Physics of Music Laboratory Manual Fall 2010

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Past versions by Prof. Dave Douglas.
With help from teaching assistants and students of phy103.

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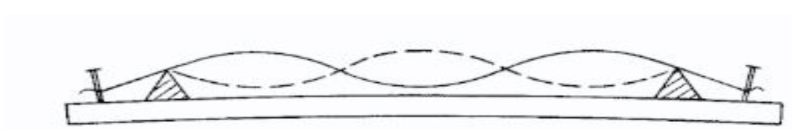

Guide to Writing a Lab Report

In General

- The goal of a lab report is to describe your experiments and results. Your report should be modeled after reports of research results or scientific journal articles. These are different than text books or papers written for classes.
- The most important parts of a scientific journal article are those that are read or looked at the most. The title is important. Other scientists will be searching for subjects related to their interests. The title should be descriptive enough that people in the field interested in the topic will be able to find and read the paper.
- The author list is important. The authors can be reporting on years of effort, and creative and innovative work. Papers are advertisements for research results. The authors are also responsible for the accuracy of the work and taking intellectual credit for the ideas and discoveries presented.
- After the title, the abstract is the most read (and so most important) part of any paper. In a few sentences the point/goals of the experiment and **major results** should be clearly stated. If a reader can't figure out what the paper is about and **what was found** in the paper, it is not going to be read. The abstract should be clearly enough written that somebody in a related field should be able to understand it. Consider making the abstract interesting enough that a potential reader will stop to look at the rest of the paper or will remember the major results. The results stated in the abstract should be derived, demonstrated and illustrated in the main text.
- After the abstract the most looked at part of any research paper are the figures. Readers often look at the figures before they read the text. Consequently the figure captions should be full of detail and tell the reader what the figures are about. The figures should illustrate the major results of the paper. The captions should tell the reader enough that he/she can understand what is shown in the Figures without reading the text. The captions should also point out to the reader what is important about the figure, and why the figure is important. Most of the results of the paper should be illustrated through the figures and data tables.
- You do not need a data notebook, however you should keep notes on your experimental set up and observations so you can describe them later.
- Please do not email the report to the teaching assistants or professor.

Sections of the Report

1. Heading

- Name of you, and your partner(s) and TA/TI
- Your title.
- Name of lab experiment, lab section and date.

2. Abstract

- Brief and clear summary of the experiment and **your results.** Summarize not only what you measured but also what you found from the measurements (e.g., trends, accuracy, consistency with estimates etc.).
- About 6 sentences. It can be longer and it can be shorter. A professor of mine was known for 1 sentence abstracts that were in the form "We searched for and failed to detect ……"

3. Theory/Introduction

- Brief description of the ideas underlying the experiment. Describe to the reader the motivation for carrying out the experiment without giving away any of the results. Often this is done in hindsight.
- Explain equations and basic physical principles of experiment if relevant. Remember to define symbols in equations.
- Explain historical context. Refer to other people's work and why yours might be different or new.
- Use your own words.

4. Data Analysis

- Describe your procedures and measurements. How did you make measurements? Refer to tables and lists of data measurements that are included in your lab.
- A note on figures: The major results of your paper can be illustrated with figures. It's much easier to look at figures than read text, or worse, drift through rows of numbers in a table. The figures can explain things better than text. But that means you can work hard at choosing and formatting your figures so they are successful at illustrating the major results of your paper. If you discover a trend then the figures could show the trend. If you measure differences between one experiment and another a figure could contrast these differences or 2 figures side by side could show different curves.
- I have mentioned that Tables are not very exciting to read but they are a way of recording measurements and calculations based on them in a clear fashion. Data points plotted in figures can be listed in tables. Tables are an effective way of comparing lists of numbers.
- Remember to define axes of plots and define variables used in equations.
- Remember to describe the data in your tables and your figures. Where do the numbers in each column or row of the table come from? How were the measurements made? What points are plotted? What are the figures showing? Look for and discuss trends. Contrast and compare different lines in plots, figures with other figures and different columns of data.
- Results should be demonstrated or illustrated with data or figures!
- Include captions on your tables and figures. The reader should be able to figure out what is listed in the Tables and shown in the figures without reading the main text.
- **Every** figure and table should be discussed and referred to in the text. Every measurement should be both discussed and described. How did you calculate it or measure it? How is it used for analysis and discussion? Results should be demonstrated and supported by your display of data. Refer to your figures and tables in the main text to demonstrate and support your conclusions.
- Let the reader know why and what you have plotted or drawn.
- Discuss possible sources of error and whether you believe that your results might have been affected by these errors.

5. Conclusions

- Summarize in a paragraph or two what you have learned from your measurements and experiments. The data analysis section should present figures and tables that show these results. However the main points can be restated here.
- Compare what you found with what was expected. Comment on differences. Sometimes your experiment screws up and you can't measure what you wanted to measure. If this happens discuss why you thought the experiment didn't work.
- The questions posed in the lab manual can be used to guide discussion of your experiments and results.
- Discuss ways that the lab or experiments could be improved. Discuss possible future work that is motivated by the results presented in this paper.

6. **Data**
   - Attach large data sheets (if any) and materials to back of the report. Summary data can be put in tables and discussed in the main text of the report.
   - Attach the notes that you took during the lab. You do not need to make changes in them. Changes or modification go into the presented data analysis section.
7. **Rules**
   - You can collaborate in the lab with taking measurements and sharing observations. However your lab report should be written by you. Plagiarizing a lab report is likely to result in academic action. If you are sharing measurements please simply state that you are doing so. Please discuss and describe your measurements and results separately and individually in your own words.

