2x}$
- $\log_{10}\left(4x^2\right) = y$
- $y = e^{a x}$
- $\ln(4x) = 2y$
- $y\, e^{a x}\, e^{-a x} = 2\, e^{x/3}$

7 – Quiz questions

7.1 The speed of sound in a gas is faster with higher temperature.

True ☐ False ☐

7.2 SF6 is a gas a lot heavier than air. The speed of sound is expected to be higher in SF6 than in air.

True ☐ False ☐

7.3 The Rijke tube turns heat into sound.

True ☐ False ☐

7.4 Sound is an electromagnetic wave.

True ☐ False ☐

7.5 We can measure the speed of sound using a laptop, a clap of hands, and a reflective surface.

True ☐ False ☐

7.6 We hit a 440 Hz tuning fork above an aquarium. Inside the aquarium, the sound hitting Nemo's skin has a different frequency.

True ☐ False ☐

7.7 We record a source in open air at a distance of 1 meter and note the intensity. To measure the same intensity at a distance of 4 meters, we can place the source near the L-shape formed when a wall meets the floor at 90 degrees.

True ☐ False ☐

7.8 The sound energy produced by one person shouting is enough to get a cup of coffee to boiling starting at room temperature in

An hour ☐ A day ☐ A month ☐ A year ☐

7.9 $\log_{10}\left(10^{41}\right) = 41$

True ☐ False ☐

7.10 Tuning forks are able to ring for a long time because they have a node at the handle.

True ☐ False ☐

7.11 We record dolphin whistles underwater using a special underwater microphone. When we listen to the recording on a sound system in air, we need to change the pitch to hear the same frequencies as the dolphins, in order to compensate for the different speed of sound in water and in air.

True ☐ False ☐

7.12 With Seebeck's siren, we can hear a periodic sound with sinusoidal components, whereas the disturbance creating the sound was nothing like a sinusoidal source.

True ☐ False ☐

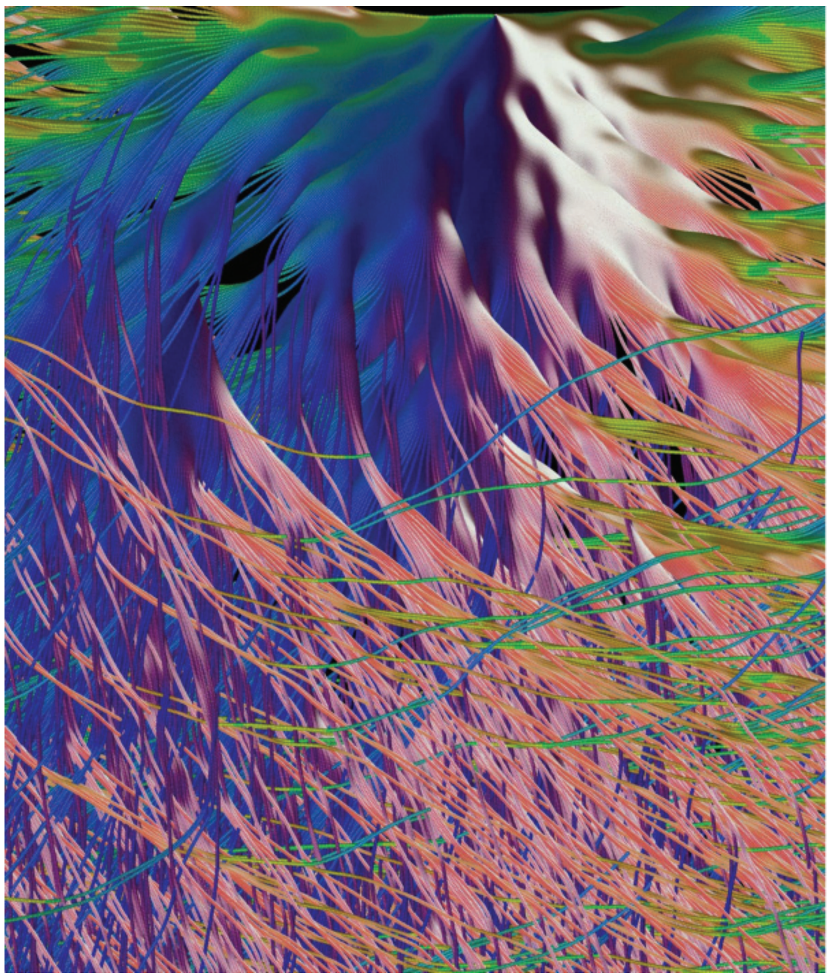
7.13 A car is honking continuously, and an observer is standing next to the road. When the car approaches the observer, then passes and recedes, the pitch of the honk heard by the observer goes:

Down then up ☐ Continuously up ☐ Up then down ☐

7.14 If the gas suddenly changes from air to SF6 in a tube, but the tube is of constant diameter (say the change is one meter down a two meter tube), no reflection of sound will occur at the change.

True ☐ False ☐

Analyzing Sound



8 – Problem: Audio-periodic exploration

Read in Why You Hear What You Hear

- Chapter 3: Sound and Sinusoids

Tools

- **Sonogram Visible Speech** (details on http://www.christoph-lauer.de/Homepage/Sonogram.html)(or another tool such as **Audacity**)

In this problem, we are going to use the sound analyzer **Sonogram Visible Speech**. We are going to focus on the power spectrum, given in the **FFT Window**.

8.1 Take a short sound file (**.wav** or **.aiff**) including constant musical tones (not changing pitch, etc.) You can use your own recordings. Open the sound file with **Sonogram Visible Speech**. You will see the sonogram and the waveform. Open the **FFT Window**. When you hover the mouse over the sound, the **FFT Window** is updated with the corresponding power spectrum.

✓ Indicate on a screen shot evidence for a fundamental frequency and it's harmonics. Give the frequency.

✓ Change the tone (new place in the sound file, or new sound file, etc.) and do again.

✓ Compare and comment on peaks.

Here is a screen shot of this kind of analysis, although this screen shot is for a chime tone, which does not have equal spaced harmonics. On the right, you can see the window **General Adjustments** (in menu **Options...**)

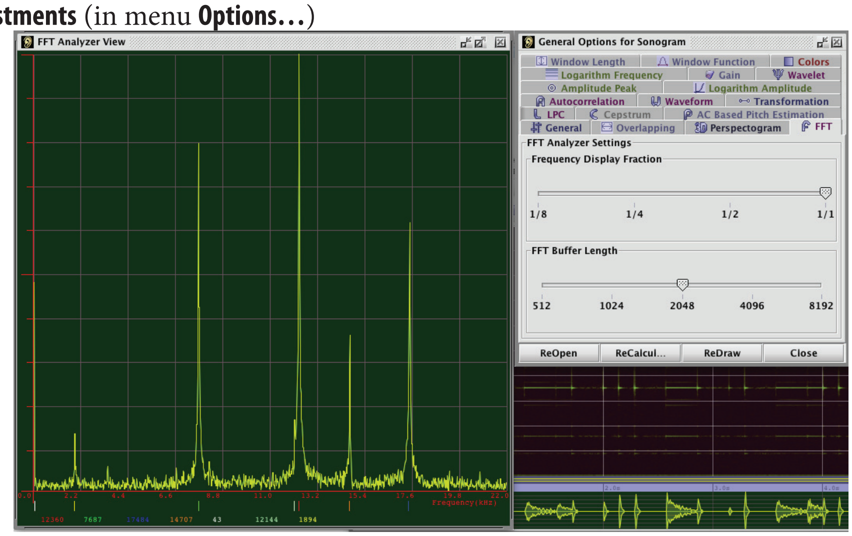


Figure 7: Sonogram Visible Speech

8.2 Invent your own investigation with another complex tone of your choosing; tell us what you did and your conclusions. Example: alter the sound file somehow and investigate the changes in the power spectrum.

9 – Problem: Fourier analysis & time/frequency uncertainty

Read in Why You Hear What You Hear

- Chapter 3: Sound and Sinusoids
- Chapter 5: Sonograms

Tools

- **Max Partials** (on http://www.whyyouhearwhatyouhear.com/subpages/MAX.html)
- **Audacity** (details on http://audacity.sourceforge.net/)
- **Sonic Visualiser** (details on http://www.sonicvisualiser.org/)
- Two sound files: **chirp2.wav** and **shortandshorter.wav** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

The Fourier theorem works both ways: using sinusoidal partials, you can synthesize any waveform. Conversely, any waveform can be analyzed into a set of partial amplitudes and phases. In this question, you are going to synthesize a signal using **Max Partials**, then analyze it using **Audacity**.

9.1 With **Max Partials**, generate a sound made of two partials: 500 Hz and 600 Hz. Using the **record** button, save a sound file with this signal. Open it with **Audacity**. Make a screen shot of the power spectrum, using different window sizes. Comment on your ability to identify both peaks in function of the window size.

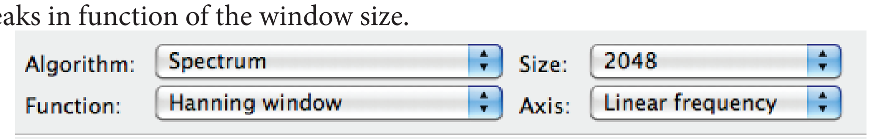


Figure 8: Audacity spectrum options. Here, window size is 2048 samples.

The window size is given in number of points (or samples). When the sampling rate is 44100 samples per second, the duration of the window size in seconds is (number of points) / 44100.

9.2 In **Sonic Visualiser**, when you see a sonogram (for instance, menu **Pane** -> **Add Spectrogram**), you can easily adjust the size of the window analysis. When you vary the analysis window size, should the frequency uncertainty vary proportionally or inversely proportionally to the window size?

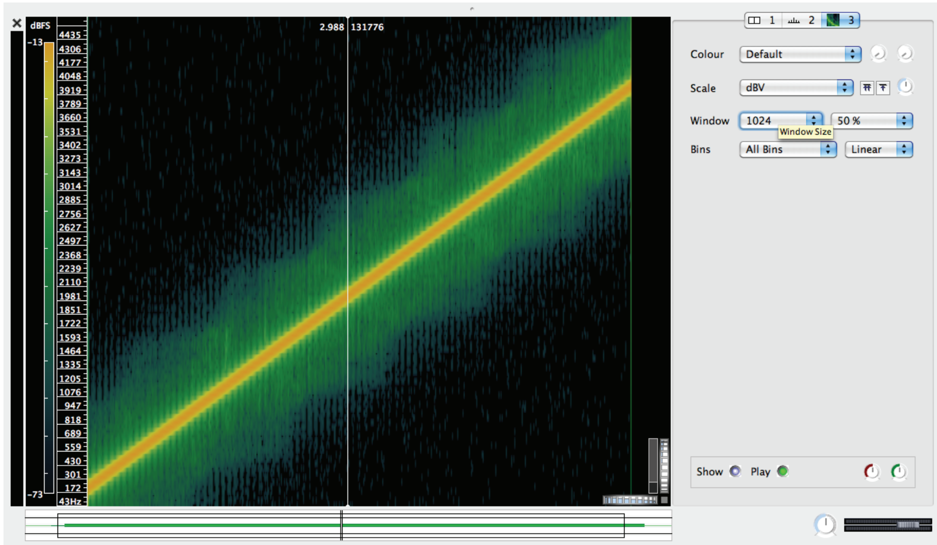


Figure 9: Chirp sonogram in **Sonic Visualiser**,
showing the **Window** size control, here 1024 points

9.3 Using the file **chirp2.wav**, make at least three screen shots of sonograms made with different window sizes (for instance between 64 and 2048), and show graphically this proportionality. Show that it is independent of the average frequency at the moment where you measure the frequency uncertainty.

9.4 What are the beginning and ending frequencies of the chirp?

9.5 What happens when you use a greater window size, for instance 8192, compared to 2048?

9.6 Show that there is a minimum frequency uncertainty in each of the two pulses in the file **shortandshorter.wav**, below which you cannot go no matter what the software setting of the window size. Explain.

10 – Problem: What's not on your iPod

Read in Why You Hear What You Hear

- Chapter 5: Sonograms

Tools

- **Raven Lite** (details on http://www.birds.cornell.edu/brp/raven/RavenOverview.html)
(or other software tool)
- Two sound files: **Tchaikovsky_clip.wav** and **Tchaikovsky_clip.mp3** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

To be able to carry around as much music as possible, most people load up their computers and digital music players with compressed audio files in formats such as **MP3** and **AAC**. To make the file significantly smaller, something had to be thrown away, but what?

10.1 Compare the sonograms of the uncompressed file **Tchaikovsky_clip.wav** and the compressed file **Tchaikovsky_clip.mp3** in a program such as **Raven Lite**. What jumps out at you as a way the **MP3** format saves space?
Hint: you may need to play with the sonograms' brightness and contrast settings.
Comment on any difference you see between the two sonograms.

11 – Problem: First I was afraid of sonograms

Read in Why You Hear What You Hear

- Chapter 3: Sound and Sinusoids
- Chapter 5: Sonograms

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- **Sonic Visualiser** (details on http://www.sonicvisualiser.org/)
- Seven sound files: **01-FirstIWasAfraid.aif** to **07-FirstIWasAfraid.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

In this problem, you are going to explore seven sound files, in order to get a feeling of the relations between a sound and its sonogram. You will need to make sonograms for several of the excerpts, for instance with **Sonic Visualiser**. To see the sonogram in **Sonic Visualiser**, do **Add Spectrogram** in the **Pane** or in the **Layer** menu.
Make sure you change the analysis window size according to what kind of results you want; remember the uncertainty principle: time versus frequency resolution.

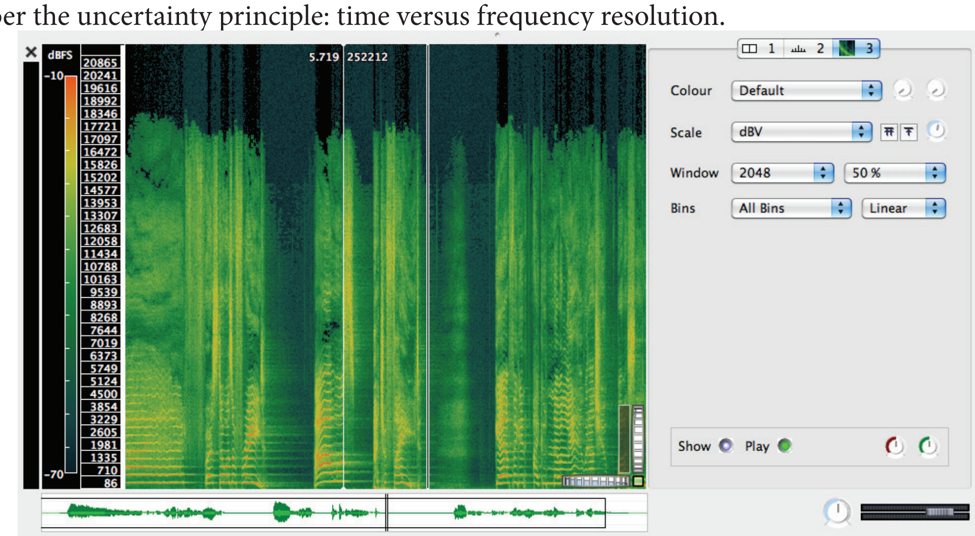


Figure 10: a **Sonic Visualiser** screen shot, here with a FFT size (**Window)** of 2048 samples

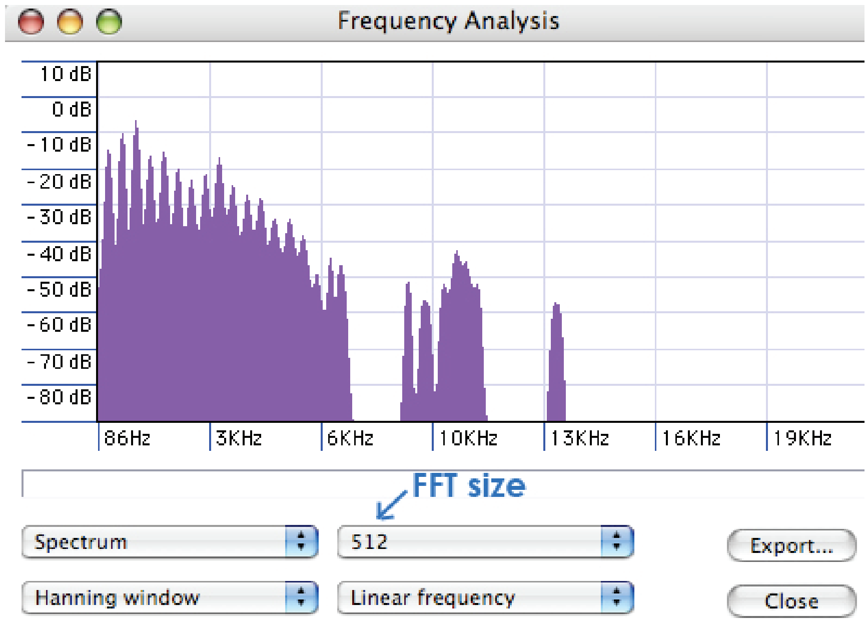


Figure 11: FFT size for **Audacity**'s spectrum, here 512 samples

11.1 Listen to the original sound: **01-FirstIWasAfraid.aif**
This is an excerpt of a song by the Musica nuda duet; a live version is there:
http://www.youtube.com/watch?v=4Zet3eLu-ms
Observe characteristics of vowels and consonants.
Is the first vowel an audio-periodic signal?

11.2 The second sound, **02-FirstIWasAfraid.aif**, is a transposition of the first. In this example, the speed has been increased by 10 percent. The effect is similar to what you can experience when playing a vinyl record at the wrong speed, for instance 45rpm instead of 33rpm. Here, we don't have the 45:33 ratio, but rather a 10% speed increase. By how many cents has the pitch increased compared to the original example?

11.3 Example 3, **03-FirstIWasAfraid.aif**, is another transposition example. Here, the speed of the original file has been reduced. By how much? Comparing spectrum analysis of the first vowel with Audacity's Spectrum is a good way to answer.

11.4 In the sound **04-FirstIWasAfraid.aif**, a process called frequency shifting has been applied (note that frequency shifting is different than transposition, i.e. pitch shifting). The frequencies in the original sound have been shifted down all by the same amount, as you will see comparing the sonograms. Guess roughly by how many Hz the frequencies have been shifted.

11.5 Another sound processing has been applied to the original file in order to produce **05-FirstIWasAfraid.aif**
Can you guess which kind of sound processing, maybe by listening, or by looking at the sonogram, focusing on the lower partials. While investigating this question, look at the first "vowel" (where the vowel was in question 11.1): is it an audio-periodic signal?

11.6 Frequency shifting can transform a harmonic sound (audio-periodic signal) into an aperiodic signal. But applying frequency shifting to an audio-periodic signal upwards, is it possible to get another audio-periodic signal? Explain.

11.7 Guess what sound processing was used here to produce **06-FirstIWasAfraid.aif** from the original file. Comparing the sonogram with the one for the original file should suffice to answer the question.

11.8 What about **07-FirstIWasAfraid.aif**? Look closely at both sonograms of the original sound and this one. What is missing in this one compared to the original? How does it sound? Can you guess the name of the kind of effect used? If not, feel free to make up a name of your own for the sound processing applied.

12 – Problem: A Hard Day's missing fundamental?

Read in Why You Hear What You Hear

- Chapter 3: Sound and Sinusoids
- Chapter 4: The Power of Autocorrelation
- Chapter 23: Pitch Perception

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- A sound file: **AHDN-chord.wav** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

The opening chord of the Beatles' song *A Hard Day's Night* is one of the most analyzed and debated in the history of pop music, in part because of the difficulty amateur guitarists have had in reproducing the characteristic sound of the "mystery chord". The reason is that this complex chord is played not only by George and John on 12- and 6-string guitar, respectively, but also by Paul on bass and producer George Martin on piano (you may be able to pick out the piano if you listen closely to the right channel near the end of the chord).

The chord is so famous, in fact, that an entire section is devoted to it in the Wikipedia article about the song: http://en.wikipedia.org/wiki/A_Hard_Day%27s_Night_(song)
The article includes a detailed breakdown of which notes are present in the chord from Dominic Pedler's book *The Songwriting Secrets of the Beatles*.

It is a difficult (if not impossible) task to determine which instruments actually played which notes, but let's suppose that Pedler's list of notes is accurate (neglecting Ringo's aperiodic contributions):

Harrison:	F2, F3, A2, A3, F3, F4, C4, G4
Lennon:	F2, A2, F3, A3, C4, G4
McCartney:	D3
Martin:	D2, G2, D3

12.1 Examine the power spectrum of the chord, and check whether the fundamental of each note is present (recall that **Audacity** tells you F2, F3, etc. when the cursor is near a peak). Are the fundamentals of any of these notes missing? Support your findings with an annotated power spectrum. Be sure to pick an appropriate window length for your spectrum so you don't miss anything.