Physics of Music
Lab 1 – Measurements of Frequency

EQUIPMENT and PREPARATION

Part A:

- Oscilloscopes (get 5 from teaching lab)
- BK Precision function generators (get 5 from Thang or teaching lab)
- Counter/timers (PASCO, 4 working + 1 other)
- Connectors and cables (BNC to BNC's, banana plug pairs) connecting function/signal generators to oscilloscopes and speaker.
- BNC to 2 leads
- Adaptors: mike to BNC so that output of preamps can be looked at on oscilloscope.
- BNC to banana adapters
- BNC T- adapters
- Oscilloscope probes
- Pasco open speakers
- Microphones, preamps, mic stands

Warning: do not place speakers near oscilloscope screens as they can damage the CRT (cathode ray tube) screens.

Note: The professor will attempt to have the equipment out and available for the labs. However the TAs should check that the equipment is ready to use, that every lab setup has all the necessary equipment. The TAs should also be very familiar with the lab and know how to troubleshoot the equipment.

INTRODUCTION

In this lab, we will be measuring the frequency of a signal using three different methods. The first method involves the use of the function generator. A function generator is an instrument that can produce sine, square, and triangular waves at a given frequency. The second method makes use of the oscilloscope. An oscilloscope is an instrument principally used to display signals as a function of time. The final method for measuring the frequency uses the counter/timer (optional). A counter/timer is an instrument that can give a very accurate measurement of the frequency of a signal.

The frequency, f, and period, T, of a wave are related in the following way:

$$f = \frac{1}{T} \qquad \text{(Equation 1)}$$

For frequency in Hz (cycles per second), the period is given in seconds. This equation makes sense because the frequency is the number of cycles that fits into 1 second (fT=1).

The electrical signals created by a function generator can become "sound waves" when passed through a speaker. In this lab we will verify this by looking at the signal output from the microphone on the oscilloscope. To send signals between the various instruments we will use wires/cables with the following connectors:

	BNC connector (male)
	Banana Plug (male)
	¼’’ phone connector (male)

PURPOSE

The purpose of this lab is to gain a working knowledge of a function generator, an oscilloscope, and a counter/timer. We will look at signals picked up by an oscilloscope probe and what that from a microphone. We will experiment with different ways of measuring frequency.

PROCEDURE

Part I – The Function Generator

1. Turn on the function generator.
2. Press in the button on the front panel with the sine wave picture.
3. Where it is labeled “Range,” press the “500” button.
4. Use the “Course” and “Fine” tuning dials (where it is labeled “Frequency” on the front panel) until the display reads 300 Hz.
5. In your notebook, write “300 Hz” for the value you obtained for your frequency using the function generator on your first trial.
6. Turn the Output level (amplitude) knob on the function generator so the output is at least half of the maximum.
7. Connect the output of the function generator to the speaker. Adjust the amplitude. Vary the frequency from the function generator. Make sure you can hear a tone from the speaker. Get a feeling for what frequencies correspond to what sounds.
8. Explore the range of your hearing in frequency. What is the lowest frequency you can hear and the highest you can hear?
9. Set the frequency back to 300Hz and disconnect the speaker.

Please keep the speaker away from the oscilloscope to prevent damage to the CRT screens!

Part II – The Oscilloscope (you will be using Tenma scopes)

1. Turn on the oscilloscope by pushing in the power. It will take a few seconds for a trace to appear on the screen. If one does not appear in a few seconds, try increasing the trace intensity (see #3).
2. Connect the output of the function generator to one of the inputs of the oscilloscope.
3. Adjust the intensity. This is done by turning the intensity button on the Tenma scopes in the Display section.
4. There is a FOCUS. You can adjust this knob to focus the line in the monitor.
5. Note that if the speaker is near the oscilloscope, its magnet can distort the display. This is kind of fun to show but if you do too much of this it can damage the CRT and permanently warp it (we had this happen last year!)
6. There is a MODE switch in the “Vertical” section. If your signal from the function generator is going into Channel 1 then move this switch to “Ch 1.” This allows you to look at channel 1. There are two channels available so that it is possible to compare two traces at once. To show both traces at once (that is the

signals going into channel 1 and 2), select the DUAL mode. Ordinarily you look channel 1 or 2 in the y axis vs time in the x axis.. The X-Y mode allows you to look at an x versus y display. The horizontal (x) signal is connected through the CH1 X input connector and the vertical (y) signal is connected through the Ch2 Y input connector. The ADD mode allows two traces to be added together. If your signal is going into Channel 2 then you would make sure that the MODE switch is either on "Ch2" or on "Dual".

7. There is a SOURCE switch in the "Trigger" section. Make sure this switch is on "Ch1." (if your signal from the function generator is in Channel 1) or on "Ch2" (if your signal is going in Channel 2).
8. Using the CH1 Y shift control (the vertical arrows) in the "Vertical" section to make sure that the trace is in the middle of the screen vertically.
9. Using the X shift control (the horizontal arrows) in the "Horizontal" section, make sure that the trace is in the middle of the screen horizontally.
10. Turn the VARIABLE knob, in the "Horizontal" section, clockwise to the CAL position. If this is not done you might measure incorrect times.
11. In the "Vertical" section, there is a GND button. Make sure that this button is sticking out (i.e. not pushed in). This will keep the signal from being grounded or shorted out. On other Tenma scopes you must select either GND, AC or DC. Make sure GND is NOT selected. If GND is selected you will not be able to see your signal (but you can center the signal and make sure you know where 0Volts is).
12. If the waveform on your screen is not stationary, adjust the LEVEL knob in the "Trigger" section. When the waveform is not stationary it is said to be "free-running". If your signal is not stationary then the TRIGGER may not be adjusted properly or you make have selected the wrong channel for the trigger.
13. The Ch1 dial contains an inner and outer dial. **Make sure the inner dial is pushed in and set all the way clockwise to CAL.** This way the voltage read from the screen will be exact.
14. The TIME/CM or TIME/DIV knob in the "Horizontal" section specifies how long it takes for the trace to sweep through a centimeter on the oscilloscope screen. Adjust the TIME/CM or TIME/DIV knob so you can see one full cycle on the monitor.
15. The VOLTS/CM or VOLTS/DIV knob in the "Vertical" section determines how tall the signal will be on the screen. Adjust the VOLTS/DIV knob for channel 1 to obtain a waveform that fills up almost all of the screen vertically.
16. Re-adjust the INTENSITY and FOCUS knobs to get a clear trace.
17. At this point you should have a stationary waveform positioned nicely on your screen. If this is not the case, get help from the TA or TI.
18. Count the number of squares in one period of the wave (peak to peak). Record this number in your notes. Remember that each square is 1 cm by 1 cm.
19. Multiply this number by the value that the TIME/CM knob is set to. Make sure to convert the value that the TIME/CM knob is set at into seconds per centimeter. For example, 0.2 ms/cm = .0002 s/cm. A conversion table is provided at the end of the lab manual in appendix A. The number that you have just calculated is the period of the signal in seconds. Record this value in your notes.
20. Determine the frequency of the signal using formula (1.1).
21. In your lab notebook, write the value you obtained where it asks for the value of your frequency using the oscilloscope on your first trial.
22. Error estimation: Suppose you mis-measure by 1/5 box on the screen. Redo your calculation. Compare this frequency to the one that you calculated in #19. The difference can be an estimate of the error or your uncertainty of your measurement. Record your estimated error (uncertainty) in your notes.