12.2 The fingerprints of a missing fundamental are its higher partials. For any missing fundamentals you identified in 12.1), are its higher partials present? Are there any partials present that don't match the fundamental of any of the notes listed above?

12.3 Now examine the autocorrelation of the chord. Is there a peak corresponding to the missing fundamental(s) you found? Please support your conclusion with an annotated autocorrelation plot.

13 – Short questions

Log plotting

Consider the signal $S(t) = \cos(2\,\pi\,40\,t) + 40\cos(2\,\pi\,15\,000\,t)$

13.1 Plot a power spectrum (make sure to label correctly the axes):
- ✓ on a linear plot (linear scales for power and frequency)
- ✓ on a log-log plot (logarithm of the power versus logarithm of the frequency)

Matching waveforms with spectra

You have to match the three signals with the corresponding power spectra.

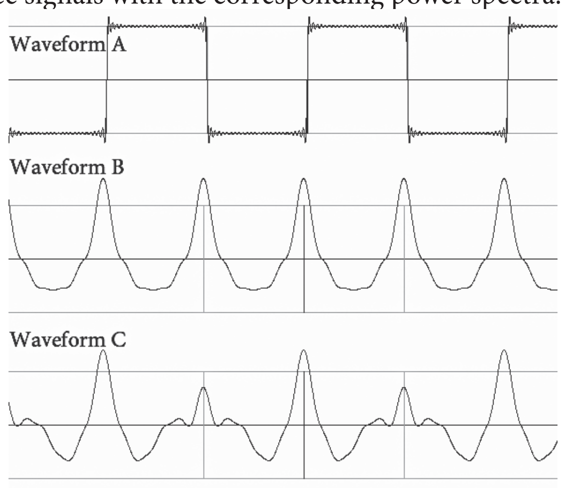


Figure 12: Three waveforms

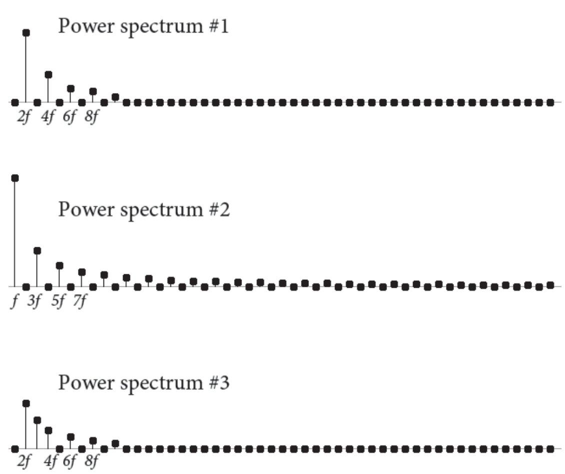


Figure 13: Three power spectra

13.2 We suggest three options. Which is the correct one? Explain.
- ☐ A-1, B-2, C-3
- ☐ A-2, B-1, C-3
- ☐ A-3, B-2, C-1

Autocorrelation math

A chime tone has four partials: at f_1 = 100Hz with a power of $(A_1)^2 = 3$, f_2 = 230Hz with a power $(A_2)^2 = 2$, f_3 = 370Hz with a power $(A_3)^2 = 3$, and f_4 = 480Hz with a power $(A_4)^2 = 1$.

13.3 Make a plot of the autocorrelation function for time 0 to 0.05 seconds and say what the perceived pitch might be.

Hint: think about the sum of sine waves that would add up to give this power spectrum, and remember that the autocorrelation function of $y(t) = A\sin(2\pi f t)$ is $\log_{10}(4x^2) = y$. Finally, note that the autocorrelation function of the sum of waves is just the sum of the autocorrelation functions of each individual wave.

Waveform & autocorrelation

A signal has peaks every 0.0025 seconds, but they alternate in width, as seen below.

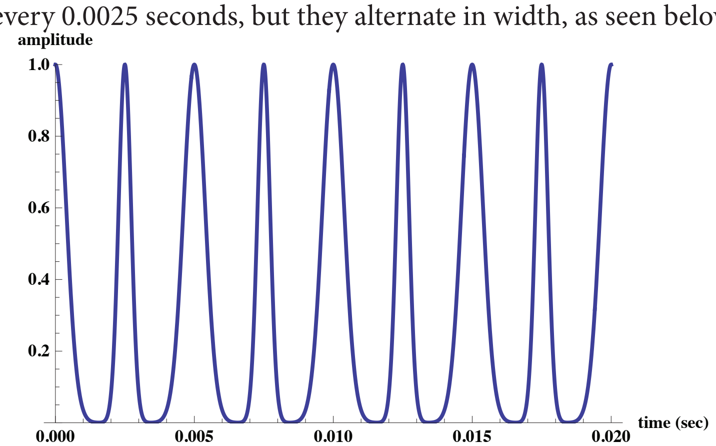


Figure 14: Waveform B

13.4 What is the period of this signal? What pitch will you hear if it is played?

13.5 Make a sketch of the autocorrelation function for the signal.
Hint: think of sliding a paper cutout copy over the original and recording the amount the two overlap. (If you are good with computer graphics you might try this for instance in Illustrator or Photoshop with transparency turned on.)

13.6 What is the period of the autocorrelation, when is the first large peak, and when is it? Does it corroborate your answer in question 13.4?

Autocorrelations & spectra

Periodic (with one exception) signals are shown below.

13.7 For each signal, draw an appropriate autocorrelation function and power spectrum. Your graphs are not expected to perfectly match what **Audacity** would generate, but they should capture the most important features of the signals. Your horizontal axes should be roughly quantitative but the vertical axes can be qualitative.

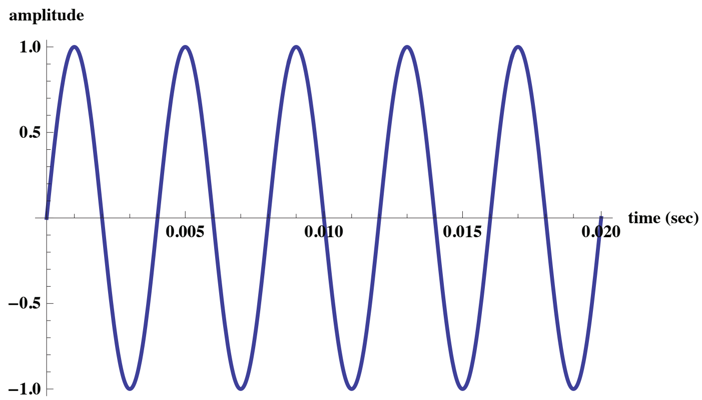


Figure 15: Waveform A

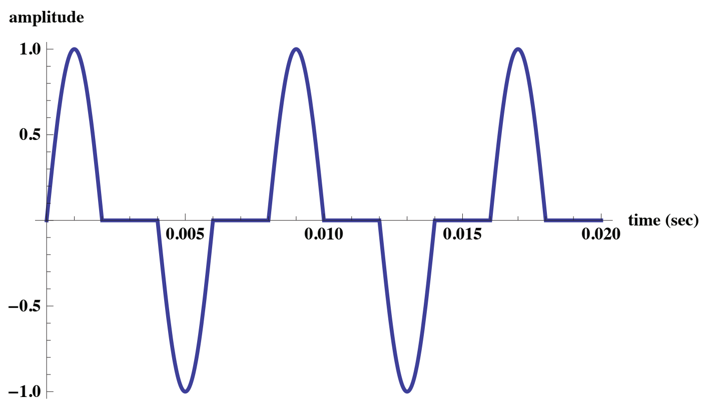


Figure 16: Waveform B

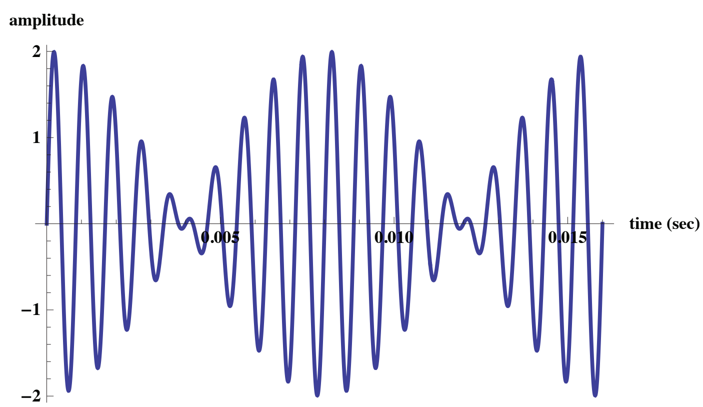


Figure 17: Waveform C

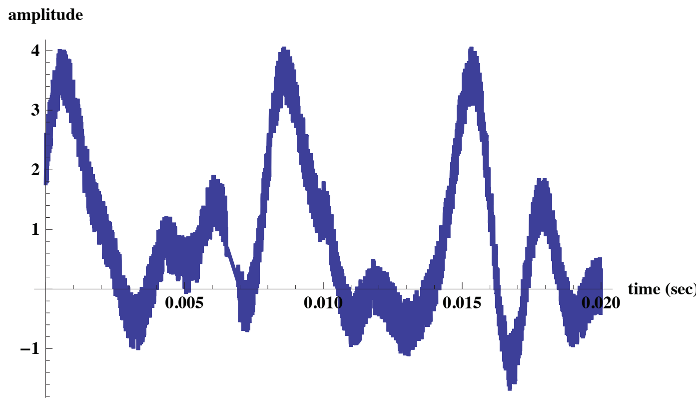


Figure 18: Waveform D

Matching soundtraces with sonograms

13.8 Match each of the three soundtraces (A,B,C) with its corresponding sonogram (1,2,3). Briefly explain your reasoning.

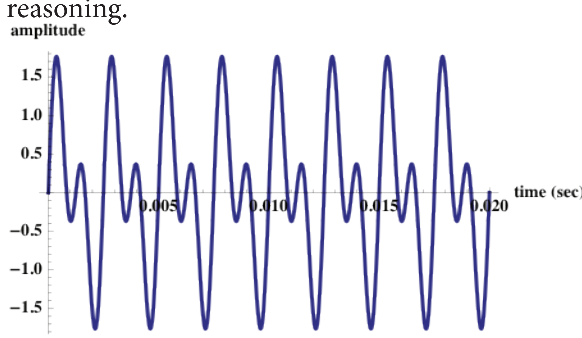


Figure 19: Soundtrace A

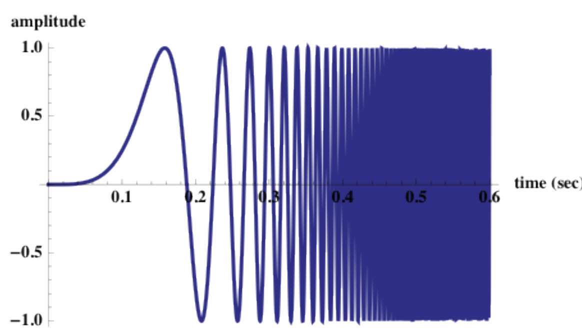


Figure 20: Soundtrace B

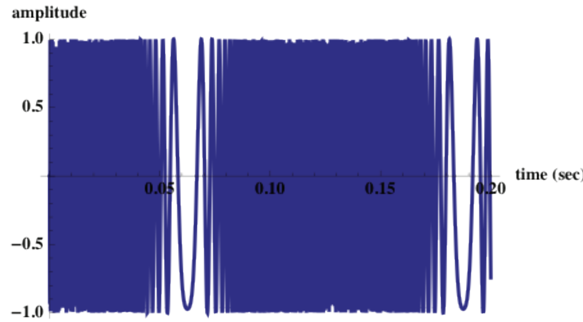


Figure 21: Soundtrace C

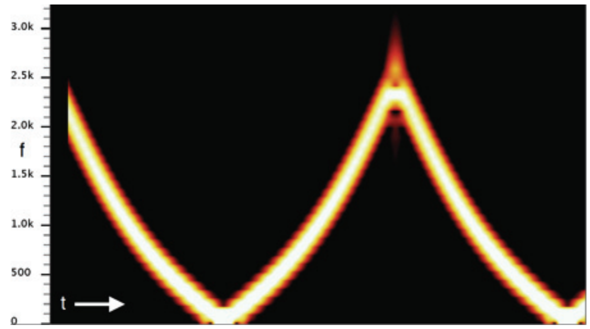


Figure 22: Sonogram 1

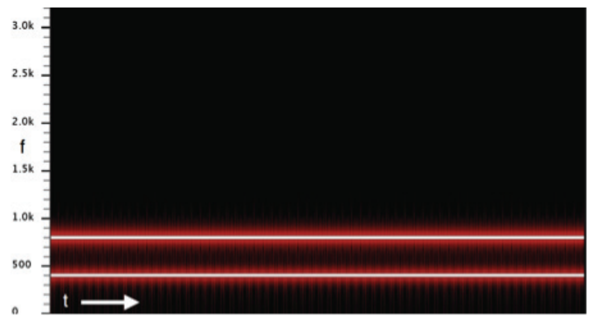


Figure 23: Sonogram 2

Figure 24: Sonogram 3

Matching sounds with sonograms

Tools

- Four sounds: **mystery-sound-01.aif**, **mystery-sound-02.aif**, **mystery-sound-03.aif**, and **mystery-sound-04.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

You should be able to answer after listening to the sounds, but feel free to analyze them with a software tool of your choice.

13.9 Match each of the four sonograms with its corresponding sound file (1, 2, 3, 4). For each match, point out a feature that you find interesting in the relation between graphic and sound.

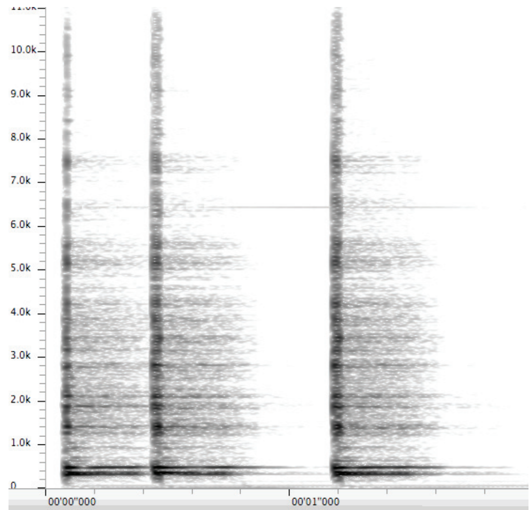


Figure 25: Sonogram A

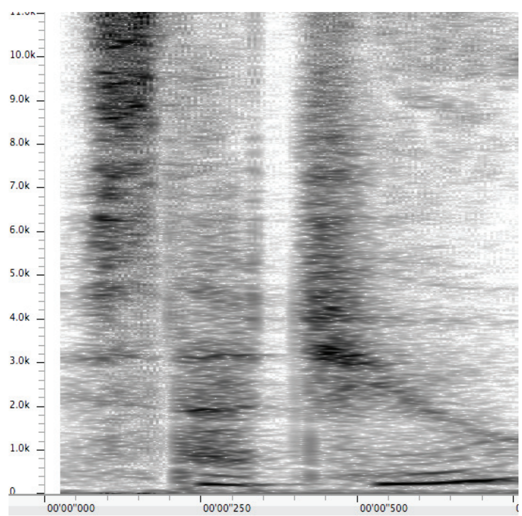


Figure 26: Sonogram B

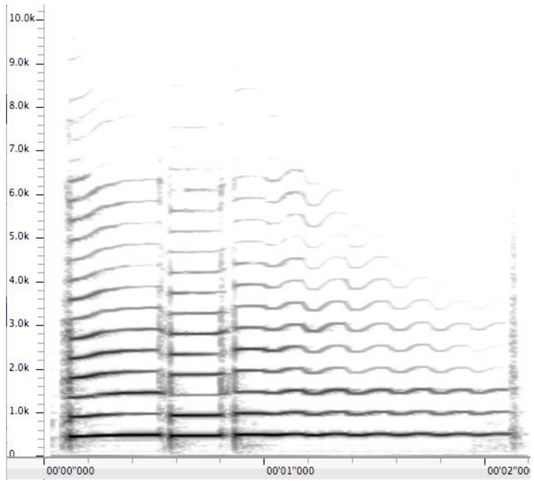


Figure 27: Sonogram C

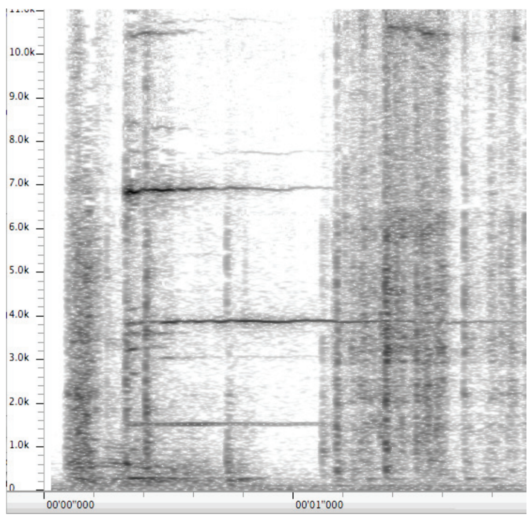


Figure 28: Sonogram D

Car sounds

13.10 Match the sound description and its sonogram, among the sonograms A to E below.

✓ Sound #1 is the horn of a Daf 66 SL classic automobile.

Sound #1 matches sonogram______

✓ Sound #2 is the siren of a New Zealand Police patrol car on wail mode.

Sound #2 matches sonogram______

✓ Sound #3 is a VW Passat door being opened and slammed, recorded inside a garage (6 × 6 meters).

Sound #3 matches sonogram______

✓ Sound #4 is a diesel engine starting.

Sound #4 matches sonogram______

✓ Sound #5 is a diesel locomotive moving rather slowly, with about 20 cars in tow. Whistle blows nearby and in distance.

Sound #5 matches sonogram______

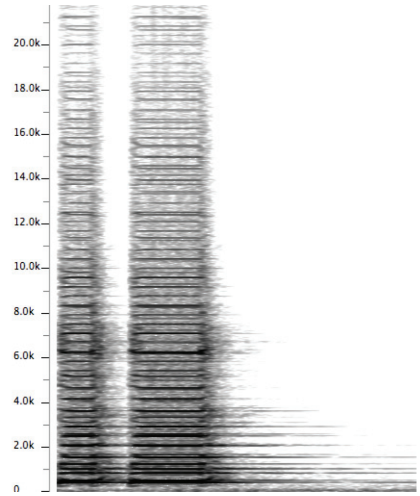


Figure 29: Sonogram A

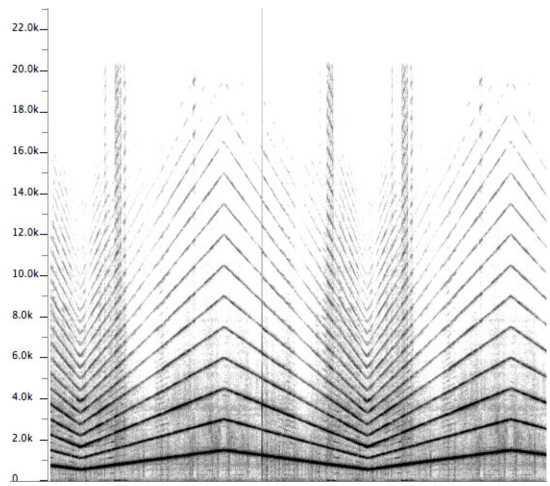


Figure 30: Sonogram B

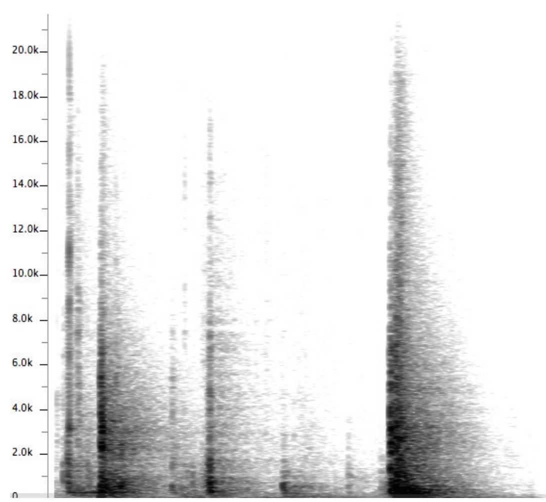


Figure 31: Sonogram C

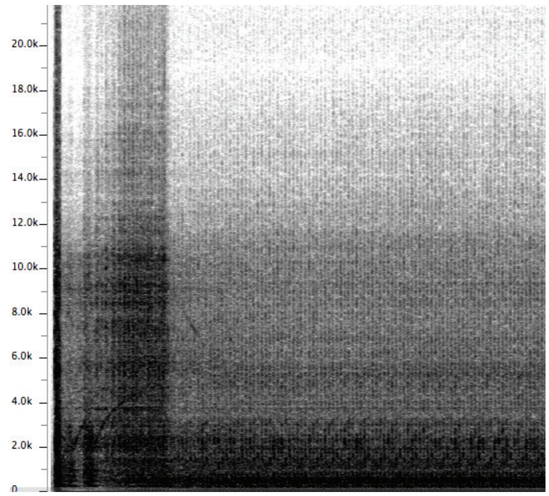


Figure 32: Sonogram D

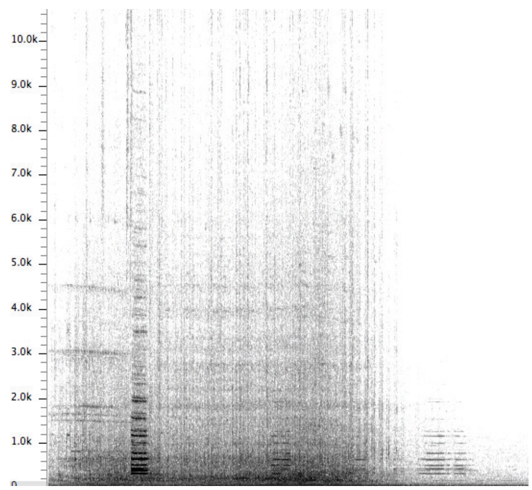


Figure 33: Sonogram E

Low frequency

13.11 To analyze low frequency sounds with a good resolution, you need long samples, which involve lots of data at a 44.1 kHz sampling rate. For example, the uncertainty principle implies a 1 Hz uncertainty for 1 second of data. This statement does not depend on frequency. One Hz is not much of a percentage error for a 1000 Hz partial, but it is a 5% error at 20 Hz. What might you do to analyze low frequency sounds (say less than 50 Hz) without increasing the amount of data?