Part III – The Counter/Timer

1. Turn on the counter (switch on power supply module on left). A counter toggles every time the signal crosses a certain level. It can be used to count the number of crossings per second.
2. Connect the output of the function generator to the counter *input* area. The black lead from the generator should go to the GND (ground, black) input. The red lead from the generator should go to the white or 0.5V p/p MIN lead. This stands for 0.5Volts peak to peak. To get the counter to trigger the input must go above 0.5Volts peak to peak. **Note:** there is one counter that is simpler than all the rest and connects with BNC and no leads. If you are at this lab set up the counter should simply show you the frequency in Hz (though I think there is also a reset button).
3. In the counter module flip the black mode switch (up and down switch) to measure frequency (kHz) (bottom position). Note 1kHz is 1000 Hz.
4. In your notebook, record the value you obtain for the frequency measured by the counter.

5. You can make a more accurate measurement by setting the counter module to counter (middle position for the black mode switch). The set the counter to 1s (this is the horizontal switch in the counter section). On the upper right press the START/STOP button. The counter should measure the number of pulses in 1s. To make another measurement, you can press RESET (button on top left) before you start the counter again. You can do this a number of times and average the value.

Part IV – Additional Measurements

1. Carry out at least two additional trials at different frequencies. For example at 1000 and 5000 Hz or other frequencies that you might choose. For these additional trials use the three methods (function generator, counter and oscilloscope) to record or estimate frequency. In your lab report you will discuss differences between the frequency measurements and your feeling about the uncertainties in the frequency measurements.

Part V – Looking at the output of the microphone on the oscilloscope.

1. Connect the output of the pre-amp into an input of the oscilloscope. You will need an adapter (male BNC to female 1/4" audio) that should be part of your lab equipment on the desk.
2. Make sure that your microphone is connected to the preamp and that the preamp is plugged in. The green power indicator should light up. The +48V button should be pushed in. This powers the microphone that employs a capacitor. The capacitor creates an electric field between two plates. The plates respond to small pressure variations (such as those made by sound waves) creating a signal that is amplified by the preamp and passed on to the computer. Powered microphones are usually superior to un-powered ones.
3. Play a sound from the computer speaker or by connecting the function generator to the open speaker. If you use the function generator you can send that signal into channel 2 of the oscilloscope and display both the signal from the microphone and that from the function generator together.
4. Compare the output of the preamp (fed by the microphone) to the signal from the function generator. You can adjust the strength of the microphone output with the dials on the preamp. This signal should be stronger than that you will see by using the speaker as a microphone in the next lab.
5. Try singing or speaking into the microphone while looking at the signal on the oscilloscope. Can you sing notes that cause the microphone signal to be a periodic signal even with the speaker on? (Pure intervals such as octaves or fifths or thirds?)

Part VI – Your body as an Antenna.

1. Remove the BNC cable from the Ch1 input. Plug in the oscilloscope probe into Ch 1.
2. Touch the end of the oscilloscope with your finger.
3. Adjust the Ch 1 VOLTS knob so that you can see a signal. While your body is a pretty good antenna, you don't actually pick up a large voltage. You can use the oscilloscope to look at very small voltages and you will need to do so to see a signal.
4. Adjust the TIME/CM knob to very small time intervals. You should see a lot of high frequency noise.
5. Adjust the TIME/CM knob to larger time intervals. Look for a sine wave pattern in the noise. Your body is picking up a pretty strong low frequency signal. What is this frequency coming from? Measure its frequency to see if the frequency itself gives you a clue. Hint: *

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* Electric power in the United States is in the form of alternating current (AC). The voltage on an electric outlet looks like a sine wave (is alternating) with a frequency of 60Hz. The amplitude of the voltage oscillation is approximately 110Volts. The standard for most of North, Central, and South America and the Caribbean is 50Hz and 120V AC. The standard for most of the rest of the world is 50Hz and 220V AC. Your body because it is full of electrolytes can act as an antenna. The ground level seen by the oscilloscope may differ from the voltage of your body. If you touch one hand to the tip of the probe and the other hand to the ground of the probe you will still see the 60 Hz signal but it will be weaker. 60 Hz noise is often a problem in electronic equipment. You may have heard an undesirable "60 Hz Hum" on a speaker system. There are a variety of ways to get rid of 60 Hz noise, including the use of shielded or twisted cables and well grounded instrument housings.