14 – Quiz questions

14.1 The power spectrum of a pure sine tone shows regularly spaced partials.

True ☐ False ☐

14.2 In an autocorrelation function, we see a peak at 10ms. In the power spectrum, there is almost no power at 100 Hz. Is this possible?

Yes ☐ No ☐

14.3 The autocorrelation of a pure sine wave is a single peak.

True ☐ False ☐

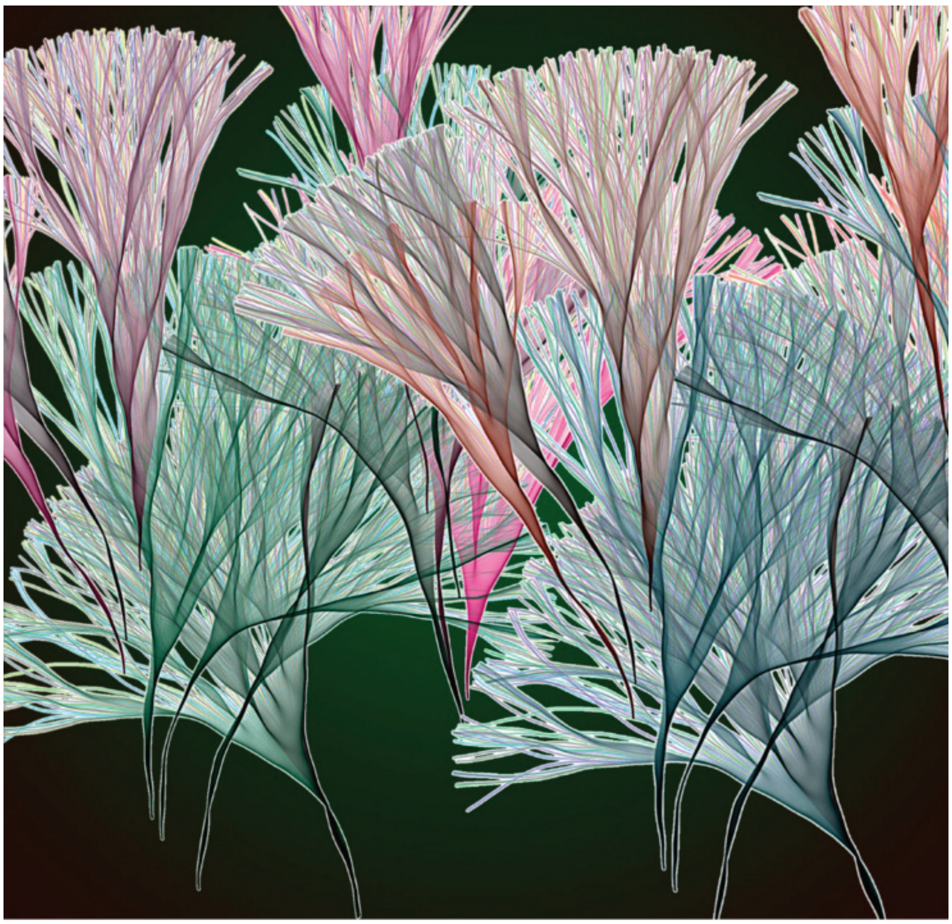
14.4 When we multiply the frequencies of all of a harmonic sound's partials by the twelfth root of two ($\sqrt[12]{2}$), its pitch becomes a semitone higher (this is true, it's not a question!) Now, the question statement: when we multiply the frequencies of all of a harmonic sound's partials by the thirtieth root of two ($\sqrt[13]{2}$), the sound becomes inharmonic.

True ☐ False ☐

14.5 A viola plays a A3 (fundamental frequency 220 Hz). One way to transpose the note an octave up is (using a specific software tool) to add 220 Hz to all of the note's partials.

True ☐ False ☐

Making Sound



15 – Problem: String simulation

Read in Why You Hear What You Hear

- Chapter 8: Making a Stretched String
- Chapter 23: Pitch Perception

Tools

- **Falstad Loaded String applet** (on http://www.falstad.com/loadedstring/)
- Optionally **Audacity** with **JackAudio** or **Soundflower** to route and record sound

---

In this exercise we see the effect of periodic and non-periodic motion on sound and the vibration of objects.

In **Falstad Loaded String applet**, you should set up your experiment with the **Stopped** box checked, and only un-check the box when you're ready to run your experiment. Make a three load string by moving the **Number of Loads** slider most of the way to the left, until there are only 3 loads left. You can excite individual modes of oscillation separately by pulling up the vertical sliders under the **Magnitudes**.

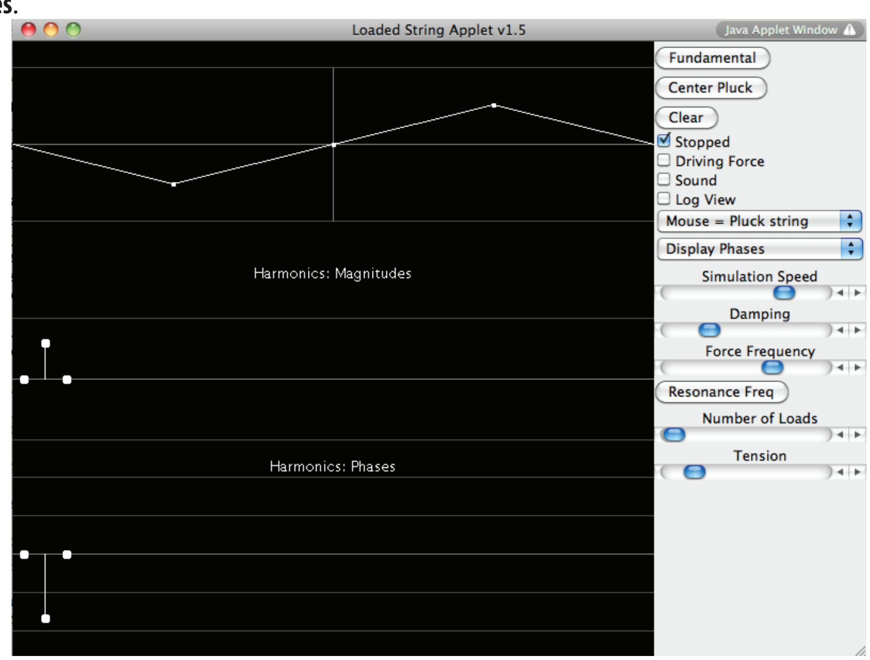


Figure 34: Falstad Loaded String Applet, with three loads, here, second mode active

Measure the frequencies of the three independent modes by exciting them separately. Using a stopwatch to measure the periods (then convert to frequency), or record the sound from each one and use **Audacity** to measure the frequency. Note that you can record the sound using a microphone & speakers connected to your computer, but for the best quality, you would route internally the sound from one application to the other with tools such as **JackAudio** (cross-platform) or **Soundflower** (Mac).

15.1 Report all three frequencies. Are they evenly spaced?

15.2 Now, begin with some string shape that excites all three modes together and find out whether the motion is periodic in a reasonable amount of time. You can do that visually with a low simulation speed, or you can record the sound and observe the waveform. Relate your finding to your answers in question 15.1.
Note that you can excite the three modes either with a pluck or by dragging up all three magnitudes before un-checking **Stopped**.

15.3 Listen to the sound with all three modes excited as in question 15.2. Would you call it a "harmonic" tone or a "chime" tone? Explain.

15.4 Now max out the number of loads, and repeat parts 15.1, 15.2, and 15.3, using just the first three modes of the string, leaving the others with no amplitude. You can do this by pulling up the vertical sliders corresponding to those modes at the bottom of the screen. If it's hard to grab just those three, lower the number of loads, pull up the ones you want, and then max out the number of loads before you uncheck **Stopped**.

16 – Problem: Driven damped oscillator

Read in Why You Hear What You Hear

- Chapter 9: Resonance Rules
- Chapter 10: Damped and Driven Oscillation

Tools

- **Mathematica Driven Damped Oscillator** (on http://demonstrations.wolfram.com/DrivenDampedOscillator/)

We encourage you to play with the various parameters in the **Mathematica Driven Damped Oscillator**. For this problem, set the mass m = 2.5, the spring constant k = 1.0, the forcing amplitude initially to 0 (i.e. no forcing to start with; you can type in the number 0 even if the slider doesn't go there), and the damping d = 0.3. An important thing to know about this demonstration is that it uses $\omega = 2\pi f$ for frequency, instead of f. Be careful about ω versus f: the physics doesn't care which one you use, but if you mix them up your numbers might end up weird in later parts of this problem!

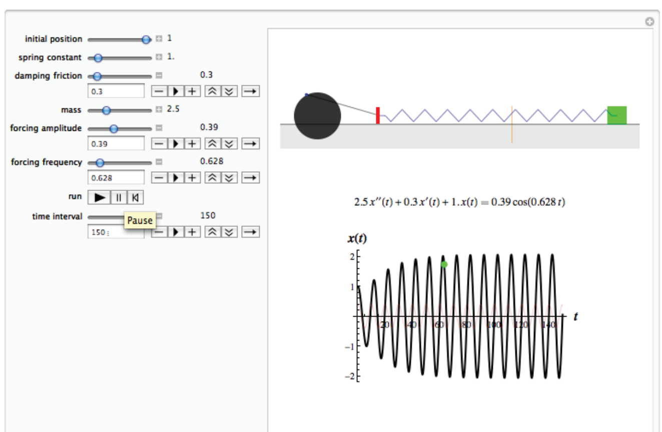


Figure 35: Driven damped oscillator with driving turned on

16.1 Predict the resonant frequency by using the formula $\omega_{\rm res} = 2\pi f_{\rm res} = \sqrt{\frac{k}{m}}$

You can give your answer as ω_{res} or f_{res} , but remember that you will use the ω_{res} number in the **Mathematica** demonstration.

16.2 A more accurate formula is $\omega_{\rm res} = 2\pi f_{\rm res} = \sqrt{\frac{k}{m} - \left(\frac{d}{2m}\right)^2}$

Calculate the resonant frequency using the more accurate frequency too.
Is it very different?

16.3 Approximate the "Q" of the oscillator in three ways:

✓ calculated, knowing the parameters of the oscillator (d, m, etc.) using the formula:

$$\Delta f = \frac{d}{2\pi m}$$

✓ based on the "4% rule" (must be done with the drive amplitude set to zero)

✓ after applying some forcing amplitude, the ratio of max displacement to amplitude of the drive (you must be driving at the resonant frequency – remember to express it in radians per second)

16.4 Now make a graph of the amplitude of oscillation (height on vertical axis after the oscillations have settled into a regular repetition) against frequency. To do this, drive the oscillator at at least 5 different frequencies on either side of the resonance frequency and make note of the oscillation amplitude at each frequency. You have to do this by hand, reading off the amplitude from the graph displayed in the simulation to collect your data. You should see a very clear difference when you are close to the resonant frequency than when you are far from it.

16.5 What is the frequency uncertainty Δf as measured from your graph (this is the width of your resonance peak halfway up the peak). Do you get the same answer using the formula from class: $\Delta f = f_{res}/Q$? Again remember to be careful about ω versus f.

17 – Problem: What is the Q of your mouth?

Read in Why You Hear What You Hear

- Chapter 9: Resonance Rules
- Chapter 10: Damped and Driven Oscillation
- Chapter 13: Helmholtz Resonators

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)

17.1 Record a "tongue pop" with your mouth in two different shapes (e.g. pursed lips, open lips). Find Q of your mouth in these shapes.

17.2 What is the center frequency of the sound produced by your mouth in each of these shapes?

17.3 Describe the steps required to determine (conceptually, not in **Audacity**!) the resonant frequency of your mouth from your recording. Why does this work? What would be the FWHM (Full Width at Half Maximum) of the largest peak in the resulting power spectrum?

18 – Problem: Simulation of a Helmholtz resonator

Read in Why You Hear What You Hear

- Chapter 13: Helmholtz Resonators

Tools

- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)

In this problem you will simulate a Helmholtz resonator and analyze the probe data to measure the frequency and Q of your oscillator.

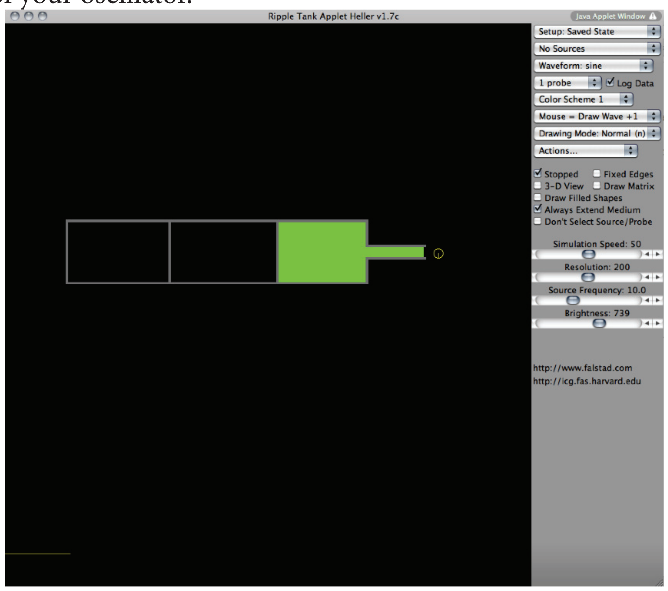


Figure 36: Helmholtz resonator in **Harvard Falstad Ripple applet**

Setting up the **Harvard Falstad Ripple applet**:
You can set the **Simulation Speed** higher or lower to make some things easier to measure, but leave it fixed after that. Some experimentation to optimize your conditions may be needed. You'll want a cavity something like that shown above. The resonator in the picture above has a particularly high Q and low frequency, and it takes a very long time for the oscillations to die enough to measure the Q. You might want to adjust the dimensions to speed things up: we advise you to design a resonator with a low Q; that will make the data more manageable.
Here's how to get the applet set up:

✓ Open the **Harvard Falstad Ripple applet**
✓ Check **Stopped**
✓ Select **Clear waves** under **Actions...**
✓ Uncheck **Fixed Edges**
✓ Set **No Sources** (2nd menu from top)
✓ Set **1 probe**
✓ Set the **Resolution** to about 200
✓ Set **Mouse = Draw Walls**, and draw walls as required. Use drawing mode vertical and horizontal if you want things to look nice. Important for filling the bottle: draw closed shapes and open them up with **Erase Walls** later.

- ✓ Set drawing mode **Fill: To Value**
- ✓ Set mouse to **Draw Wave +1** (this adds positive pressure)
- ✓ Click inside your region – it should turn green
- ✓ Set **Drawing Mode: Normal**
- ✓ Set **Mouse = Erase Walls**, and erase parts of walls as required
- ✓ Drag the probe around to where to want it
- ✓ Check **Log Data**
- ✓ When ready, unchecked **Stopped**
- ✓ Check **Stopped** to take screen shots
- ✓ After enough time has elapsed and the oscillation has grown weak, stop the simulation and under **Actions...** select **Save Probe Data**. Use **Actions… Clear Probe Data** if you want to change something and start with a fresh data file.
- ✓ A test file with your data should show up, probably in your downloads folder, which you can now analyze in Excel or any other equivalent program you like. Ignore any noisy early part of the data and start your analysis on some "clean" peak amplitude.

18.1 Find the Q and the center frequency of your resonator. You will need to look at the saved data file. Assume the units for the time data in your logfile are milliseconds.

18.2 Study the dependence of the center frequency on the "volume" (really area here). To this end, make your resonator have easily extensible straight walls.

- ✓ Find and plot at least three found frequencies as a function of area using the same neck for each.
- ✓ Take a screen shot of your three resonators.
- ✓ Does the frequency go like $\frac{1}{\sqrt{area}}$?
- ✓ Did the Q change?

19 – Problem: Four Helmholtz resonators

Read in Why You Hear What You Hear

- Chapter 13: Helmholtz Resonators

Tools

- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)
- **Harvard Falstad Ripple uploading tool** (on http://www.courses.fas.harvard.edu/~icgzmod/java_physics/saved_state/upload.html
- a file with saved applet state: **rippleState_2013-01-06_15.04.24_81.220.251.202.txt** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

The natural frequency of a Helmholtz resonator depends on many parameters, including the dimensions of the neck and body. We can gain a feel for the effects of these parameters using a software simulation.

Open the given saved **Harvard Falstad Ripple applet** state. Load it into the applet using the uploading tool. Uncheck **Stopped** to start the simulation.

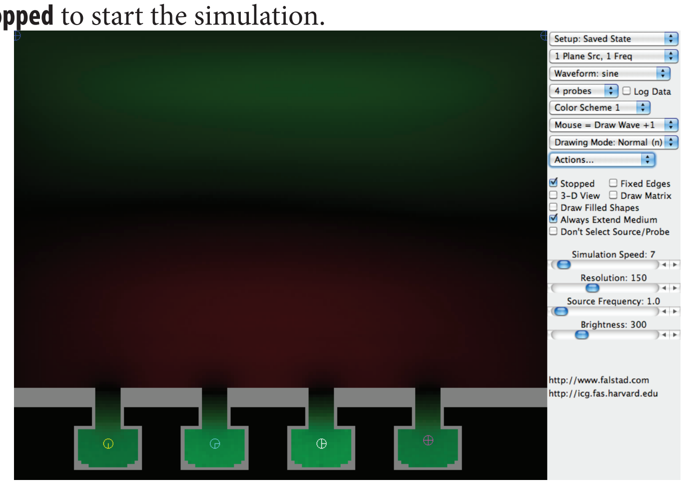


Figure 37: Simulation of four Helmholtz resonators

You should see four identical Helmholtz resonators, being driven very close to their resonant frequency by a sinusoidal plane wave drive with frequency 1.0. Note that the wavelength is much larger than the resonator.

19.1 Modify the neck of the second resonator so that it resonates when the driving frequency is about 0.5. If you need an easy way to move the body without redrawing it (e.g. to change the length of the neck), you can use **Mouse = Duplicate**.

19.2 Modify the body of the third resonator so that its resonant frequency is 1.5.

19.3 Modify the fourth resonator in any way you like so as to give it a resonant frequency of 2.0.

19.4 When a sinusoidal drive is tuned to the resonant frequency of any one of these four different resonators, how do the others behave?

19.5 Now switch to a sawtooth waveform (use **Waveform: sawtooth**). What happens when a drive frequency of 0.5 is used? What about a drive frequency of 1.0? Why?
Hint: your answer should involve the concept of Fourier decomposition.

19.6 Driving the system with a sawtooth waveform at frequency 0.5, position a probe in the body of each resonator, wait for several periods to be traced out. Print and attach a screen-shot of the entire **Harvard Falstad Ripple applet**, including the probe traces and the settings panel.

20 – Short questions

Strung out

A string weighs 0.01 kg/m, and it is under a tension of 3 N. Its lowest frequency is 500 Hz.

20.1 How long is the string?

20.2 The string is plucked. How fast will the wave thus created travel along the string?

String math

Consider a string stretched with a 1 kg mass in the earth's gravity. The string has a density of 0.003 kg/meter, and is 1 meter long.

20.3 Give the frequencies of the first 6 partials.

20.4 Make accurate pictures of the first 6 modes, showing where the nodes are located, labeled accurately.

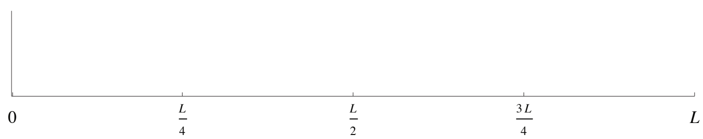


Figure 38: Example axe to draw a mode

20.5 State which modes would be affected and in which directions (lower or higher frequency) by weighting the string in the middle.

Damped string math

20.6 A violin string of length L has a fundamental frequency of 100 Hz. What are the frequencies of the first four partials of the string?

20.7 Sketch the shapes of the first four modes of the string. Label nodes on your sketches with a • and antinodes with an ×. What are the wavelengths of each of these modes?

20.8 Say that all the modes of the string have Q factors of 25. If we could pluck only the second mode and let the string vibrate, how long would it take for the amplitude of the motion to damp to 4% of its initial value?

20.9 Sketch the power spectrum of the string if it is plucked at an arbitrary point (nothing special about it). Make your frequency axis go from 0 Hz to 50 Hz above the 4^{th} partial. Do not worry about the heights of the power spectrum peaks. Please try to be accurate in drawing their widths.

20.10 In a real string, the higher frequency modes damp out faster than the fundamental modes (so when you pluck the string, the last thing you hear is the fundamental, without the harmonics). Given this fact, is it reasonable that all of the modes of the string have the same Q factor? Explain.

20.11 Sketch the power spectrum of the string if it is plucked exactly in the center. Make your frequency axis go from 0 to 450 Hz.

Wave reflection and boundary conditions

For a string $L = 1$ meter long, tied down at both ends, density $\rho = 0.02$ kg per meter, tension $T = 1$ Newton ($= 1$ kg-meter/sec^2), a shape as shown on the figure below is traveling to the right.

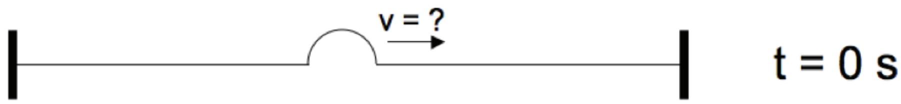


Figure 39: The string starts out at time = 0 with a pulse traveling to the right

20.12 Show six snapshots of its motion, including its encounter with the fixed end on the right, and afterwards. Label the time elapsed in each snapshot. It means you'll have to calculate its velocity of the wave, so that you know what happens when!

Drive & resonance

Here are two pairs of power spectra for a drive (solid line) and a resonant system (dashed line).

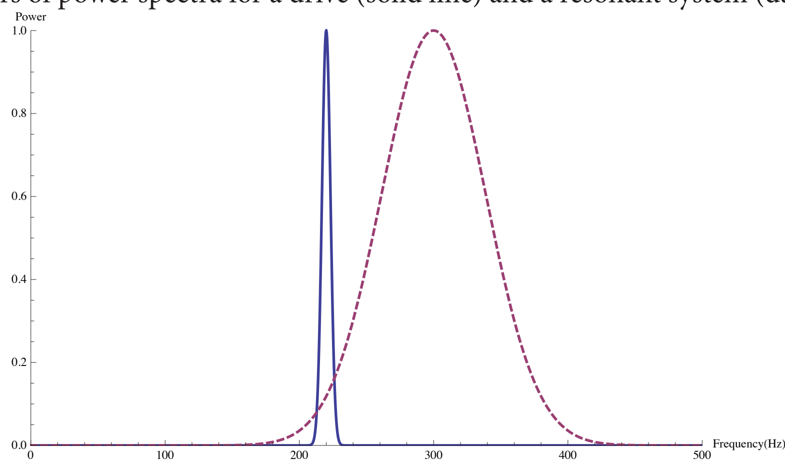


Figure 40: Drive (solid line) and resonant system (dashed line) 1

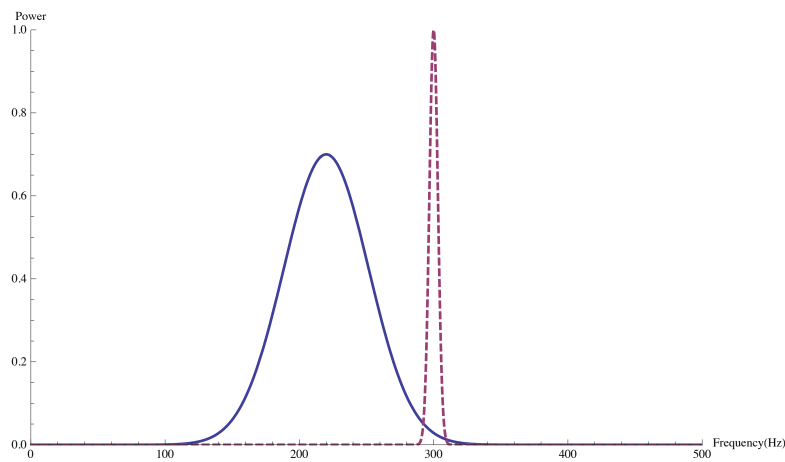


Figure 41: Drive and resonant system 2

20.13 When each drive is applied to its system, what frequency (or frequencies) will you hear the loudest? Assume there is no feedback from the system to change anything about the drive frequency.

20.14 Assuming these are all drawn on the same scale, which resonator has the highest Q? How can you tell?

Driver and driven

Below we show four power spectra, the top two belonging to high impedance drives (they don't respond much to feedback), and the bottom two belonging to the system that is to be driven. There are four possible combinations of drive and system.

20.15 In each of the four cases, discuss, as quantitatively as you can, the sound that will result.

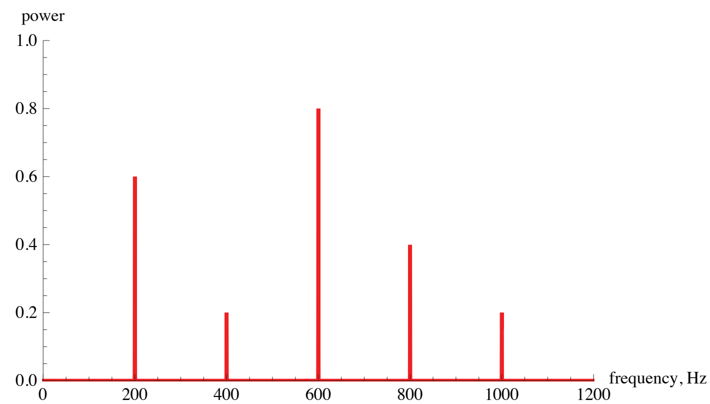


Figure 42: Drive A

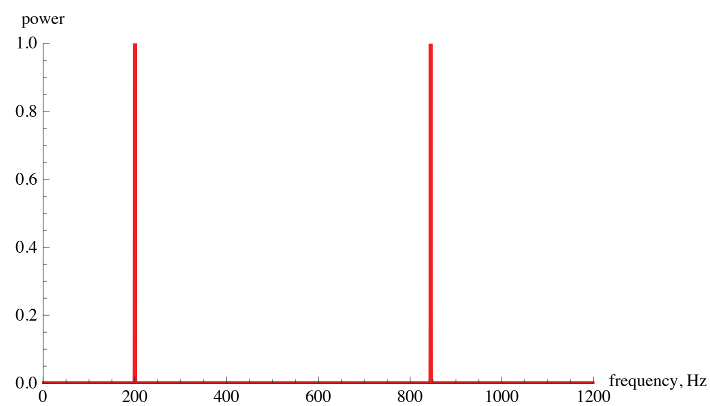


Figure 43: Drive B

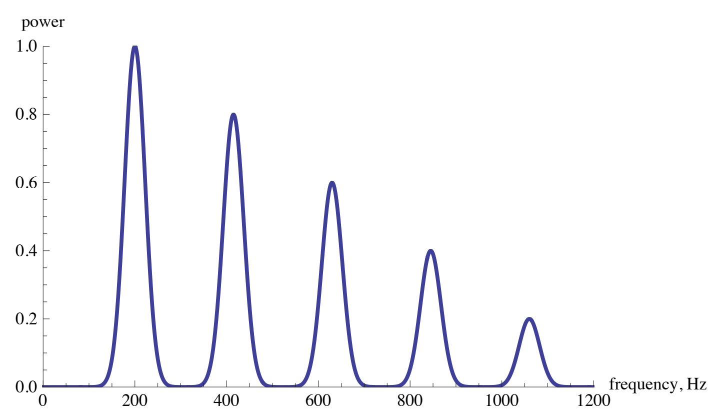


Figure 44: System 1

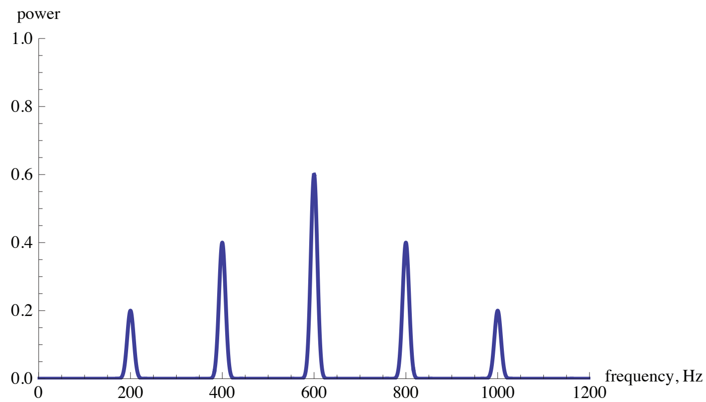


Figure 45: System 2

Damping and Q

A damped oscillator has a resonant frequency of 100 Hz, and a Q of 10. The oscillator is displaced by an amount 0.1 meter and let go.

20.16 Sketch its subsequent motion qualitatively on a plot showing amplitude (distance to the rest position, positive for one side, negative for the other) versus time. Be sure to label the axes in seconds and meters.

20.17 The same damped oscillator is driven with a strong sinusoidal drive of variable frequency. Sketch the oscillator's amplitude as a function of frequency of the drive from 0 to 200 Hz. Do not worry about the scale of the vertical axis, but label the scale of the frequency axis for the drive (the horizontal axis).

Q & decibels

A 1000 Hz metal bar rings audibly for three seconds, during which time the power output from the bar drops by 60 dB.

20.18 What is the Q for this oscillator?

Oscillator Q & R

An oscillator has a quality factor Q = 8.

20.19 Draw a graph for the displacement versus time if the oscillator is let go from rest with an initial displacement of 0.5 m and has a mass of 1 kg and a force constant of $k = 3$ kg/sec^2.

20.20 What is the friction factor R, given that the quality factor can be written as $Q = \frac{2\pi f_0 m}{R}$?

Jar room

Suppose we have a jar with a 2 cm long neck, area of neck 2 cm^2, volume of jar is 0.001 m^3.

20.21 Compute the frequency of the Helmholtz resonance.

20.22 Now consider a set of jars of the same proportions, but scaled up. This means that the neck is scaled in length by a factor of x, the area by x^2, and the volume by x^3. How does the frequency of different sized jars scale with x?

20.23 Suppose $x = 100$, making the jar the size of a small room. What is the frequency? Voilà, you have just done a bit of room acoustics!

Inside the car

20.24 Have you ever heard an annoying low frequency oscillation when driving fast with a window open? Describe what is happening.
What happens to the oscillation when you open another window?

Boom car acoustics

20.25 You notice a loud frequency in your room with a window open when a boom car (Street Pounder; Trunk Thumper) parks nearby, playing a repetitive bass beat with two successive frequencies. You have perfect pitch and notice that one is 30 Hz and the other 45 Hz. The 30 Hz note is much louder than the 45 Hz note in your room. When you step outside you notice they are now the same loudness. Explain.
What should happen to the loudness of the two tones as you bring your ear near a wall in your room?

20.26 Back in your room, you also notice the low frequency gets less loud when you open a second window. Explain.

20.27 The boom car now switches to a new tune, again with two beat notes, 28 Hz and 46 Hz. Considering question 20.25, what frequencies do you now hear in your room with one window open. Which one is likely to be louder?

Resonant frequencies

20.28 For the following objects, predict each one's fundamental resonant frequency f_0 using the formulas and concepts discussed in *Why You Hear What You Hear*. Assume you are in normal air, so the speed of sound is $c = 330$ m/s.

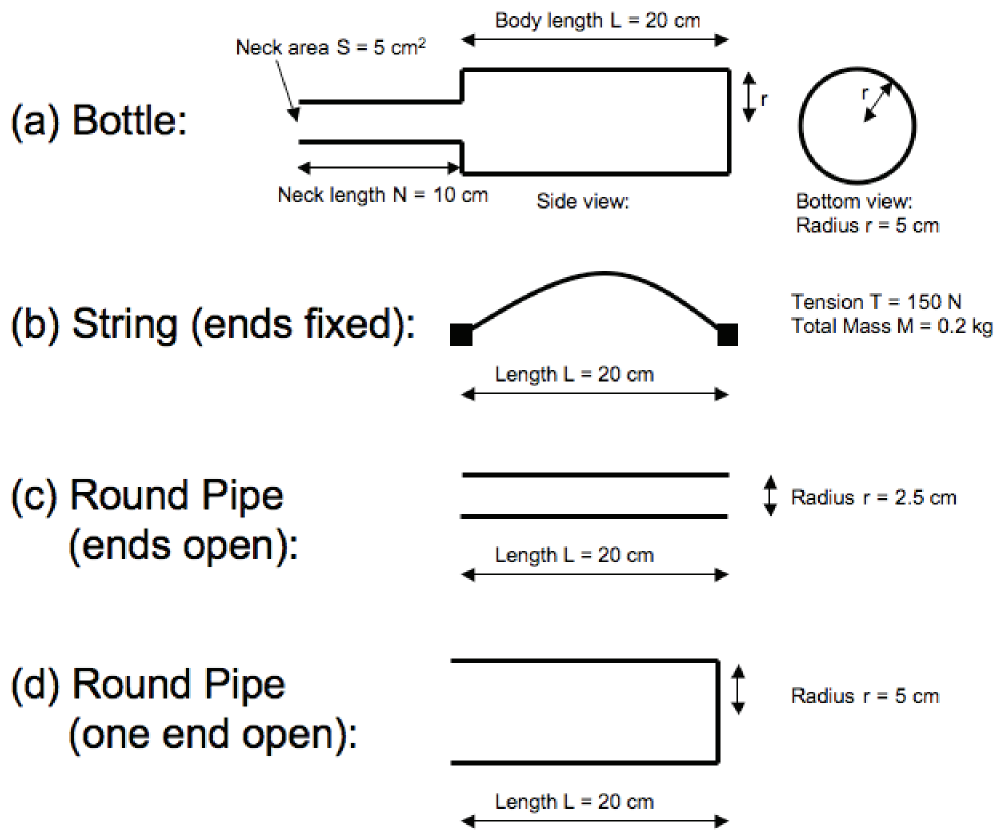


Figure 46: Four objects

Pipes

In the following replica of a **Harvard Falstad Ripple** simulation, a pulse is traveling inside a pipe. On parts 1, 2, and 3, the wave pulse is green.

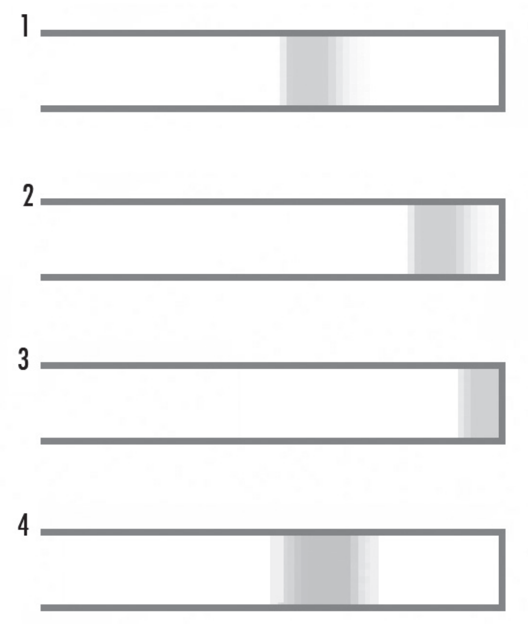


Figure 47: Pulse in a pipe

20.29 Which color is the wave pulse in part 4 of the graphic?

Green ☐ Red ☐

Let's imagine that in the following pipe,

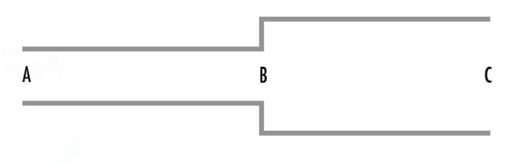


Figure 48: Pipe with two diameters

a single pulse of positive air pressure is traveling from point A, in the direction of point C.

20.30 What is happening when the single pulse reaches point B (check all that apply)?
- ☐ Reflection
- ☐ Refraction
- ☐ Diffraction
- ☐ Destructive Interference

20.31 Considering the same pipe as in question 20.30, the impedance in section BC is:
- ☐ Greater than the impedance in section AB
- ☐ Equal to the impedance in section AB
- ☐ Less than the impedance in section AB

Vortex shedding & string vibration

20.32 Describe in your own words how vortex shedding might get into resonance with string vibration. Since vortices tend to shed faster at higher wind velocities, how is it that wind can drive the same string to vibrate at almost the same frequency over a range of wind velocities?

21 – Quiz questions

String

21.1 A 0.2 m string and a 0.4 m string have fundamentals an octave apart if they are the same material under the same tension.

True ☐ False ☐

21.2 If the speed of a wave on a string is 10 m/sec, and its frequency is 5 Hz, its wavelength is 4 m.

True ☐ False ☐

21.3 When we put the tape in the middle of the monochord string, we raise some of its frequencies and lower others.

True ☐ False ☐

21.4 The lowest frequency mode of a 10-bead system is activated no matter which bead is driven at the mode frequency.

True ☐ False ☐

21.5 A string vibrating purely in its fifth mode generates a sound with a frequency 25 times higher than the frequency of its first mode.

True ☐ False ☐

Resonance

21.6 On resonance, an oscillator with damping is taking more power from the drive than it would if the drive were off resonance.

True ☐ False ☐

21.7 Blowing across the top of a bottle to get it to resonate is an example of cooperative resonance.

True ☐ False ☐

21.8 The breaking of a wineglass by loud singing is accomplished by Helmholtz resonance with the glass as a resonator.

True ☐ False ☐

21.9 A damped oscillator oscillates at its natural frequency, independent of the drive frequency under a sinusoidal drive force.

True ☐ False ☐

21.10 Resonant driving results in large amplitude because power dissipation is minimized on resonance.

True ☐ False ☐

21.11 The Q of a typical 1000 Hz tuning fork is approximately

10 ☐ 1,000 ☐ 100,000 ☐

21.12 Running your finger along the teeth of a comb inside a tube to find the resonances of the tube is an example of an impulse response measurement.

True ☐ False ☐

21.13 When we push a child on a swing, if we use the same force for the same time for each little push we give, the child will go higher if we give the impulses when the swing is exactly vertical, rather than if we give the impulses as usual when the swing is just accelerating after a stop at the top of its arc.

True ☐ False ☐

21.14 A system driven by a perfect sine wave with frequency *f* always oscillates at the drive frequency (assuming it is physically possible to do so).

True ☐ False ☐

21.15 For a high impedance drive with many partials in its power spectrum, the system being driven can cause there to be more power output at some frequencies than others.

True ☐ False ☐

21.16 The Jew's harp is an interactive drive whose frequency depends on the player's mouth configuration.

True ☐ False ☐

21.17 A church bell and a wineglass are examples of systems with a high Q.

True ☐ False ☐

21.18 Let's consider two similar pendulums: one is oscillating in air, the other in oil. Which one has the lowest Q?

Pendulum in air ☐ Pendulum in oil ☐

21.19 A system with a high Q loses energy faster than a system with a low Q.

True ☐ False ☐

21.20 Does a drive have to be precisely tuned to drive a tuning fork?

Yes ☐ No ☐

21.21 Is the Q of Jello high or low?

High ☐ Low ☐

Helmholtz resonator

21.22 For fixed neck parameters, the frequency of a Helmholtz resonator depends on its volume and not on its shape.

True ☐ False ☐

21.23 The frequency of a Helmholtz resonator will go up when more holes are put in its side.

True ☐ False ☐

21.24 A bottle is filled half-full with water and you blow over the top, producing a tone. When you pour out the water and blow again, the pitch is lower by one octave.

True ☐ False ☐

21.25 Uncovering a hole on an ocarina makes the pitch go up because the neck area increased without changing the neck length or volume of the resonator.

True ☐ False ☐

Membranes & shells

21.26 The distance between nodal lines on a metal plate vibrating at 1000 Hz is 6 cm. The speed of the surface wave is higher than the speed of sound in air.