DATA ANALYSIS (Lab Report)

1. Create a data table showing your measured frequencies and estimated errors. Explain where each row of numbers comes from and how you measured or calculated each. Explain how you measured frequencies using the oscilloscope and using the counter.
2. Remember to define symbols (and possibly units of quantities) if you give equations.
3. Refer to the table in your text. For example: "Column 2 of Table 1 lists frequency measurements based on the oscilloscope and calculated with the equation $f=1/T$ and using the procedure described above. Column 3 of Table 1 lists frequencies measured with the counter."
4. Discuss your table in your text. For example: "Comparison of Column 1 and 3 shows that the counter and frequency generator measured the same frequencies to an accuracy of 1Hz. The difference between Columns 1,2 and 1,3 in Table 1 suggests that the frequency measurements errors from the oscilloscope are the largest."
5. Look for trends. Are the oscilloscope measurements systematically high or low compared to the settings of the function generator and counter/timer? Are the differences larger than you expect? Are the errors larger or smaller at high frequencies than low frequencies?
6. Estimate the sizes of your errors. The difference between two measured frequencies would be a Δf in Hz. You could also compute $\Delta f/f$ which would give you the error as a fraction of the signal.
7. In this lab your discussion of data could be used to argue which instruments are most accurate and estimate how accurate they are. These can be considered "results" and so can go into your abstract. For example. "We find that the frequency given on the function generator agreed with that measured by the counter, with differences between the two measurements that were within 2-4 Hz for signals in the range 100- 1kHz. We infer that these two instruments probably give accurate frequency measurements with errors less than 2-4Hz. We find that frequencies measured on the oscilloscope were consistent with the other measurements but tended to differ from the other measurements by a larger number or ~10Hz. We find that the counter and function generator are more reliable measures of frequency than the oscilloscope."

A note on error analysis: Sometimes scientists are under the impression that error analysis must be done in order to achieve a type of professionalism. However error analysis requires work and in order to accomplish more one must figure out where and when it is important to do this work. If experimental errors are not significantly affecting your measurements then it is silly to spend a lot of time analyzing them (your time is too valuable!). Errors are a problem if they are significantly affecting your measurements. For example, in your report you might want to argue that experimental errors are not affecting or invalidating your major results. As part of this argument you could demonstrate understanding and estimates of the size of the errors. These estimates might then be used to show that the errors wouldn't significantly change your measurements or invalidate the major results of your experiment. Errors are a part of any experiment. Sometimes your analysis suggests a trend or gives you a measurement that might not be certain. In this case you should discuss the uncertainties in your results and how they might be affecting your measurement or trend. Your conclusion would give a summary of your trend or measurement and then would explain the problems with the measurements. You could also suggest future ways (if any) of improving your experiments to validate (with better data) the results of your study.

QUESTIONS and DISCUSSION

1. Did you measure the same frequencies with different measurement techniques? Why or why not? Are the differences larger than you expected based on your error estimates?
2. If you measured the frequency with both oscilloscope and counter, which method did you find most accurate? How could you tell it was the most accurate? Can the counter measure fractional frequencies? Is it more accurate at low or high frequencies?
3. About how accurately can you measure the frequency with the oscilloscope and counter? (to what % or what difference in Hz?). Is the accuracy of your measurements higher or lower at low frequencies than high frequencies?
4. Discuss trends or lack of them. You may find it easier to illustrate trends by making a figure plotting the measurements from each instrument or by referring to lists of numbers in a data table.

5. Describe your experiences with viewing the signal from the microphone and your body. Is the difference between the frequency you measure and 60Hz from AC power within your estimated errors?

Physics of Music

Lab 2 – Periodic Signals – Triangle and Square Waves

EQUIPMENT and PREPARATION

- Computers
- Check that email works on the computers
- Know how to screen snap so that images can be sent home by students. MWSnap installed.
- Headphones
- Microphones + Preamps + cables and associated connectors, adaptors and cables. It should be possible to record sound into the computer using Adobe/Audition. Do a record sound check on all computers!
- Microphone stands
- Musical instruments and/or sound-making devices
- Pasco open speakers
- Oscilloscopes, and way to connect speaker to oscilloscope (BNC to banana adapter + banana plug cables)

Warning: do not place speakers near oscilloscope screens as they can damage the CRT screen.

INTRODUCTION

In this lab we explore the gap between electronics and music by generating and observing signals while listening to them. We will explore the relation between the shape (waveform) of a sound, its spectrum and the timbre or character of the sound. We will have the chance to create some sounds of our own and record them onto the computer. The computer software lets us look at sound waves as a signal as a function of time (waveform view) and as in terms of their frequency distribution (frequency view or frequency analysis). To do compute the spectrum we will use the algorithm known as the Fast Fourier Transformation (FFT). The FFT is an efficient computational algorithm that can, through Fourier Analysis, break down a signal into a number of basic sine waves of specific frequencies.

In this lab we will become familiar with the Adobe-Audition audio software. Adobe/Audition contains its own function generator, along with the ability to look at a waveform and its spectrum using the FFT and record sounds.

Triangle, square and sine waves are all periodic signals. In this lab we will find that periodic signals have frequency spectra with particular properties (integer harmonics). We can record our voice and music instruments and will find that the spectra of these sounds have similar properties to the periodic signals generated by a function generator.

PROCEDURE

Part I – Setup

1. Make sure the computers are on and booted.
2. If there is a lot of noise in the room, you can use headphones by unplugging the speakers and connecting the headphones into the speaker jack. If you are using speakers connected to the computer, make sure they are turned on (turn the knob on one of the small speakers).
3. Make sure that your microphone is connected to the preamp and that the preamp is plugged in. The green power indicator should light up. The +48V button should be pushed in. This is to power the microphone which employs a capacitor. The capacitor creates an electric field between two plates. The plates respond to small pressure variations (such as those made by sound waves) creating a signal that is amplified by the preamp and passed on to the computer. Powered microphones are usually superior to un-powered ones.