True ❐ False ❐

Turbulence

21.27 Let's consider a wooden electrical pole. One day, wind is blowing fast, at 100 km/h. We hear a tone at 50 Hz. The next day, we hears a tone at 60 Hz. How fast is the wind blowing then?

❐ 80 km/h
❐ 110 km/h
❐ 120 km/h

21.28 Following up on question 21.27. On a third day, next to the electrical pole, we hear a tone at around 400 Hz, with a wind blowing at 20 m/s (i.e. 72 km/h). This tone is due to the vortex shedding around the electrical wires, and we tell you that the Strouhal number for this kind of electrical wire is 0.4. What's the diameter of the electrical wire?

❐ 1 centimeter
❐ 2 centimeters
❐ 4 centimeters

Musical Instruments

22 – Problem: Good vibrations

Read in Why You Hear What You Hear

- Chapter 16: Wind Instruments

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- a sound file: **Fsharp-carbon-alphorn.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

In this problem, we are looking at some excerpts from Barry Parker's book "*Good Vibrations – The Physics of Sound*", first edition, 2009.

Open-end tube

In Chapter 4 "**Making Music Beautiful – Complex Musical Tones**", in the section "**Vibrational Modes of a Column of Air**", the author describes the model of the cylindrical tube that we use to study the behavior of instruments such as, for instance, the flute.
The following sentences are copied from the book, page 72:

> *"Let's begin with a cylinder with two open ends. In an open-end tube, if a compression is introduced into one end of the tube, it will reflect as a compression; in other words, there will not be an inversion. Now, suppose that exactly when the reflection occurs at the far end, we introduce a rarefaction at the near end. This rarefaction will travel down the tube and interfere with the reflected compression; in particular, the two waves will interfere destructively at the center of the tube, and there will be a node. In this case (the fundamental), half a wavelength fills the length of the tube, and since we always have an alternating pattern of nodes and antinodes, the two open ends will have antinodes (fig. 49)."*

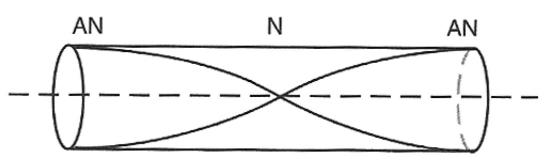


Fig. 49. Fundamental waves in a tube with two open ends.

Figure 49: Fundamental waves from *Good Vibrations* (Fig. 49 in the book)

22.1 Find the error(s) in the second sentence and suggest a corrected version.

22.2 Where is/are the node(s) of the first mode of a standing wave in an open-end tube situated? Suggest an alternative to the third and fourth sentences of the paragraph, with your own words.

22.3 In the case of the first mode of an open-end tube, which proportion of the wavelength fills the length of the tube? Suggest an alternative to the last phrase of the paragraph (keep it short) and a new version of Fig. 49, including title.

Alphorn

In Chapter 10 "**The Brass Instruments**", page 166, we read:

> *"In Switzerland, (…) the first horns were made from trees, and were called alphorns; they*

were particularly long, sometimes as long as five meters and produced only very low notes. They were used mainly for calling the cows home in the evening so would hardly qualify as a musical instrument."

Let's use this remark as a trigger to study this traditional mountain instrument.

The alphorn is a wooden horn of conical bore, having a wooden cup-shaped mouthpiece. The sound is produced by the vibration of the performer's lips.

The traditional alphorn is in the key of F#. That means that the first resonant mode has a frequency of 46.25Hz (F#1) (in conditions when the speed of sound is 344 m/s).

22.4 What is the approximate length of the resonant part of this standard alphorn?

22.5 The standard range used by alphorn players includes around 17 notes. List the frequencies and approximate corresponding musical notes of the modes number 2 to 17. You may present your results are a table, or write the mode number and frequency linked to notes on a music staff.

22.6 Let's call "very low notes" the notes that are in the lowest third of the piano range, i.e. the notes which fundamental frequency is less than 146 Hz. How many modes correspond to the fundamental frequency of a "very low note"?

22.7 The company SwissCarbonAlphorn (http://www.swisscarbonalphorn.com/)is marketing a carbon alphorn. Do the acoustics of the instrument depend on the material? Would you agree with a player telling you that the carbon instrument's sound is less "warm" than the wooden instrument? Comment.

22.8 The modularity of the carbon alphorn makes it easy to tune the instrument, using different parts to build the whole instrument. To transform a horn in F# into a horn in G, the tube length must be shortened; by which proportion of the original length?

22.9 Using the sound file **Fsharp-carbon-alphorn.aif**, produce a nice sonogram on which we can clearly identify all of the different pitches produced (no need to include repeated pitches). Measure the fundamental frequency of each of the different musical notes (you might use **Audacity**'s **Spectrum view** or **Autocorrelation view** for each note). Report the values on your sonogram (for instance, using arrows to point the fundamental frequency to the corresponding notes on the sonogram).

22.10 Compare your sonogram to the list of modes from question 22.5. Is there a note appearing in the sonogram that is not a resonance mode of the instrument? Comment.

Clarinet

Let's read now an excerpt from Chapter 11 "The Wind Instruments", page 183, we read about the clarinet:

"*(…) the clarinetist uses what is called the speaker key – a small hole on the bottom side of the instrument, usually about 15 cm from the mouthpiece. When the speaker key is open, it excites the third harmonic (fig. 99).*"

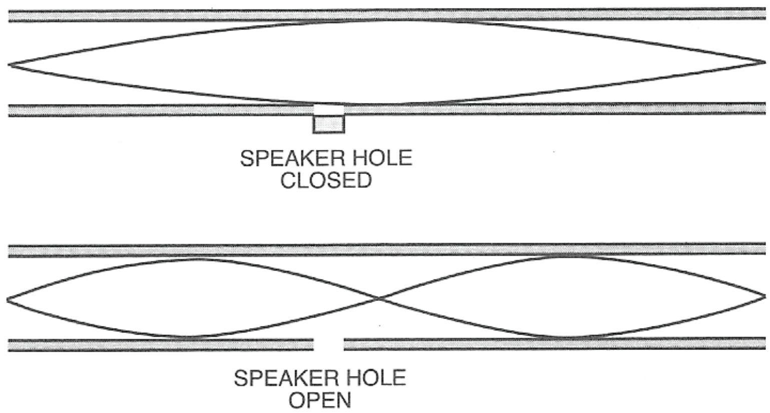


Fig. 99. The speaker key (hole) in the bottom of the clarinet.

Figure 50: Speaker key from Good Vibrations (Fig. 99 in the book)

22.11 The speaker key is also known as the register key. It enables to play easily the note corresponding to the second resonant mode of the instrument. What is the author talking about when he mentions "the third harmonic"? Comment.

22.12 The figure includes several major errors, both in the drawing of the tube, and in the drawings of the waves. Provide a new corrected version.

23 – Problem: Smooth & sharp sax

Read in Why You Hear What You Hear

- Chapter 16: Wind Instruments

Tools

- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)
- **Harvard Falstad Ripple uploading tool** (on http://www.courses.fas.harvard.edu/~icgzmod/java_physics/saved_state/upload.html)
- a file with saved applet state: **rippleState_2013-01-04_09.36.43_213.228.61.104.txt** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

Pipes in musical instruments generally use smooth curves rather than sharp corners. Below are two possible saxophone shapes.

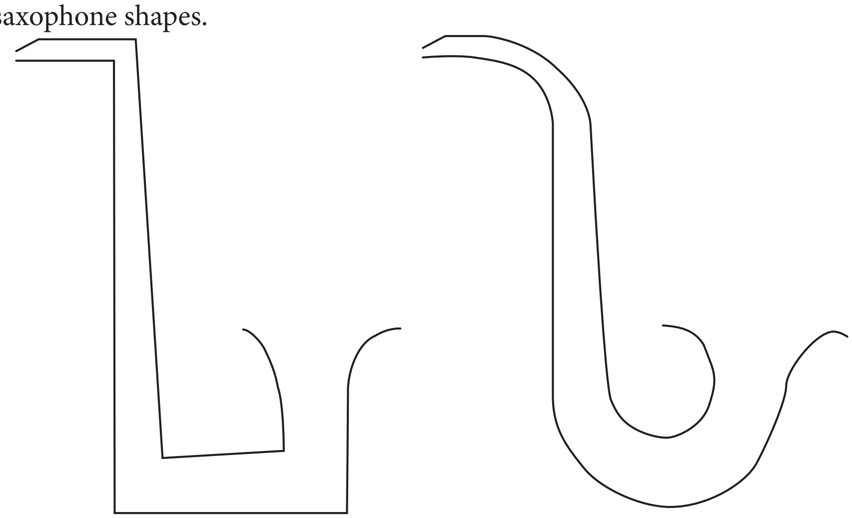


Figure 51: Two saxophone shapes

23.1 Starting from a source just inside the mouthpiece in the upper-left of each instrument, use ray tracing to show how sound waves travel through them. Draw 3 to 5 rays, and follow them until they either leave the instrument or are reflected 15 or 20 times. Use different colors if needed to make the rays easily distinguishable. Based on this analysis, which design appears to be better? Why?

We've implemented these two instruments in the saved **Harvard Falstad Ripple** simulation. Load this simulation with the uploading tool.

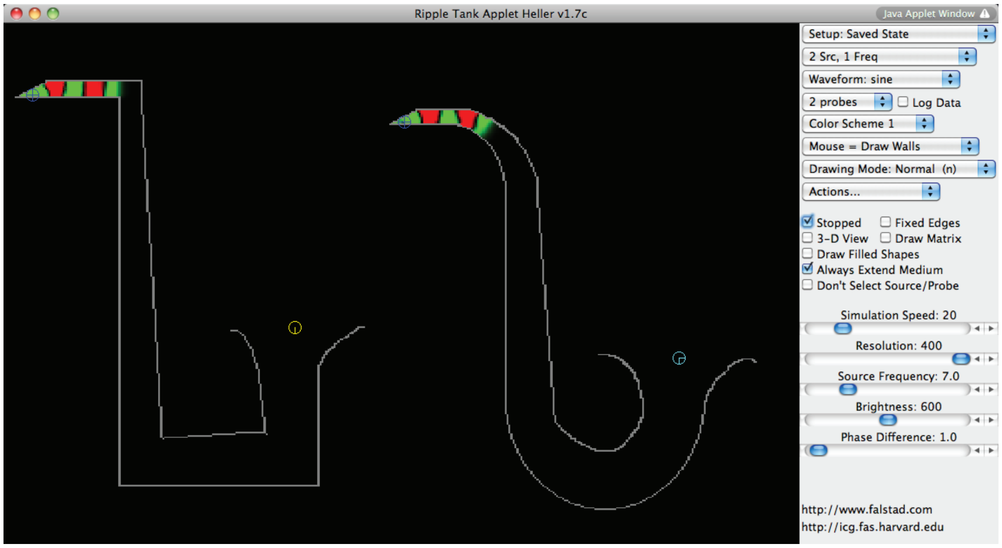


Figure 52: Wave propagation in the two saxophones

23.2 Start running it with a frequency of 7.0. Observe the amplitude of the sound emitted from each instrument using the two probes. Do the results agree with your ray-tracing prediction?

23.3 Now lower the frequency from 7.0 down to 3.0, stepping by 1.0 and clearing the waves after each step. What happens as you lower the frequency? Do the results agree with your ray-tracing prediction now? What phenomenon caused this change in behavior?

23.4 Suppose you could shrink your body to fit inside the mouthpiece of each instrument. If you clap your hands there, in which instrument would you hear a greater number of echoes (assume that the shrinking process also allows you to hear with excellent time resolution)? How might this fact affect the sound of one instrument versus the other when played?

24 – Problem: Clarinets

Read in Why You Hear What You Hear

- Chapter 16: Wind Instruments
- Chapter 17: Voice (especially section 17.1: "Tubes that change diameter or shape")

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- **Praat** (details on http://www.fon.hum.uva.nl/praat/)
- 10 sound files: **contra-C-reed3.aif**, **contrabass-clarinet-sub-to-norm.aif**, **bb-clarinet-split.aif**, **bb-clarinet-multi-1.aif**, **bb-clarinet-multi-2.aif**, **bb-clarinet-multi-3.aif**, **bb-clarinet-multi-4.aif**, **bb-clarinet-multi-5.aif**, **bb-clarinet-multi-series.aif**, and **bb-clarinet-octave-fade.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

In this problem set, we are going to explore a few unusual clarinet sounds. First, we are focusing on a rather rare instrument in the clarinet family: the contrabass clarinet. Second, we are going to examine multiphonics on the usual clarinet: multiphonics are sounds played on a monophonic instrument, like the clarinet, but where listeners can perceive several distinct pitches at the same time.

Contrabass clarinet

The contrabass clarinet sounds two octaves lower than a standard clarinet. For the recordings in this problem set, we used a metal model made by French clarinet maker Leblanc. Anthony Braxton plays a Leblanc contrabass clarinet:

Figure 53: Anthony Braxton, Rochester, N.Y., 1976
http://www.flickr.com/photos/tommarcello/1865290326

The sound file **contra-C-reed3.aif** is a recording of a low concert C played on the contrabass clarinet, the same note as the lowest C on a grand piano. If the instrument were perfectly tuned to an A4 at 440 Hz, the fundamental frequency of this C would be 32.7 Hz.

24.1 Using the sound's waveform, find the signal's actual period and frequency.

24.2 Produce an auto-correlation plot for the signal, and check the first high peak in the auto-correlation corresponds to the period.

24.3 What's the length of the clarinet tube assuming that the fingering to produce this note shuts all possible holes on the instrument?

24.4 Looking at the waveform, find if the clarinet reed corresponds to the model of the stopped reed (oscillating one side of its rest position) or of the free reed (oscillating both sides of its rest position).

There are different definitions for a "subtone" sound on a woodwind instrument. Here, the sound file **contrabass-clarinet-sub-to-norm.aif** contains a recording of a low C played from "subtone" to "normal". At the beginning, the clarinetist touched the reed with his tongue, trying to mute as much of the sound as possible, while still producing a tone. Then, he slowly slid his tongue down the reed, the tongue being in contact with smaller and smaller a surface of the reed. Finally, the tongue didn't touch the reed anymore, and the clarinetist tried to play a sound as "full" as possible.

24.5 This is a sonogram of the sound. Interpret the picture in light of the above explanation.

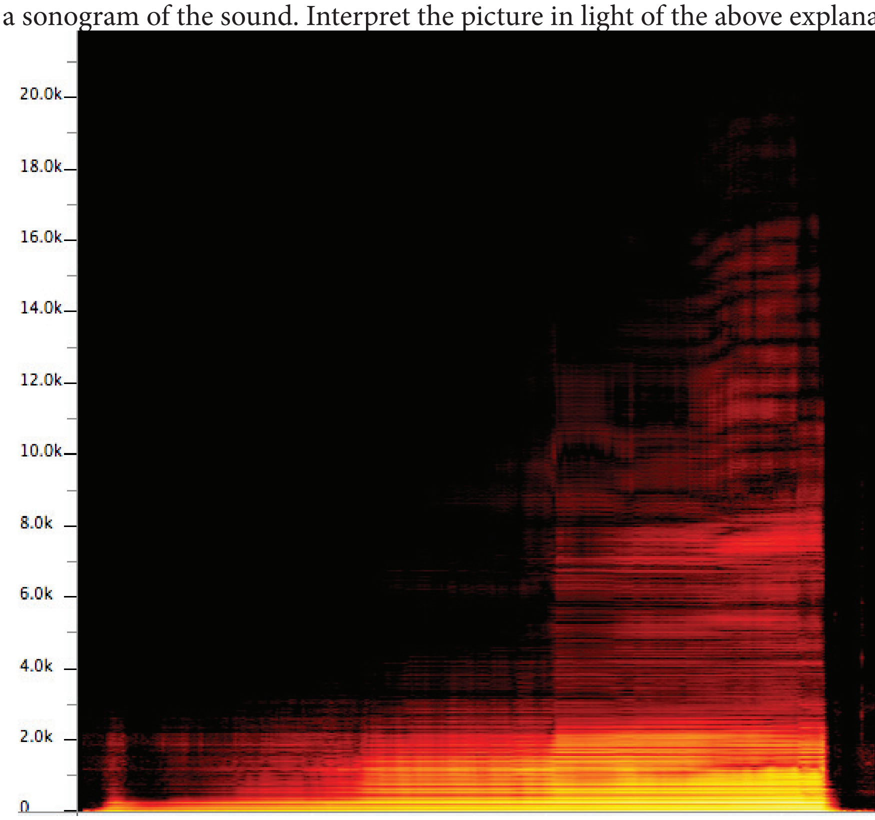


Figure 54: Sonogram of a contrabass clarinet sound from subtone to full sound

24.6 Using the same sound file, make two power spectrums: one from the beginning, one from the end of the sound, approximately where the two arrows are pointing:

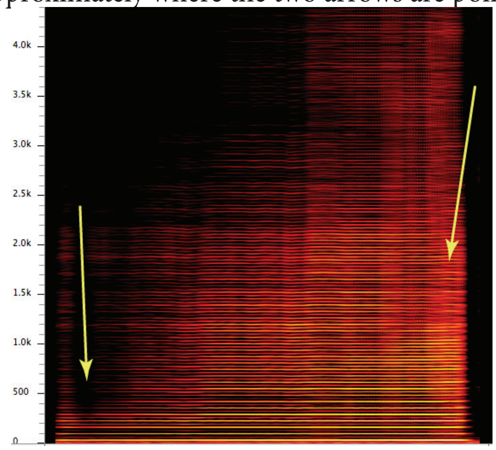


Figure 55: Sonogram, zoomed on

24.7 Compare the presence and intensity of odd and even partials in the power spectrums. As you know, in an ideal clarinet, the resonance peaks match odd partials. Take a guess at what could explain the presence of even harmonics – you might find clues in: http://www.phys.unsw.edu.au/jw/z.html

Bb clarinet

Making an instrument sound in tune is a complex task. Although the model for a clarinet is a cylindrical tube, clarinet makers refine the inner shape of the tube to help tuning in different ways. Ernest Ferron, a French clarinet maker, writes in his book *Clarinette, mon amie* (Published by IMD International Music Diffusion, in Paris, 1994):

"to tune ONE note that is too flat, you have to narrow the diameter on a pressure anti-node."

Here is a reproduction of figure 17.1 in *Why You Hear What You Hear*, section 17.1:

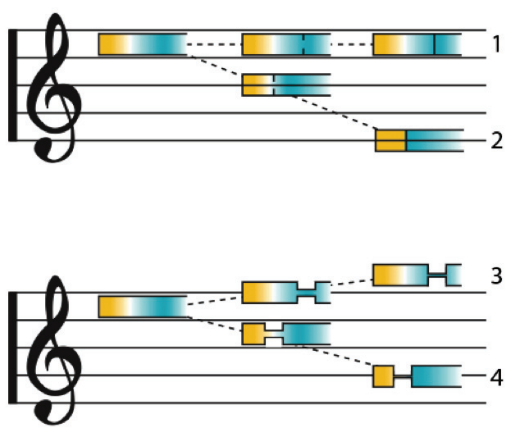


Figure 56: Effect of constrictions in tubes (from *Why You Hear What You Hear*).

24.8 With the method proposed by Ernest Ferron, are we in one of the configurations illustrated, and in which one (1, 2, 3, or 4?)

24.9 Ernest Ferron actually suggests two methods. The complete sentence goes like that: *"to tune ONE note that is too flat, you have to either widen the diameter on a ______________________, or narrow the diameter on a pressure anti-node."*
Choose the correct option to fill in the blank:

Velocity anti-node ☐ Velocity node ☐

Multiphonics

Using special techniques, woodwind players can play sounds that seem to contain several pitches at the same time. These sounds are called by many musicians "**multiphonics**".
A first type of multiphonic is often called split tone by clarinetists, also known as "overblown" multiphonic. By modifying the way s/he plays (air pressure, vocal track shape), a clarinetist emphasizes the presence of many partials.
Composer Iannis Xenakis calls for this technique in his composition ***Charisma for clarinet and cello***. He writes in the score: "Harm. Zone I, II, III, and IV".

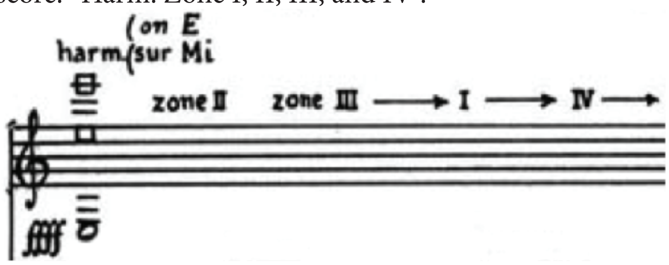


Figure 57: Charisma, by Iannis Xenakis - score excerpt

He asks the clarinetist to play a unique split tone in four different ways, emphasizing different "Harmonic zones" or harmonic regions.
The sound **bb-clarinet-split.aif** illustrates this technique (the clarinetist is not playing the score above, but is playing with the same technique).

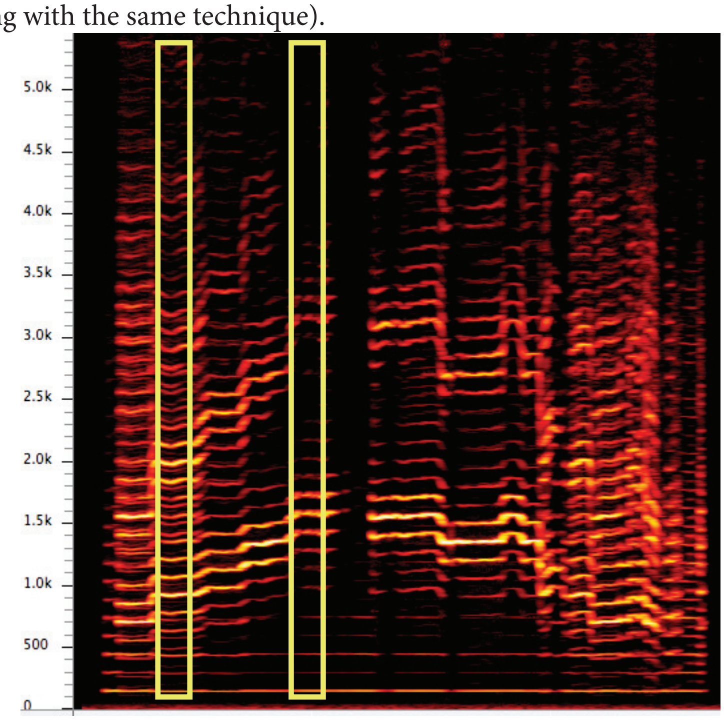


Figure 58: Sonogram of the sound **bb-clarinet-split.aif**

As you can see on this sonogram, when the clarinetist emphasizes harmonic regions, s/he actually moves formants. This is similar to moving formants in voice production. It makes sense, since the clarinet player uses a lot the shaping of her/his vocal track in this playing mode.

24.10 Use the software **Praat** to analyze this sound file. Here is an plot showing three formants for the second half of the sound file:

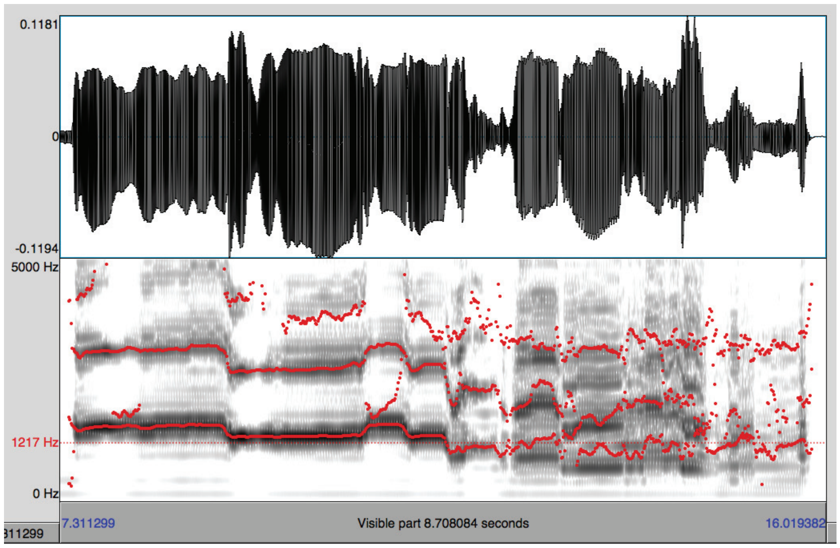


Figure 59: Clarinet split tone - formants

Generate an equivalent plot for the first half (the first half contains the yellow boxes on the sonogram above).

24.11 For each of the places in the two yellow boxes in the sonogram above (Figure 58), make use of the commands under the **Formants** menu to get the center frequency and bandwidth of the first two formants. You may present your results in a table:

	(first yellow box)	(second yellow box)
First formant frequency		
First formant bandwidth		
Second formant frequency		
Second formant bandwidth		

After the split tone, let's explore now a second type of multiphonic sound. Let's consider a clarinet fingering such as a low concert F, with most holes on the clarinet closed:

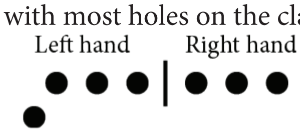


Figure 60: Concert F fingering (on a Bb clarinet, this note is called a G)
fundamental frequency is 174.6 Hz

Now, let's tweak this fingering: if we raise the left middle finger, two things happen:

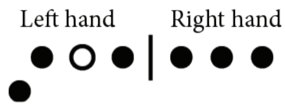


Figure 61: Same fingering with open hole in left hand
this hole acts both as a terminator and a register hole

Clarinetist E Michael Richards' explains the dual role of such a hole on his web site (http://userpages.umbc.edu/~emrich/chapter3-2.html) and quotes Ronald Caravan's 1974 dissertation:

> *"Caravan has labeled this hole the register-terminator hole. The register-terminator hole performs two simultaneous functions: it terminates the shorter tube and acts as a register opening or vent for the longer tube."*

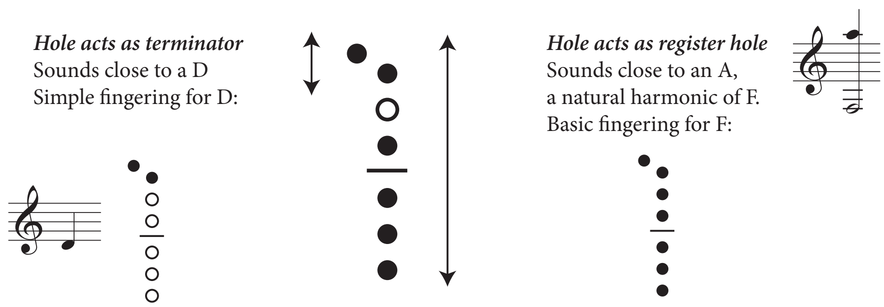


Figure 62: Illustration of both roles played by the "unusual" hole in the multiphonic fingering

Here are four note aggregates, along with the theoretical fundamental frequencies of each written note. These aggregates were found in a list of clarinet multiphonic fingerings. The sign in front of the bottom note in chord B is a quarter tone, an interval of half a sharp, raising the pitch of the written note by 25 cents.

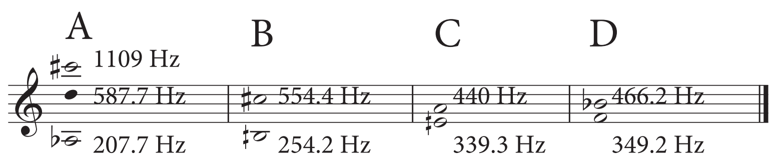


Figure 63: Score of 4 clarinet multiphonic sounds

24.12 Let's explore the sound **bb-clarinet-multi-1.aif**
This sound corresponds to the set of notes "C" on the score above (Figure 63).
On a power spectrum, try to find the two peaks corresponding to the fundamental frequencies of the notes. The written notes are approximations, so it's normal if you find some difference between the experimental result and the frequencies written on the score above. Write the frequency values on your power spectrum.

24.13 On an auto-correlation plot, try to see if these same peaks are apparent. Interpret.

24.14 Match each sound with the corresponding note aggregate. You will probably need to produce the power spectrum or the auto-correlation of the chords.

✓ **bb-clarinet-multi-2.aif** ________
✓ **bb-clarinet-multi-3.aif** ________
✓ **bb-clarinet-multi-4.aif** ________

In a clarinet method, we see this multiphonic:

Figure 64: Clarinet multiphonic with a "ghost" tone

The author explains that both the note at the bottom and the note at the top can be isolated, i.e., using the multiphonic's fingering, a clarinetist can play in isolation either of these notes. But the author calls the black note a "ghost note", because it is impossible to isolate. This black note sounds only when the clarinetist plays the two white notes at the same time.
The question you have to answer is the following:

24.15 Is this note physically created in the air, or is this note created in the brain of the listener?
You should be able to answer the question by observing the power spectrum of the sound. Sound file is **bb-clarinet-multi-5.aif**

One hypothesis is that a non-linear phenomenon similar to ring modulation produces the non-isolable note. Ring modulation is known as an effect in electronic music – it consists in the multiplication of two signals (multiplication of the waveforms). When the two inputs of a ring modulator are fed with two sinusoidal signals, the device outputs the sum and the difference of the frequencies. A similar operation is performed when the two signals are not sinusoidal, the sum and difference applying to each pair of sinusoidal partials.

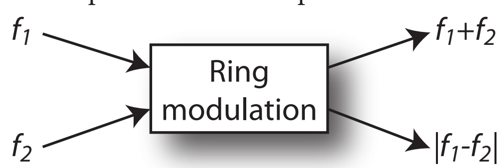


Figure 65: Effect of a ring modulator, producing difference and summation tones

24.16 In the case above, check if the non-isolable (black) note's fundamental frequency is close to the difference of the fundamental frequencies of the two other (white) notes. Check as well if there is a partial corresponding to the sum of these frequencies.

And now, let's brace for a challenging extra-credit question!
We want to check the "ring modulation" hypothesis for the sound **bb-clarinet-multi-series.aif**, a rough recording of a series of multiphonic sounds presented in Example #2 on http://userpages.umbc.edu/~emrich/chapter3-2.html
In concert pitch, notes on the score are approximately:

Figure 66: Series of 8 clarinet multiphonics

24.17 To check the hypothesis that the black, non-isolable notes, are the result of a ring modulation occurring between the two white notes, you can:

✓ read on the power spectrum of each multiphonic the actual peaks for the white notes
✓ compute the sum and difference frequencies
✓ see if a peak is present at the sum and difference frequencies.

As mentioned earlier, the score is indicative, actual results are likely to differ from the score values. Note that if a tone is present at the sum frequency, it was not indicated by the author on the score above.

Un-clarinet sound

Finally, we want to share with you the impossible clarinet sound.

A characteristic of a clarinet sound is that the octave partial is absent. But if the clarinetist finds a special fingering with which s/he can play a multiphonic sound made of a fundamental and its octave, s/he will produce a un-clarinet sound… Here is such an example, played with a fade into the octave. Sound file is **bb-clarinet-octave-fade.aif**

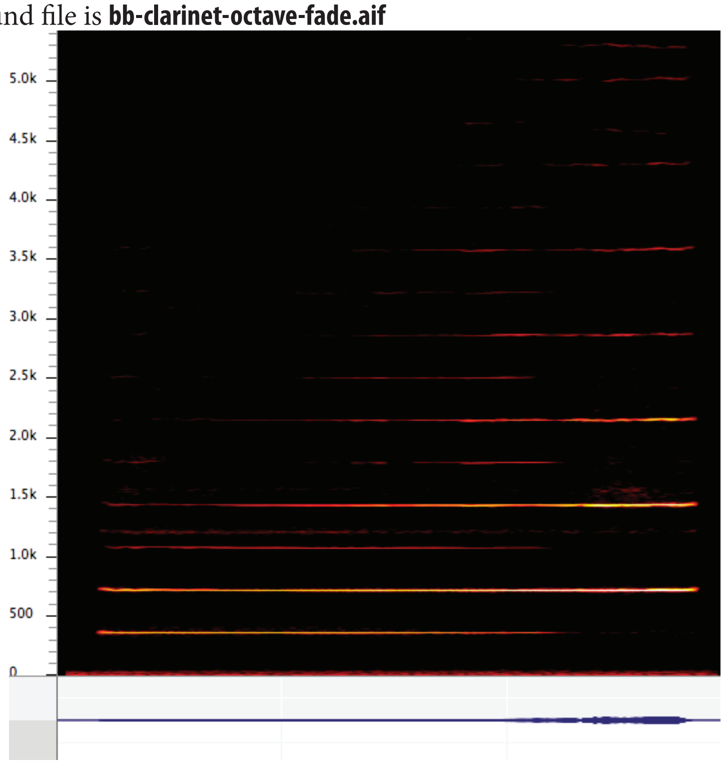


Figure 67: Clarinet sound with octave harmonic!

25 – Problem: Mary had vowels

Read in Why You Hear What You Hear

- Chapter 17: Voice

Tools

- **Praat** (details on http://www.fon.hum.uva.nl/praat/) or **Sonic Visualiser** (details on http://www.sonicvisualiser.org/) or another tool of your choice

Mary Had a Little Lamb

25.1 Make a sonogram of your voice singing the first seven notes of *Mary Had a Little Lamb*. Then speak the same words and make a sonogram of that. Compare and comment, analyzing frequency, frequency relationships, harmonics, pitch. Make screen shots of both.

Vowels & fricatives

25.2 Using your own voice, check formant changes as you speak four different vowels, at a couple of different pitches. You should be able to see traces of the formants directly on the sonogram. Also check fricatives like the **s** in "**stay**" and the **f** in "**fine**". Show pictures of your experiments and comment.
You might be interested in Peterson's formant table. He conceived this table in 1952, averaging data from 76 speakers. F1, F2, and F3 are the center of the three most important formants he got for each vowel/speaker.

vowel		*f1* (Hz)	*f2* (Hz)	*f3* (Hz)
ee	male	270	2290	3010
	female	310	2790	3310
	child	370	3200	3730
e	male	530	1840	2480
	female	610	2330	2990
	child	690	2610	3570
ae	male	660	1720	2410
	female	850	2050	2850
	child	1030	2320	3320
ah	male	730	1090	2440
	female	590	1220	2810
	child	680	1370	3170
oo	male	300	870	2240
	female	370	950	2670
	child	430	1170	3260

Figure 68: Peterson's formant table

26 – Problem: Hollywood

Read in Why You Hear What You Hear

- Chapter 17: Voice

Tools

- **Praat** (details on http://www.fon.hum.uva.nl/praat/)
- **Audacity** (details on http://audacity.sourceforge.net/)
- three sound files: **Hollywood.aif**, **Neighborhood.aif**, **Neighborhood-boring.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

In the file **Hollywood.aif**, you can listen to a voice morphing example, an excerpt from Madonna's song *Hollywood*. In this problem, you will investigate what happens to a voice when it is shifted down in pitch, and how to fix the problems that result.

26.1 Consider the sound file **Neighborhood.aif**.
Transpose this sound down 6 semitones in **Audacity**, using the command **Change Pitch...** in the **Effect** menu.
Using **Praat**, compare both sounds, focusing on the following points:

✓ pitch on the "o" in "neighborhood": use the pitch detector of **Praat** to show that there is indeed a transposition 6 semitones down for the modified sound file.

✓ looking at this same part of the sound ("o" in "neighborhood"), were the formants transposed down?

Hints: when you want to analyze a sound with **Praat**, open the sound by dragging it onto the **Praat** icon or opening it under the **Read** menu. Then click the **View & dit** button on the right of the main window to see the soundtrace and sonogram of the file. The analysis commands are in the menus of the opened **Edit window**. In your formant analysis, it is sufficient to compare two formants (change the number in **Formants -> Formant settings...**).
For finding the fundamental frequency, the screenshot below shows parameters working well with this file. Note that this is a singing voice, not speech, so the range where we are looking for pitch value is greater than the spoken voice range.

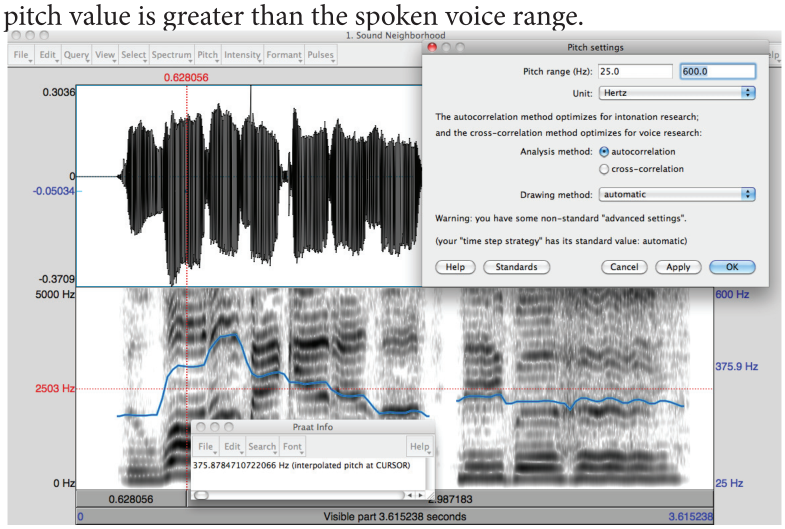


Figure 69: Praat pitch analysis with singing voice

To get the relations between note name and frequency, you may use the table at http://en.wikipedia.org/wiki/Piano_key_frequencies

26.2 You will find that the voice doesn't really sound like Madonna anymore after it is transposed down. After reading chapter 17 in *Why You Hear What You Hear*, you have the idea of moving the formants back up to make the voice sound like Madonna again.
To change the formants in a voice without other transformation, you can use **Praat**. In the main **Praat** window (not the **Edit window**), select your sound on the left, then use the command **Change Gender...** in **Synthesize -> Convert**. Keep **New pitch median** to 0.0 (no change), **Pitch range factor** to 1.0 (no change), and **Duration factor** to 1.0 (no change). Formant shift ratios greater than one move formants up, while ratios less than one move formants down. Experiment to find a formant shift ratio for which you hear a credible female voice, possibly the same voice as the original. What formant shift ratio works best for you? Compare the formant values in this file to the ones in the original file.

Now, let's study the sound file **Neighborhood-boring.aif**.

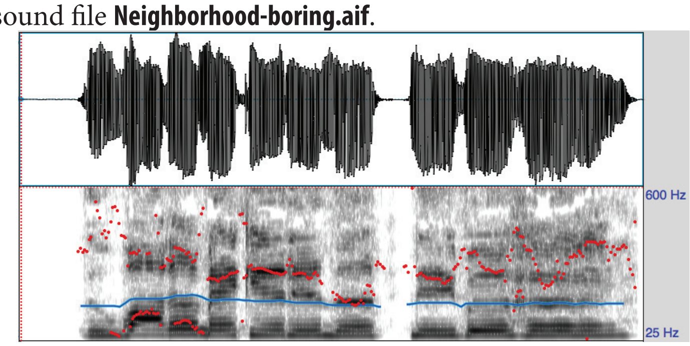


Figure 70: Sample of a **Praat** analysis - sound **Neighborhood-boring.aif**

Use **Praat** to answer the next three questions.

26.3 Considering as a reference the note for the vowel "o" in "neighborhood", was a transposition applied? If yes, tell if it is a transposition up or down, and by how many semi-tones.

26.4 Around this vowel "o", the pitch range (difference between the lowest and highest note) in the original sound file is about 12 semitones (one octave): pitches (as they appear Figure 69) range from 234 Hz (Bb3) to 470 Hz (Bb4). What is the pitch range in the file **Neighborhood-boring.aif**?

26.5 Listen to Madonna's excerpt **Hollywood.aif**. Comment on the techniques you think might have been used to create the voice morphing excerpt.

26.6 Here is a challenge question, or rather an idea for a project. Using the software tools of your choice, and source voices from friends, conceive your own scheme for voice morphing, using pitch and formant shifting of your own design.

27 – Problem: The biggest piano

Read in Why You Hear What You Hear

- Chapter 8: Making a Stretched String
- Chapter 19: Piano
- Chapter 23: Pitch Perception
- Chapter 25: Phantom Tones

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- four sound files: **Piano-01.aif**, **Piano-01-Pedal.aif**, **Piano-02.aif**, and **Piano-03.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)
- article to read: *Stiff-string theory: Richard Feynman on piano tuning*, by John C. Bryner, published in Physics Today, December 2009 (look this up on the web, or get it from a library)

When preparing this problem set, we met with a piano technician and tuner for Harvard University. Hearing the low strings of a piano, he is able to tell if it is a big or a small instrument. He even told us that for some pianos, the lowest strings "*hardly have any pitch*". You are going to explore piano strings, especially the bass ones, producing the lowest pitches. Given samples of a low note coming from different pianos, you are going to determine which instrument is the smallest.

Mechanism

When a pianist presses a key on a piano, a hammer is thrown against the corresponding string(s) and quickly bounces back automatically to let the string vibrate. Here is a reduced schematic description of the piano mechanism.