4. Make sure that the preamp is connected to the computer through the preamp's output jack and the computer's input microphone jack.
5. Load MWSnap by double clicking the camera icon. This program will allow you to take pictures of the screen that you can then email to yourself at home. This will allow you to insert nice figures into your lab report later on.
5. Load Adobe-Audition on the computer by double clicking the speaker icon.

Part II – Looking and Listening (triangle, square and sine waves)

In this part of the lab you will generate tones using Adobe-Audition, you will view their waveforms and listen to their sound.

1. Click the generate menu and choose TONES. The first time you do this a window titled "New Waveform" will come up. Set the sample rate at least at 48000Hz. Set the Channel to "MONO". Set the Resolution to at least 16 bit. Then click OK.
2. In the box labeled "Presets" choose a A440 default.
3. On the bottom right, set the duration to 1.0 seconds.
4. Just above that, in the box labeled "General", chose a "flavor" of "Sine". The flavor is the type of waveform created. You can listen to this by clicking "Preview". Click OK Now you have a wave.
5. Click on the wave you have created and hit the key on your keyboard labeled "END". This will send the cursor to the end of your sound file. You will now create two mores sounds with the same pitch.
6. Repeat step #4 with a flavor of "Triangle/Sawtooth". Repeat step #5.
7. Repeat step #4 with a flavor of "Square". Hit the "Home" key on your keyboard to place your cursor at the beginning of the file.
8. You now have 3 one second waves. Click on the green right arrow key (bottom left) to play the sounds. You can adjust the volume with the volume icon on the computer on the lower right or with the knob on one of the speakers.
9. Describe in your notebook your perception of these sounds.
10. Looking at the waveforms. Left click on the waveform somewhere. Use the magnifying glass button (top row, on bottom of screen, circle with a + sign in it) to expand the x-axis. Take a closer look at the waveforms. You can play the sounds again while watching the waveforms move by hitting "Home" and then play. You can click and drag the green bar above the waveform to view different parts of the wave. You can also expand the horizontal axis by right clicking and dragging. To zoom out: right click on the axis and choose "Zoom full." The button with a circle and a – sign will also allow you to zoom out.
11. Draw in your notebook, the shape of the waveforms (including labeling the x and y-axes) or send yourself a snapped view.
12. If you would like to save images for your lab report: click on the MWSnap program bar at the bottom of the screen. Click on "Any rect. Area". Click on "Snap any area." Left click and drag to chose the window. Left click again to sent the picture out to MWSnap. Save it to the desktop. While saving you can chose the format for the picture (bmp or jpg are pretty common).
13. Optional: Experiment with different "Flavor Characteristics" or different frequency components using the "Generate Tone" function in Audition.

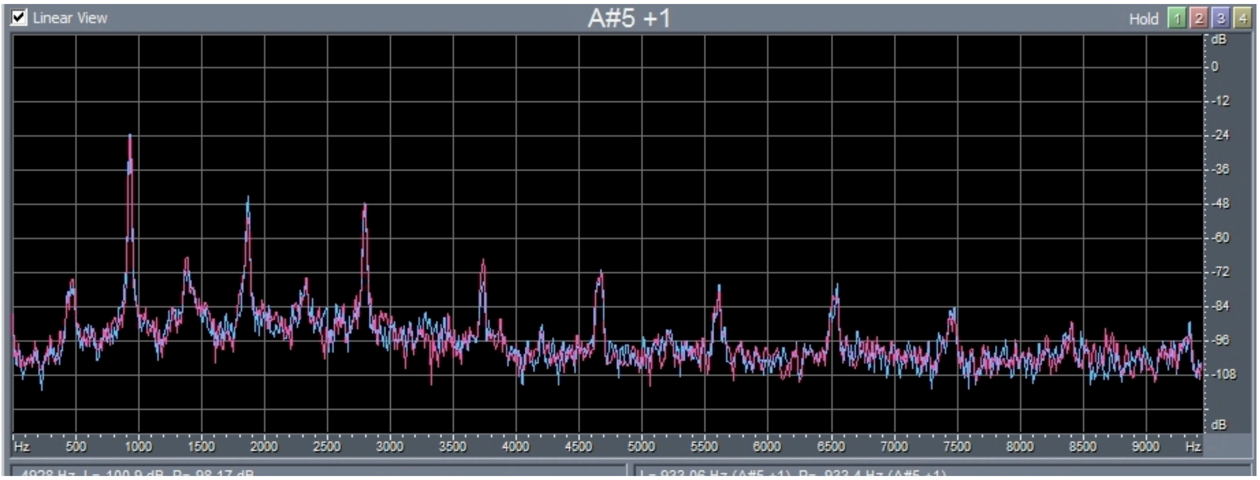

Figure 1. Above is a spectrum of a flute playing note A#5. Each peak is an overtone (partial or harmonic). The vertical scale is in dB, or 20log10 (amplitude) or 10log10(power). You can measure the strengths of the overtones from the strengths of each peak. The fundamental is at about 932 Hz, the first harmonic is at about twice this at 1900Hz and is weaker than the fundamental. Notice that 1900Hz is about twice as big as 930Hz. The note played is A#5 (the A# in the octave above middle C) but actually the lower octave A# is also weakly excited and you see some A#4 in the spectrum corresponding to peaks at 466 and 1400Hz. You can adjust the x-axis of the spectral window to make a better measurement of the frequencies of the peaks. The y-axis here is in dB which means decibels. This is a logarithmic scale (note the negative numbers!). A small change in dB represents a pretty big change in power. A change of 10 dB is a factor of 10 in power. The first few overtones are strong and narrow and so will allow both an accurate frequency measurement and strength measurement. The higher overtones are weaker and so harder to measure. The uncertainties in their measurements will be larger than the lower overtones.