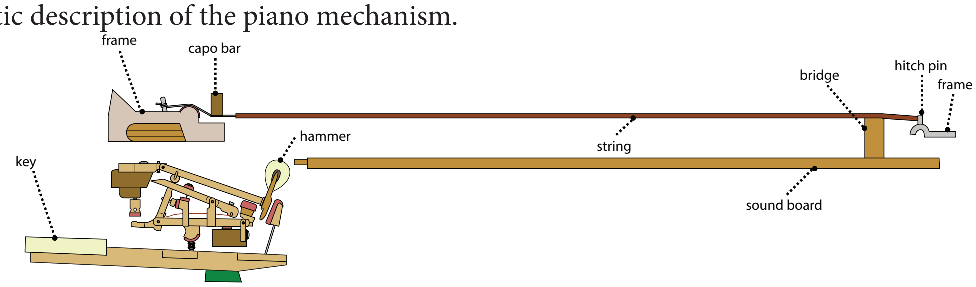


Figure 71: Piano mechanism, partial view

Piano maker

In this part, you might use the following constants and indicated measures of the piano:

✓ Steel's density: 7.80 g/cm^3
✓ Speed of sound in air: 343 m/s
✓ Speed of sound in maple wood, the one used for the wooden parts of our piano: 4110 m/s

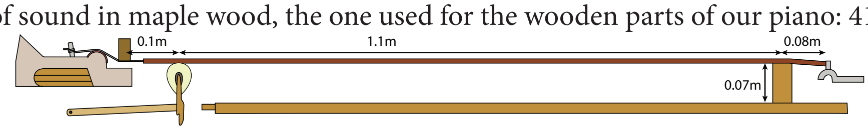


Figure 72: Piano mechanism, from hammer to soundboard

Let's consider a steel piano string of diameter 1.1 mm, under a tension force T = 680 N.

27.1 If the string is 1.2 meter long, what are the resonant frequencies of the lowest 5 modes of the string? (We consider for this question the case of an ideal string.)

27.2 At what velocity will the wave travel along the string?

27.3 What is the delay between the moment the hammer hits the string and the moment the wave puts the soundboard in vibration?

27.4 If the piano maker were to use the same wire under the same tension to make a string for a note an octave higher, how long would this string be?

27.5 The piano maker wants to study another option to realize this string, for the note an octave higher. He wants to keep the same length (1.2 m) and the same tension. What must be the diameter of the new wire?

Piano tuner

A piano technician uses a tuning wrench to tune each string of a piano by changing the tension applied on the string.

In a grand piano, for the note A2 (fundamental frequency 110Hz), the hammer hits two strings at the same time, giving more power than with one string. The piano tuner has already tuned one of the strings, and hears now some beating when hitting both strings. He hears that the pitch of the second string is too low, and he hears 5 beats per seconds.

27.6 Should the piano tuner increase or decrease the tension of the second string if he wants to make it completely in tune with the first one?

27.7 By what factor must the tension be changed? Hint: read *Why You Hear What You Hear*, section 25.4 ("**Beat tones**"), and deduce the frequency difference.

Piano player

By default, all strings in a piano are damped (except the highest couple of strings), and the string is un-damped only when its key is depressed. (The damping system has been left out of the mechanism illustration.) When depressing the pedal, the pianist un-damps all the strings of the piano; she actively opens all of the strings, which become free to vibrate or not.

27.8 Using the tools of your choice (waveform, sonogram, spectrum at some points…), analyze the recordings with and without pedal (compare the audio files **Piano-01.aif** and **Piano-01-Pedal.aif**.) Describe how they differ, and suggest an explanation for the difference. Attach all the plots that you use in your analysis.

Low sounds

We are going to study low piano strings. We suppose that we work at a sampling rate of 44100 samples per second. The fundamental frequency of the lowest pitch of the piano, an A0, is around 27.5 Hz.

27.9 How many samples do we need if we want to observe two full periods of a low A0?

27.10 To analyze low frequency sounds with a good resolution, you will surely use a rather large analysis window size (**Size** in **Audacity**). What is the drawback of using a large analysis window?

Low strings

Observe the sonograms corresponding to the sound files **Piano-01.aif**, **Piano-02.aif**, and **Piano-03.aif**, three samples of the same note played on three different pianos.

Note that not only do the power of partials fade out after the attack, but also that the intrinsic relative power of different partials evolve over time.

27.11 Choose one sound for which the proportions seem to change most. Plot the sonogram and indicate on the sonogram two places where you see different proportions in the power of partials.

27.12 Plot the power spectrum of the sound at the two places that you indicated in your sonogram (of course, you need some time to analyze these low frequencies, the place will be an approximation, but keep it as accurate as possible.)
For both places:

✓ list the 20 first partials and their relative intensity
✓ mark the most powerful peak
✓ plot the autocorrelation
✓ comment on the relation between the autocorrelation and the list of partials

27.13 Which note is the string playing?

Big & small pianos

In the article *Stiff-string theory* (look it up on the web, or get it from a library), theoretical physicist Richard Feynman proposes a formula describing the effect of wire stiffness on the vibration frequency of strings. He suggests:

$$TrueFrequency = f\left(1 + \frac{\pi \mathrm{E} A^2 \mu\ f^2}{2\,T^2}\right)$$

with *f* the frequency you would get without stiffness ("ideal string"), *T* the tension in the string, *A* the area of the string cross-section, *E* a positive constant representing the stiffness of the wire, *μ* the weight of wire per unit length.

27.14 Considering the formula above valid for all of the vibrating modes of a string, what happens to the partials of the sound when the stiffness of the string is not negligible? Are they still in a harmonic relation, i.e. multiples of the fundamental frequency? Does this phenomena increase with the stiffness of the wire?

Piano makers may use relatively thin strings in very long pianos, but if the piano must be small, they have to increase the mass of the low strings to lower their fundamental frequency. This increased mass increases the string stiffness.

27.15 The formula above is given for normal steel strings. The bass strings of the piano are wire wound strings, so it's not exactly the same formula, but it turns out that the behavior is qualitatively equivalent.
Studying the three samples with tools of your choice (sonogram, power spectrum, autocorrelation), can you tell which sample comes from the smallest piano? Can you tell which sample comes from the biggest piano? Comment and attach any plot that you mention in your comments.

27.16 Given what is said in *Why You Hear What You Hear*, how do you understand the tuner's comment that for some pianos, the lowest strings "*hardly have any pitch*"? Would you guess he was talking of the biggest or smallest pianos?

28 – Short questions

Build-a-Trumpet

Nick Drozdoff built an inexpensive natural trumpet by starting with a PVC tube, adding a mouthpiece, and finally attaching a funnel as a bell. He explains the construction in online videos, some of which are linked on http://www.whyyouhearwhatyouhear.com/subpages/chapter16.html .

28.1 A high impedance drive is placed in the end of a PVC tube of length *L* in such a way that the tube is half-open. Describe with a power spectrum the power that would be measured coming out the other end of the tube as the frequency of the drive is tuned over a large range with constant amplitude. Label frequencies in terms of *L* and *c* (speed of sound), and explain why your spectrum has the features it does.

28.2 How does adding the mouthpiece affect the spectrum and why?
Hints: what kind of resonator is it like, what are its frequency and Q?
Give the impulse response picture of the shaping of the trumpet spectrum by the resonance.

28.3 Finally, how does adding the funnel (bell) affect the spectrum and why?
Hint: think about the wavelength-dependence of reflections at the end of the trumpet with and without the funnel.

Overtones

28.4 A sound signal with no or very weak overtones is periodic at 100 Hz. On a first set of axes below left, sketch a signal *S(t)* consistent with this statement, and sketch a corresponding power spectrum *P(f)* to its right. Label all axes.

28.5 Suppose that there are many overtones or partials, with the fundamental still at 100 Hz, having formants at 500, 1200, and 1800 Hz. On a second set of axes, sketch a signal consistent with this statement, going all the way to the right at 0.02 seconds, and sketch a corresponding power spectrum to its right. Label all axes.

28.6 Now, suppose the signal is from a male singer with marvelous control of his formants, as in Tuva singing. Still singing at 100 Hz, but with lots of overtones (partials). But his formants have been adjusted to a single prominent 1000 Hz peak, with a Q of 10. On a third set of axes, sketch a signal consistent with all these traits on the left, and sketch a corresponding power spectrum to its right.
Hint: the sketch of the signal can be quite qualitative.

28.7 Could a Tuva singer lower and raise this tone continuously in principle? Explain.

29 – Quiz questions

29.1 We can play an octave harmonic on a piano by touching lightly the middle of a string while playing the corresponding key.

True ☐ False ☐

29.2 In the sound of a low piano string (for instance a C1!), the amplitude of the partials above the 10^{th} partial is negligible.

True ☐ False ☐

29.3 The lowest frequency formants are made by shaping the vocal tract approximately like a Helmholtz resonator, constricting the lips as a neck and opening the track to achieve the largest volume.

True ☐ False ☐

29.4 Constrictions along the vocal tract always raise the frequency of formants.

True ☐ False ☐

29.5 Tongue ram is a name given to a special percussive technique in flute playing. The tongue ram is produced by completely covering the embouchure hole with the mouth and forcibly sealing it with the tongue. The tone produced will sound close to

The same as the fingered note ☐
An octave below the fingered note ☐
An octave up the fingered note ☐

29.6 Placing a thin constriction in a pipe at a pressure anti-node has little effect on the resonant modes of the pipe.

True ☐ False ☐

29.7 The lowest frequency resonant mode of a conical-bore tuba 5 meters long is approximately

17 Hz ☐ 34 Hz ☐ 68 Hz ☐

29.8 The end of a trumpet bell is a pressure node.

True ☐ False ☐

29.9 On Mars, where the mostly carbon dioxide molecules making up the atmosphere are more massive than the oxygen and nitrogen molecules on Earth (we suppose that other variables such as humidity, temperature, etc. take the same values on Earth and on Mars), a clarinet plays lower than on Earth.

True ☐ False ☐

29.10 A low, solo male voice is recorded on a LP. By playing back the LP faster at the correct speed, we can realistically transform the voice into the voice of the same person singing at a significantly higher pitch.

True ☐ False ☐

29.11 A singer singing a single note with no vibrato would generate a periodic signal.

True ☐ False ☐

29.12 When we play a vinyl LP at 66rpm instead of 33rpm, we hear the recording an octave up.

True ☐ False ☐

29.13 When we play a vinyl LP at 66rpm instead of 33rpm, with a singer, the formants are transposed an octave up.

True ☐ False ☐

29.14 Formants have such a strong impact on vocal sounds because of their high Q.

True ☐ False ☐

Violins

29.15 When you bow a violin, at each moment, the bow is sliding perpendicularly to the string.

29.16 The Helmholtz wave describing a bowed violin string has that name because the shape is sinusoidal, like the waveform radiated by a Helmholtz resonator.

True ☐ False ☐

29.17 The Helmholtz wave on violin strings is a pure sinusoid.

True ☐ False ☐

29.18 When you bow a violin string, you apply some tension downwards on the string (this is true); therefore, the force on the bridge is constantly oriented downwards.

True ☐ False ☐

29.19 The violin body plays the role of a Helmholz resonator. The role of the F-holes is analog to the opening at the top of the neck of a bottle.

True ☐ False ☐

29.20 When you tape both F-holes of a violin, the body doesn't act as a soundboard any more.

True ☐ False ☐

29.21 In a violin, the violin shell plays a major role in transmitting the frequencies from the bridge to the air. The frequency at which the shell vibrates is the same as the frequency at which the air next to the shell is moving.

True ☐ False ☐

29.22 Sound short-circuiting does not happen too much with the violin mainly because:

of the shape of the violin's body ☐

of the F-holes in the violin's body ☐

of the sound post and bass bar inside the violin's body ☐

of the supersonic speed at which the wave is traveling in the violin's wooden shell ☐

The following figures are given for reference:

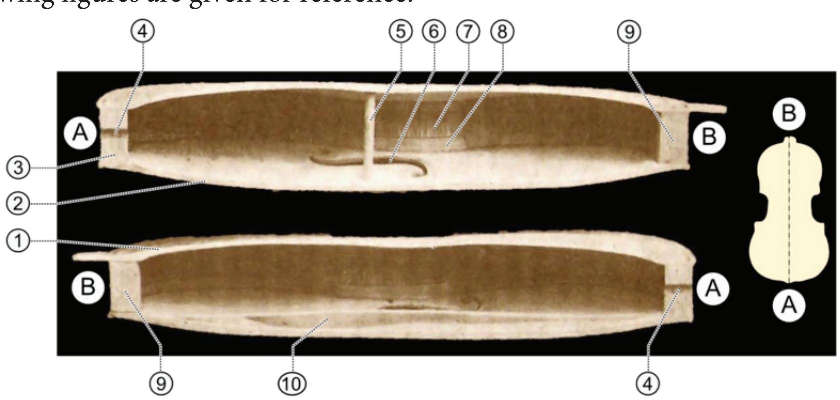


Figure 73: Inside a violin – sound post is piece number 5

Figure 74: Inside a violin – you can see the bass bar on the inside face of this violin body

29.23 Let's consider a violoncello. Its strings are thicker than the strings of a violin. Remembering what we learned about thick strings with the analysis of the piano strings (see problem *The biggest piano*), we can predict that when we bow the cello, the spectrum of the sound will be slightly inharmonic (the partials will be stretched upwards).

True ☐ False ☐

29.24 The bridge on a violin is carved to change the timbre of the instrument.

True ☐ False ☐

29.25 A piano soundboard is resonant in the sense of many sharp resonances appearing in its power spectrum as a function of frequency, through most of the piano range.

True ☐ False ☐

29.26 A viola presents a Wolf tone when a violist plays the note A played on the C string. We tell you that this same note A can be played on the G string as well. Will the Wolf tone show up when the violist plays the same A on the G string?

True ☐ False ☐

29.27 When we play a note on a violin with vibrato, the relative amplitudes of the partials change. This would not happen if the body of the violin didn't have resonance peaks.

True ☐ False ☐

29.28 The bridge of a violin transfers the vibrations of the strings to the body more efficiently at some frequencies than others.

True ☐ False ☐

29.29 Acoustical short-circuiting does not make violins quiet because sound travels faster in wood than in air.

True ☐ False ☐

29.30 The sound produced by a low piano string is inharmonic because of the thickness of the string. One way to make the sound perfectly harmonic would be to play the string with a bow.

True ☐ False ☐

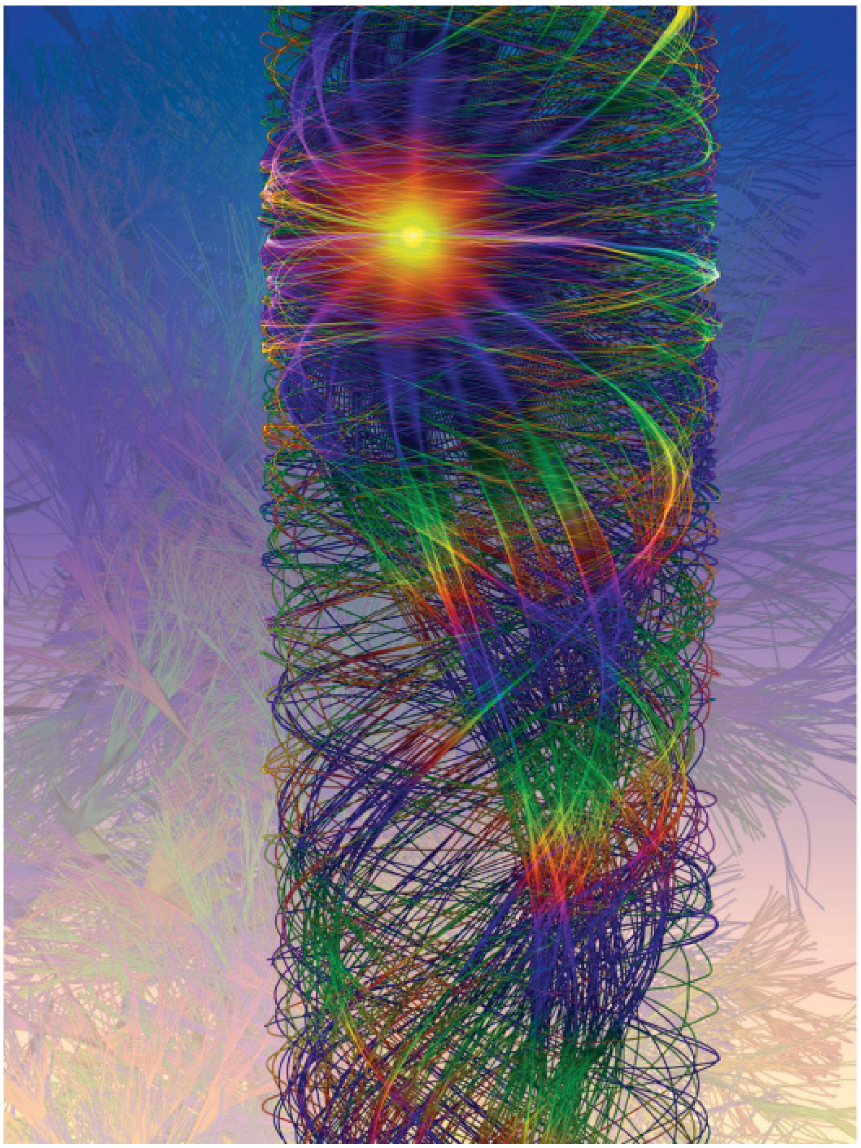
29.31 On Mars, where the mostly carbon dioxide molecules making up the atmosphere are more massive than the oxygen and nitrogen molecules on Earth, (keeping constant other variables such as humidity, temperature, etc.), a violin plays lower than on Earth.

True ☐ False ☐

29.32 A vibrating buzzer lightly touches the bridge on a violin. It has no harmonics in common with the string modes. The body will vibrate at the buzzer partials, and the strings will therefore get excited at their natural harmonics.

True ☐ False ☐

Psychoacoustics & Music



30 – Problem: Tubular bell

Read in Why You Hear What You Hear

- Chapter 12: Impulse and Power for Complex Systems
- Chapter 23: Pitch Perception

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- **Max Partials** (on http://www.whyyouhearwhatyouhear.com/subpages/MAX.html)
- A sound file: **tubular-bell.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

We are going to analyze the sound of a tubular bell, an instrument normally used in the symphonic orchestra. The sample used is a mono version of **85795__sandyrb__TUBULAR_BELL_STRIKE_001.wav**, a sound file made available on http://www.freesound.org

30.1 Analyze the sound with **Audacity**. List the most important peaks in the spectrum. Give an annotated power spectrum showing the frequency values of the first most important peaks (10 peaks or more). Is the signal audio-periodic?

30.2 Plot the autocorrelation function corresponding to the spectrum. What is the first important peak?

30.3 The bell was labeled as playing the note A4. Comment on your results in questions 30.1 and 30.2.

30.4 In **Max Partials**, make a new sound with the same frequency values you found in question 30.1 (at least 10 first values), and experiment with relative amplitudes for these partials. Look at the autocorrelation section of the patch. Playing with the amplitudes, can you replicate the peak you found in question 30.2? How does it sound? Comment and attach a screenshot if necessary.

31 – Problem: Missing fundamental

Read in Why You Hear What You Hear

- Chapter 3: Sound and Sinusoids
- Chapter 4: The Power of Autocorrelation
- Chapter 23: Pitch Perception

Tools

- **Max Partials** (on http://www.whyyouhearwhatyouhear.com/subpages/MAX.html)
- **Audacity** (details on http://audacity.sourceforge.net/)

Max Partials allows you to explore the residue pitch, AKA "missing fundamental", effect.
Build a complex tone with frequencies separated by 200 Hz, starting with 600, 800, 1000, 1200 and equal amplitudes. Can you hear the "missing fundamental" at 200 Hz? It may be helpful to create at the same time a 200 Hz tone and turn it on and off so you know what to listen for. However, bear in mind that you will not hear the 200 Hz partial specifically, but rather the pitch, of 200 Hz in the residue pitch effect. Turning off the sound then back on can help, as well as trying to sing the "pitch" of the composite sound.
Save the sound (with the 200 Hz tone off), open it in **Audacity**, and look at the autocorrelation.

31.1 Show us your autocorrelation plot and note how high the first prominent peak is.

31.2 Now create a complex tone with a dozen tones separated by 200 Hz starting at 600 Hz with the same amplitudes as before. Does the missing fundamental effect seem stronger or weaker to you?

31.3 Look at the autocorrelation of this sound. Show us your plot and note the height of the peak at 200 Hz. Does this agree with what you heard in **Max Partials**?
Note: The reason for equal amplitudes is so you can more consistently compare the autocorrelations in the two cases, but play around with the weighting of the partials – you may find that different weights make the effect stronger and weaker to your ear.

32 – Problem: Limits of hearing

Read in Why You Hear What You Hear

- Chapter 21: Mechanisms of Hearing
- Chapter 22: Loudness

Tools

- **Max Partials** (on http://www.whyyouhearwhatyouhear.com/subpages/MAX.html)
- A tool of your choice to produce a sonogram
- A sound file: **clap.wav** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

32.1 The file **clap.wav** is a perfectly periodic signal (as long as it lasts) with a period of only 1 Hz. Why you can hear it? Explain, using a sonogram of the sound.