Part III – Frequency Analysis

You can now learn to use a tool that enables you to look at the frequencies in the sound wave. This tool will give you a graphical representation of the fundamental tone and its harmonic frequencies.

1. Click on the Analyze menu at the top of the Audition screen and choose "Frequency Analysis". Click on "Advanced" (if not already set to this) to bring up more options. Set the FFT Size to an intermediate resolution of 4096.
2. Left click on your waveform window in the middle of a region showing one of your generated waves. The frequency analysis window should now be showing the spectrum of that region.
3. Using the right mouse button, in the frequency analysis window, click on the horizontal rule at zero Hz and drag to 1000Hz. This allows you to see the frequency distribution between 0 and 1000Hz. You ear is sensitive past this to 20,000 Hz but most musical tones are well below this. For example concert A is at 440Hz. Expanding the x axis to between 0 and 1000Hz will allow you to measure the amplitude and frequency of the fundamental mode.
4. If you right click and zoom out you will be able to see more overtones. If you generated a tone that is 440Hz, twice this would be 880Hz and three times this would be 1320Hz. Look in a region from 0 to 10000Hz. Are the overtone frequencies integer multiples of the fundamental? Odd integers? Even integers? Exact integers?
5. In the waveform window, click on different regions of the sound file. Look at the different frequency spectra for the sine wave, the triangle wave and the square wave. Draw the frequency spectra that you see, or save/snap images of them that can later be placed into your lab report.
6. Click on a peak in the frequency analysis window. The frequency of this spot should appear on the lower left in the frequency analysis window. Record the frequencies and strengths of the fundamental and overtones for each waveform. Are the overtones integer multiples of the fundamental and if so which integers? Write down the strengths of the overtones. The y-axis of this window is dB or decibels which is a log scale. Don't be concerned if you have negative numbers as a decibel is a log scale and the log of small number is negative.

7. Compare the strengths of the harmonics or overtones between the different sounds. Are there overtones in each sound? Are the overtones at the same frequencies for the triangle and square wave? Are they the same amplitudes (strengths)?
8. Experiment with the FFT. What happens to the spectrum as you change the FFT size or length? What happens to the spectrum as you change the window function? Is there a relation between precision and FFT size? If the FFT length is too long then different notes can run together.
9. For those who know how to compute a Fourier transform: Compute or look up the expected strengths of the overtones for a triangle and square waves. Are the strengths of the overtones you measure consistent with your calculations? Note: the spectrum window is usually displayed in dB (decibels, that would be 20 $\log_{10}$ the signal or 10 $\log_{10}$ the power).
10. For those who how to compute a Fourier transform: Why does the FFT of a sine wave NOT give a spike at a single frequency. Explore how the different windowing functions and length of the FFT affect the shape of the tail and the width of the spike for the spectrum of a sine wave.
11. Inter-harmonic spikes are probably artifacts related to a mismatch between the FFT window size and the number of periods of the one that fit inside it. If you are interested: Look at the artifacts in the spectrum for tones generated at frequencies other than 440Hz and compare them to the inter-harmonic spikes you see in the 440 Hz tones. Is it possible to adjust the frequency so that the window and sampling rate conspire to remove all these inter-harmonic spikes?

Notes: The FFT window is not a good match to the period of 440 Hz for standard sampling frequencies. We see power at frequencies between the main overtones. These weak spikes are probably artifacts. The y-axis for the frequency view is in decibels. This is a log scale so don't be concerned if the numbers are negative.

Part IV – Using the speaker as a microphone.

A speaker turns an electronic signal into physical motions creating sound. A microphone is something that turns physical motions in the air into an electronic signal. However sounds also move speaker parts creating weak electronic signals.

1. Connect the Pasco open speaker to the input of the oscilloscope. Create a pure sine tone on the computer. Place the Pasco speaker right next to the computer speaker. If you click on the infinity button the sound file will loop endlessly.
2. Turn the voltage knob on the oscilloscope to the right so that the oscilloscope can see the smallest possible voltages. Can you see a weak sine wave? Does the frequency you measure with the oscilloscope agree with that you chose for the pure tone you created on the computer? You may need the speakers right next to each other and the volume pretty high to see the weak signal on the scope as a speaker is not really a very good microphone. *

Part V – Recording and analyzing your own sounds

Health Warning: If you share wind instruments, please sanitize them first with mouthwash or by washing them in the bathroom sink before you blow into them.

For the remainder of this lab, you will create your own sounds, and observe them in the same manner as you have previously observed sounds using the generate tones function.

1. To check the record level, right click on the bar at the bottom of the screen and choose "Monitor record level." If you talk into the microphone the bar should now move. You want the pre-amp volume and gain levels set so the sound is not in the red (or clipped) and also not too soft.
2. To record you need to click on the red spot at the bottom of the screen (record button). Record some sounds such as yourself whistling or playing an instrument.

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* If you measure a 60 Hz signal then remind yourself of a common source of electronic noise that we directly observed in Lab #1. You may wish to reconsider whether you are measuring a sound wave rather than this common source of noise.