32.2 Do you think you could hear a sinusoidal signal (pressure wave arriving at your ear) of 10 Hz? What about a 10 Hz square wave? Explain.

32.3 Using the residue pitch phenomenon, and sinusoidal waves of 20, 30, 40, 50, 60 etc. Hz, could you hear a 10 Hz sound, i.e. 10 Hz sinusoidal partial? If not a partial at 10 Hz, could you hear a "tone" at 10 Hz? Can you hear a pitch of 10 Hz? Make such a sound using **Max Partials**. Explain the issues and your conclusion. Are you really hearing 10 Hz or rather counting pulses?

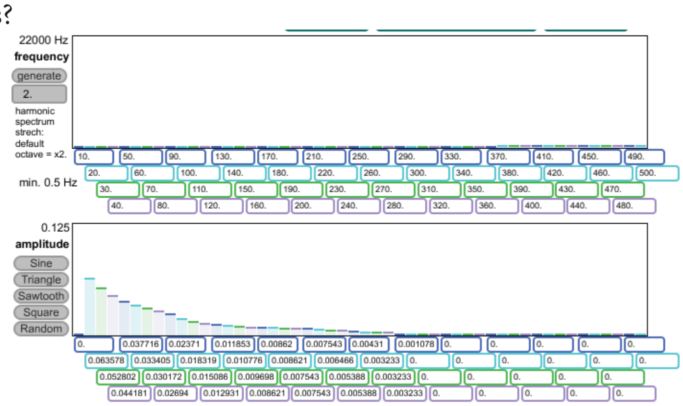


Figure 75: Zoom on **Max Partials** set-up for a "10Hz residue pitch"

33 – Problem: The great train frequency shift

Read in Why You Hear What You Hear

- Chapter 23: Pitch Perception (especially section 23.17 on repetition pitch)

There is a striking demonstration of periodicity pitch in this video of a train getting closer and then farther away: http://fabilsen.home.xs4all.nl/loc_RP.mov . There are two paths the sound can take to get from the train to your ear. One is the direct path, the other is the path that bounces off the ground in between. Clearly, the total distance of the bouncing path is longer so you get a time delay in the signal.

33.1 Right now, the straight-line distance from you to the train is $2d$ meters. If you are $h = 2$ meters tall, how much farther does the signal that bounces off the ground travel than the one that goes straight from the train to your ear? We assume the sound source is at the same height as your ear, so this distance $2d$ is just the straight line from the train to your ear, a line parallel to the ground. We assume the bouncing signal bounces exactly halfway between you and the train. It might be a good idea to make a diagram of the relevant geometry showing you, the train, and the two paths the sound can take.

33.2 Sound travels 330 m/s. How much longer does it take the sound to travel along the bouncing path? Equivalently, how much time is the bouncing wave delayed by? If the train is putting out white noise, what repetition pitch does this time shift give?

33.3 Say the train is coming toward you, and starts 12 m up the track. You are standing 5 m away from the tracks. How far away is the train from you? Make a diagram of the relevant geometry showing you and the train.

33.4 How long is the bouncing wave delayed when the train is 12 meters up the track? What periodicity pitch does this correspond to? After the train has passed you and is 12 m down the track the other way, what is the pitch you hear?

33.5 What is the lowest repetition pitch you hear from the train? Where is the train when you hear it?

34 – Problem: Staircase reflections

Read in Why You Hear What You Hear

- Chapter 2: Wave Phenomenology
- Chapter 23: Pitch Perception (especially section 23.17 on repetition pitch)

Someone stands some distance from a staircase as shown below and claps their hands once. They hear echoes coming back from the stairs. Each step is d = 25 cm wide.

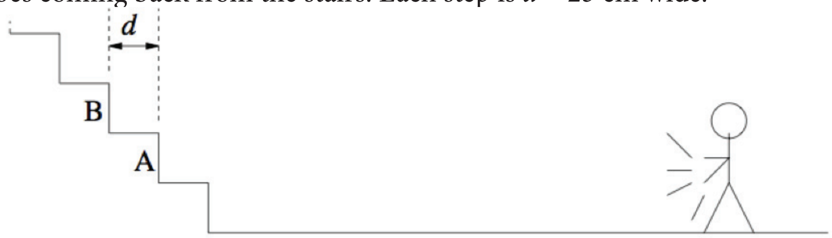


Figure 76: Set-up for Staircase reflections

34.1 We are interested in rays that return to the person (so they can hear them). Draw two such rays, one bouncing off step A and another bouncing off step B.

34.2 Which path (A or B) is longer, and by how much (ignore any vertical distances traveled)?

34.3 It is a cool day and the speed of sound is 340 m/s. What is the time delay (in milliseconds) between the echoes from steps A and B?

34.4 In reality there are *many* steps in the staircase, so the echoes will be (approximately) a periodic signal. What frequency will this signal have? In fact, higher up the staircase, the time delay is not quite the same because of the change of angle as the sound heads up and back down to the person below. Without doing any calculations, does this effect cause the echoes from successively higher steps to bunch together or spread apart?

34.5 Now the person stands a bit closer to the staircase. Will the frequency change? Why or why not?

34.6 What is the *lowest* frequency the person could sing such that the echoes from A and B interfere *destructively*?

34.7 What note (fundamental frequency plus harmonics) could be sung in hopes of hearing a false octave effect? Hint: precisely this calculation, only with trees rather than with stairs, enters into chapter 28 of *Why You Hear What You Hear* and the correct explanation of Lord Rayleigh's octave higher echo he reported in Nature magazine, giving the wrong explanation.

35 – Problem: Siren

Read in Why You Hear What You Hear

- Chapter 7: Sources of Sound
- Chapter 23: Pitch Perception

Tools

- **Mathematica** (or equivalent tool)
- **Max Siren** (on http://www.whyyouhearwhatyouhear.com/subpages/MAX.html)
- **Audacity** (details on http://audacity.sourceforge.net/)

35.1 Show that by adding two sinusoids at 300 Hz, it is impossible to get a 600Hz signal, regardless of the relative phases and amplitudes of the sinusoids. We suggest you use **Mathematica** (or other tools), and give meaningful examples. But if you are mathematically inclined, feel free to give a formal proof.
Note: when we scale both sinusoids by the same amount, or when we shift the phase of both sinusoids by the same phase value, we just need to show that for any a and b, $\sin(2\pi f)+a\sin(2\pi f t+b)$ is a sinusoid with the same frequency.

Now, let's consider two periodic signals at 300 Hz with identical waveforms, but not restricted to a single sinusoid. First, reproduce the simulation of August Seebeck's experiment as described in *Why You Hear What You Hear* in the chapter Pitch Perception, at the beginning of section 23.14 "**Seebeck's pitch experiments**".
In **Max Siren**:

- ✓ set 2 rows of 30 holes each
- ✓ set the speed to 5 turns per second
- ✓ you should hear a strong 150Hz tone
- ✓ change the phase offset between the two rows of holes: since 6 degrees correspond to 1/60 turn, you set the phase offset to 1/60 = 0.016666.

35.2 Now that you established the settings for a 150Hz tone and a 300 Hz tone, explore the questions: "*What happens in between 0 and 6 degrees offset? Do we hear both pitches in varying degrees?*"
Experiment with the siren to answer these questions. Write a short report. You may include any plot you find necessary. Tell us what you hear. Pitch is a subjective matter, so you should not expect to get exactly the same results as someone else.
Include comments about the spectrum, either as you see it on **Max Siren**, or when you analyze the sound with **Audacity**.

35.3 To conclude this siren exploration, let's have a look at the waveform generated by each hole. Experiment with different waveforms, either the four "factory presets" or draw your own waveform. You may have an intuition about the production of more or less high partials using smoother or sharper waveforms. Check the validity of your intuition by reading the spectrum at the bottom of **Max Siren**. Take screenshots of the **Max Siren** interface with two contrasting waveforms, including the spectrum analysis. Comment.

36 – Problem: Air on scale

Read in Why You Hear What You Hear

- Chapter 23: Pitch Perception

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- **Max Noisy Scale** (on http://www.whyyouhearwhatyouhear.com/subpages/MAX.html)
- A sound file: **air.aif** (on http://www.whyyouhearwhatyouhear.com/subpages/Problems.html)

Air

The sound **air.aif** was made by blowing air, sort of whispering, not really whistling. You will hear five distinct "notes".

36.1 Using **Audacity**'s spectrum and autocorrelation capabilities, fill the following table:

	Note 1	Note 2	Note 3	Note 4	Note 5
Two major peaks in the spectrum (Hz)					
First peak in autocorrelation (seconds)					
First peak in autocorrelation (Hz)					

36.2 Comment on your results in question 36.1, relating them to your own perception of the pitch. Use a tone generator of **Max Partials** for comparison notes to check your pitch sensations. If you have perfect pitch, let us know in your comments.

Noisy scale

Here is the structure of **Max Noisy Scale** when you activate only one delay line:

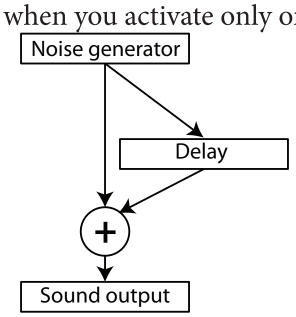


Figure 77: Max Noisy Scale structure, with only one active delay line

The delay d is specified by the keyboard in the following way: first, the keyboard gives the frequency f of the corresponding pitch. Then, this frequency is converted to the delay we need to hear this frequency as a repetition pitch. You can see the frequency displayed in **Max Noisy Scale** when you click on the keyboard.

36.3 Express the value of the delay d in function of the fundamental frequency f.

36.4 Record a sound corresponding to an A4 pitch (use the **Record 5s.** button), using just one delay line. Analyze it with **Audacity**. Attach the spectrum and autocorrelation, and comment, comparing to what you got in questions 36.1 and 36.2.

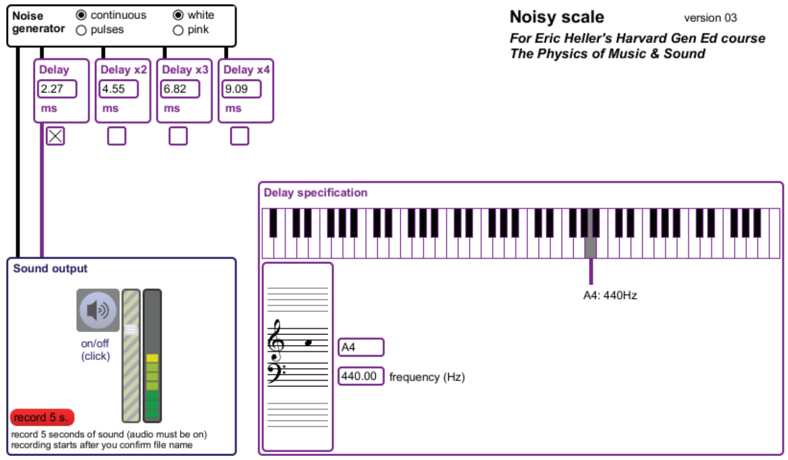


Figure 78: Max Noisy Scale screenshot

36.5 Explore what you perceive when you play with **Max Noisy Scale**:
- ✓ using one or several delay lines
- ✓ using the different delay lines in combination (for instance, delay × 2 and delay × 3)
- ✓ playing very high or very low notes on the keyboard

Write a few lines on *one* aspect of your exploration.

37 – Problem: Two ears, one simulation

Read in Why You Hear What You Hear

- Chapter 21: Mechanisms of Hearing

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)
- **Harvard Falstad Ripple applet** (on http://www.whyyouhearwhatyouhear.com/subpages/falstad.html)

Sound arriving from a source on your left naturally sounds like it is coming from that direction. However, it turns out that the sound is almost as loud in your right ear, due to the way sound bends around objects.
In this problem you will test this and the directional effect in a couple of ways.

37.1 In the **Harvard Falstad Ripple applet**, draw a mock head receiving sound from one side.
✓ pay close attention to all the settings you can read on Figure 79.
✓ make sure that **Fixed Edges** is unchecked.
✓ sound is arriving from a **Plane source** on the left of the head.

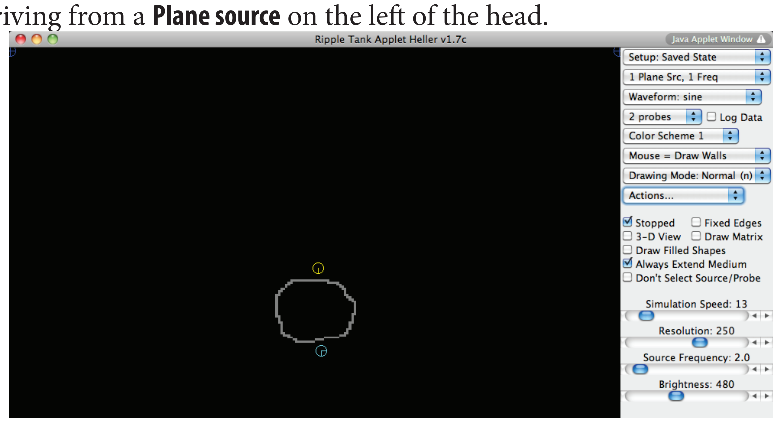


Figure 79: Mock head in the **Harvard Falstad Ripple applet**

With two probes, show to your satisfaction that:
✓ with a large enough wavelength, the sound is as loud on the "far" ear as on the "near" ear.
✓ the phase of the arriving sound is different than in the "near" ear. Attach a screenshot of your experiment in progress.

37.2 Now comes a real life test, using **Audacity**. Record a mono "click" or clap, or use the automatic click generator in **Audacity**. By switching between mono and stereo in the control panels to the left of the soundtracks, you can copy and paste as well as arrange for any time delay you wish for the left track as compared to the right. Estimate the time delay that would apply for the sound to get to the "far" ear as compared to the "near" ear if the sound came from one side. Make a stereo sound file with the same click or clap, but have one track time-delayed by this amount. You can of course experiment with the exact delay to see what gives the best effect. You need stereo headsets or earbuds. The effect is that the sound, though here by construction just as "strong" in each ear, sounds like it is coming almost entirely from one side.
Write what you did and what results you got, comment.

38 – Short questions

Viola fifths

The strings of a viola are tuned in fifths.

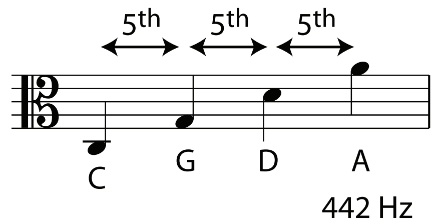


Figure 80: Viola tuning, with A at 442 Hz

The viola player tunes the viola by ear, until there is no beating between two strings. Thus, the instrument is tuned in perfect fifths, with frequency ratio 3/2.

38.1 A musical interval of a fifth in equal temperament corresponds to 700 cents. What is the frequency ratio between a note and the fifth below when using equal temperament? Hint: in *Why You Hear What You Hear*, section 26.7, you can read why a tempered half-tone, i.e. 100 cents, corresponds to a frequency ratio of $2^{1/12}$.

38.2 Fill the following table, giving the fundamental frequency of the three other viola strings if the A string is tuned to 442 Hz.
On the first row, consider the traditional way of tuning the instrument, in perfect fifths.
Fill the second row with the values that would result from an equal temperament (a fifth would be precisely 700 cents).

Note	**C**	**G**	**D**	**A**
Fundamental frequency (tuning with perfect fifths)				442 Hz
Fundamental frequency (tuning with tempered fifths)				442 Hz

38.3 For each string, give the difference in cents, between the two ways of tuning the instrument.
Hint: check the cent definition in *Why You Hear What You Hear*, section 23.18. To get the best results, do not use approximations in your calculation's intermediary steps. Round only the final numbers to integers.

Note	**C**	**G**	**D**	**A**
Difference between the two ways of tuning the instrument, in cents				0 cent

39 – Quiz questions

39.1 Each hair cell in the cochlea is responsible for receiving one frequency of sound.
True ☐ False ☐

39.2 Rock concerts can kill hair cells, and they don't regenerate.
True ☐ False ☐

39.3 The coiled cochlea receives low frequencies in the larger part near the tympanic membrane and higher frequencies near the small part at the end of the coil.
True ☐ False ☐

39.4 Normal hearing in a young person extends roughly from 20 to 20,000 Hz.
True ☐ False ☐

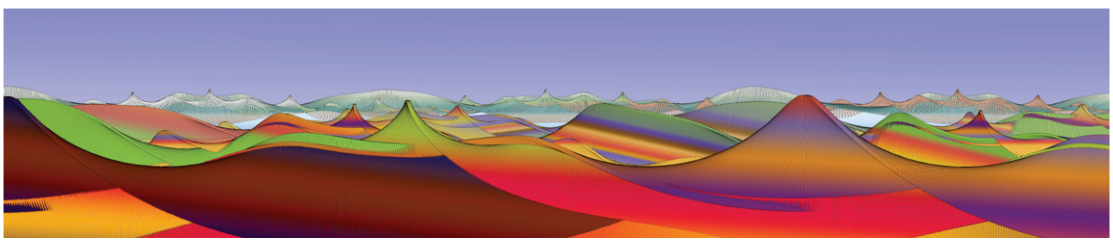
39.5 In the sound of a violin string with a fundamental of 250 Hz, the partials above the 10^{th} partial are too high to be heard, even for someone with good ears.
True ☐ False ☐

39.6 A repetition pitch can be heard even if a source doesn't repeat itself, for instance if the source is a simple clap of hands.
True ☐ False ☐

39.7 The perceived pitch of a sound is always one of the partials found from Fourier decomposition of the sound.
True ☐ False ☐

39.8 Two loudspeakers are each playing nearly white noise of a monaural recording of sandpaper being used on wood. One speaker is 2 meters further from you than the other. There is a repetition pitch heard; the pitch changes when the distance becomes 3 meters.
True ☐ False ☐

Soundspaces



40 – Problem: Soundspace investigations

Read in Why You Hear What You Hear

- Chapter 27: Modern Architectural Acoustics

Tools

- **Audacity** (details on http://audacity.sourceforge.net/)

In this problem set, you are going to investigate sound characteristics of an enclosed public space, such as a lecture hall, concert hall, dining hall, etc. You are going to make measurable and useful modifications to the sound space and compare the results.

40.1 Choose a space.

✓ Make a picture that you will include in your homework.

✓ Record a sharp pulse with **Audacity** (for instance, pierce a balloon).

✓ Find the RT_{60} for your chosen space. Remember that $\Delta dB/\Delta t = 60 \text{ dB}/RT_{60}$
Select in **Audacity** the **Waveform (dB)** view, in the **Audio Track** menu. Zoom on a linear portion of the decaying sound. Read ΔdB on the vertical scale, and Δt on the horizontal scale.

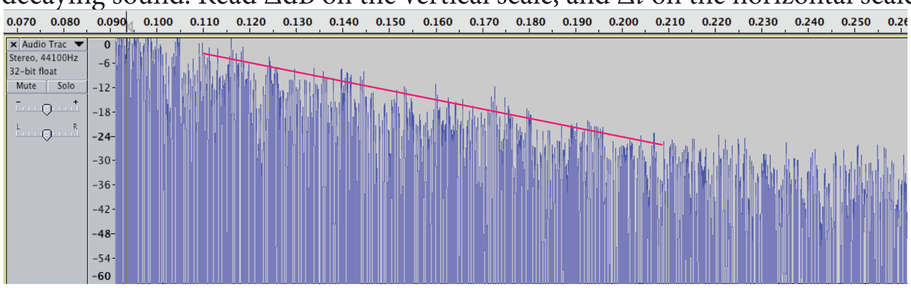


Figure 81: Linear decay shown in Audacity

40.2 Using formulas 27.15 and 27.16 in *Why You Hear What You Hear*, make a Sabine analysis of the space (estimate volume, surfaces, and materials) and calculate the RT_{60}. Show your work. Use the absorption coefficients given, for instance for 1000 Hz, on http://www.sengpielaudio.com/calculator-RT60Coeff.htm. α for an open window is 1.
Check out the RT_{60} calculator as well: http://www.sengpielaudio.com/calculator-RT60.htm.
Compare your result to your measurement from question 40.1.

40.3 Now, make measurable and useful modifications to the sound space and compare the results. Ideas for investigation:

✓ modify the sound space significantly, enough to get a different RT, and discuss the theory and findings of such modification (e.g. large windows opened, curtains drawn or opened, the space compared with and without people in it, etc.)

✓ make measurements in different parts of the room and check the theory that the RT does not depend on where the measurement is done;

✓ use sound impulses with contrasting spectral characters, i.e. one mostly low frequencies, one mostly high frequencies (using different sound pulse sources like high pitched metal objects versus low pitched ones, or two types of hand clap); see if low frequencies have a different RT than high ones.

40.4 Discuss whether the soundspace would be good for speech or music, or neither, and why.