3. While recording ensure that the volume input and outputs on the preamp are at a good level so that the signal is not heavily clipped or extremely faint. You can adjust your distance to the microphone and the input and output volumes on the preamp. To minimize the effect of noise in the room, you would like to record with your noise source (e.g., mouth) close to the microphone. With a couple of recording trials adjust the preamp knobs so that this is possible.
4. After recording, inspect the waveforms of your sounds. Zoom in with the horizontal scale and inspect the waveforms over small time intervals. Sketch or save images of your waveforms. Remember to label your axes.
5. You can view the entire recording in spectral view by clicking “View” at the top of the screen and then choosing “Spectral View”. You can adjust the vertical scale (now the frequency scale) by zooming in vertically or with a right click and drag on the axis bar. You can also look at the frequency spectrum using the frequency analysis window. Describe or sketch or save images of the spectrum of your whistling.
6. **Warning:** I recommend that you do not record in Spectral view as the software will not be able to keep up and you will miss part of the recording (you would see this as vertical bars in the spectral view that should not be there).
7. Although you must record in spectral view, you can keep the spectral analysis window open while you record or while the record level monitor is active. If you set the FFT in the Spectral analysis window to a lower FFT size such as 8192, the spectrum is updated in real time. Right click on the level bar at the bottom and choose “Monitor record level” to have it update in real time! Zoom in so the horizontal range in the spectral analysis window is 0-4000Hz. Whistle and watch the spectrum change. Record yourself whistling different pitches and watch the fundamental tone shift in frequency. Neat!
8. You might at times have a pretty big sound file. To get rid of it, hit Ctrl-A followed by delete. Or you can open a new and empty sound file. Or you can save your sound file and e-mail it home and look at it later on your own computer using the free software called Audacity (that pretty much has all the same functions as Audition).
9. Record yourself speaking. Look at the waveforms. Are they periodic (repeat in a period like the triangle or square wave)? Look at the spectra of the different vowels. What is different about the spectra of different vowels? Is voice a nearly periodic signal?
10. Record a whistle or another musical instrument that you find in the lab. What is the difference in the spectra of different musical sounds (singing vs speaking, a whistle vs a clap)?

Finish

1. When finished, close all windows on the computer. Email yourself the images you have saved using the MWSnap tool so you can put them in your lab report later.

DATA ANALYSIS

1. Explain the differences between a sine wave, a square wave, and a triangle wave. Consider the waveform and frequency views. Describe the differences in the way they sounded.
2. Include a table that lists your measurements of the overtone and fundamental frequencies for these three wave types. Do the three sounds (sine, triangle and square wave) have the same overtones frequencies? Are the overtones integer multiples of the fundamental? If so which integers?
3. How well (with what types of uncertainties) could you measure the frequencies and amplitudes of the overtones? Did your uncertainties affect your conclusions in the previous question?
4. Include a table that lists the amplitudes of the overtones. Compare the amplitudes (strengths) of the overtones for the three types of sounds.
5. For each figure you include in your lab report, describe how you obtained it. Describe what you see in it. For example you could point out that you see overtones and comment on their sizes (amplitudes).
6. For each table in your report, describe where the numbers came from and how you made the measurements. Label table columns.
7. Discuss how the FFT size (or window length) and window shape affects your spectroscopic measurements.
8. Describe spectral differences between different types of real (rather than computer generated) sounds that you recorded.

QUESTIONS and DISCUSSION

1. Why don't sine, triangle, and square waves with the same frequencies and amplitudes sound the same?
2. How did the waveform and frequency spectrum of recorded sounds differ from those generated by the generate tones function? How do the waveform and frequency spectra of things you recorded (e.g., whistle, voice and musical instrument) differ?
3. How accurate are your measurements and how did the uncertainties of measurement affect your findings? For example if you find that a triangle and square wave have overtones that are at the same frequencies you might argue that they are the same within your estimated uncertainties of measurement. If you find that overtones are integer multiples of the fundamental frequency you can also show that this is true within your estimated errors. What contributes to the errors? Factors to keep in mind that might have affected your measurements: your sampling rate, the response of the microphone, the linearity and response of the preamp, the settings of the frequency analysis FFT, the window used to make the FFT,…. Some overtones could be harder to measure than others because they are weaker. The uncertainty in your measured overtone strengths will be different than your uncertainty in your measured frequencies.

Physics of Music
Lab 3 – Spectral Analysis of Sliding Whistles

EQUIPMENT

- Microphones + Preamps + cables and associated connectors, adaptors and cables. It should be possible to record sound into the computer using Adobe/Audition, Microphone stands
- Sliding whistles (metal ones are easily washed)
- Rulers
- Antiseptic mouthwash in spray bottle for sanitizing the whistles.
- Digital Tuners

Health Warning: We wash the whistles before *every* lab. However if you share the whistles, please sanitize them first with mouthwash or by washing them in the bathroom sink before you blow into them. We don't need to contribute to the flu season!

Tis/TAs: If the whistles have a ***clean*** sign on them they are ready for the lab. If not, please wash them before the lab. There is a gallon of alcohol to sterilize after washing them.

INTRODUCTION

In the previous lab, you learned how to use Adobe/Audition to break down both simple waves (like sine, triangle, and square waves) and complex waves (like those of your voice or a whistle). In this lab, you will utilize those skills to do a deeper analysis of two complex (and beautiful) instruments: the sliding whistle and the human voice.

The whistle appears to be a simple instrument: a tube, open at one end, which resonates at many frequencies. Yet the whistle is a complex instrument. For the moment, we will only explore the simplest quality of the sliding whistle: the relationship between the position of the stopper and the tone produced. You probably are already familiar with this relationship, as you have seen a trombone player extend the length of his resonance tube by moving his slide, in order to produce a lower tone. The relationship between length (L) and frequency (f) is given by the following formula:

$$L \propto \frac{1}{f} \qquad \text{(Equation 1)}$$

The symbol in the center denotes proportionality. By this we mean $L = A/f$ where A is a constant. This relation predicts that shorter tubes will resonate at higher frequencies and so play higher notes.

Figure 1. Above is an example of a metal sliding whistle. When pushed all the way in, I find that the distance between the end of the inner stopper inside the tube and the edge of the hole in the mouth piece is

3.8 cm. As the stopper is pulled outwards this distance increases. When you blow into the whistle you get a high note if the stopper is nearly all the way in. As you pull out the stopper the note becomes lower and lower.

PURPOSE

The purpose of this lab is to use software to analyze the waveforms and frequency spectra produced from a sliding whistle and your own voice. You will adjust the stopper of a sliding whistle and establish a relationship between the length of the resonating air pipe and the frequency or note produced. In addition, you will look at the spectrum of the human voice and compare its overtones to that of the sliding whistle. The length, L, that we want to measure is the length of the air pipe that is in vibration. The length of this column of air is the distance between the edge of the hole in the mouthpiece and the flat face of the stopper that is inside the pipe. See Figure 2.

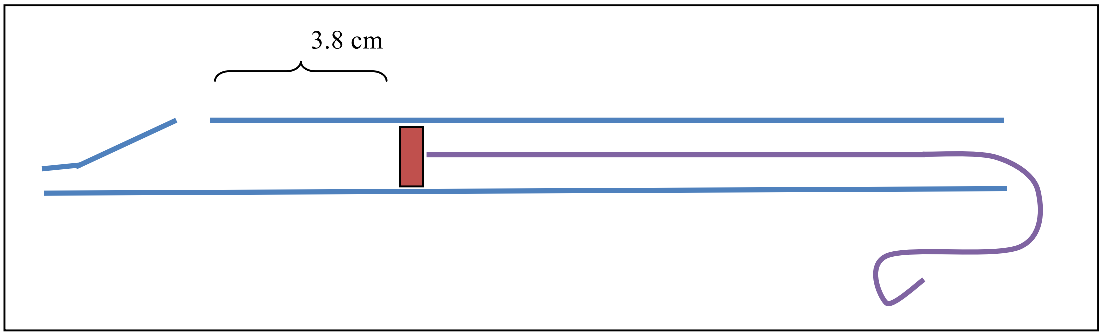


Figure 2. When the stopper is pushed all the way in the length of the air column is about 3.8cm. You can measure this by pushing a wire into the hole of the whistle until it touches the plunger. Mark or hold the part of the wire just sticking out. Pull out the wire and measure the length between the end and your mark.

To measure the length of the air column when the stopper is pulled out: Measure the length of the flute when the stopper is in. Measure the length pulled out. Subtract the two lengths and add 3.8cm. This will give you the length of the air column between the edge of the hole in the mouthpiece and the stopper.

Health Warning: If you share the whistles, please sanitize them first with mouthwash or by washing them in the bathroom sink before you blow into them. We don't need to contribute to the flu season!

PROCEDURE

Part I – Setup

1. If there is a lot of noise in the room, you can use headphones by unplugging the speakers and connecting the headphones into the speaker jack. If you are using speakers, make sure they are turned on (turn the knob on one of the speakers).
2. Make sure that your microphone is connected to the preamp and that the preamp is plugged in. The green power indicator should light up. The +48V button should be pushed in. This is to power the microphone which employs a capacitor. The capacitor creates an electric field between two plates. The plates respond to small pressure variations (such as those made by sound waves) creating a signal that is amplified by the preamp and passed on to the computer. Powered microphones are usually superior to un-powered ones.
3. Make sure that the preamp is connected to the computer through the preamp's output jack and the computer's input microphone jack.
4. Load MWSnap by double clicking the camera icon. This program will allow you to take pictures of the screen that you can then email to yourself at home. This will allow you to insert nice figures into your lab report later on.
5. Load Adobe-Audition on the computer by double clicking the speaker icon.
6. To record you need to click on the red spot at the bottom of the screen (record button). Do a sound check by recording your self talking, whistling or singing.

7. While recording ensure that the volume input and outputs on the preamp are at a good level so that the signal is not heavily clipped or extremely faint. You can adjust your distance to the microphone and the input and output volumes on the preamp. To minimize the effect of noise in the room, you would like to record with your noise source (e.g., mouth) close to the microphone. With a couple of recording trials adjust the preamp knobs so that this is possible.
8. **Warning:** I recommend that you do not record in Spectral view as the software often will not be able to keep up with the data rate and you will miss part of the recording (you would see this as vertical bars in the spectral view that should not be there).

Part II – Sliding Whistle

1. Inspecting the whistle: Note that when you blow the air is forced to hit a narrow wedge. The air stream becomes unstable and oscillates as it crosses the wedge setting the air column inside the whistle in motion. The end part can be adjusted so that the air column inside the whistle is either short or long. We want to measure the length of the air column inside the whistle. You can do this using the ruler, measuring the length of the instrument and comparing this to the length when the stopper is all the way in. As shown in Figure 1 the distance between the end of the stopper and the end of the mouth hole is about 3.8 cm. This should be enough information to allow you to calculate the distance between the edge of the mouth hole and stopper inside the tube (see Figure 1 and Figure 2).
2. You can use the tuner to adjust the length of the whistle to play particular musical notes.
3. For each note you blow and record measure the length of the instrument so you can calculate the length of the air column inside the instrument (see Figure 2 caption).
4. Blow a steady stream of air into the whistle while recording and observe the frequency spectrum. Adjust the horizontal (or frequency) axis of your spectrum until you can clearly see the fundamental and overtones. Last year we found it easier to see overtones for the lower notes and harder for the higher notes. Be careful how forcefully you blow into the whistle. Too strong a stream of air will cause the whistle to resonate more in the first harmonic than in the fundamental. If you observe the second peak on your frequency spectrum to be higher than the first, you are hearing more of a harmonic than of the actual tone. Also you will notice that if you blow harder you will get a higher (sharper) note. You can try to keep the force blown similar for all notes.
5. While one of you continues to blow the whistle, the other person can take care of the recording. You can record a series of notes and then use the software to measure the frequencies of the fundamental and its first two overtones for each note.
6. Your goal is to make a table with a list of 7 or 8 notes, each with a calculated air column length based on the length of the instrument, and frequencies of the fundamental and first two harmonics measured from the spectrum of your recording.
7. Errors: When finished, mark on your data sheet how much deviation your frequency measurements had from the frequencies around them. That is, using the cursor, discover how many Hz “lie between” any two pixels on the screen. This will be used to estimate your error graphically.
8. Right click and hold will allow you to expand the axes in Adobe Audition displays so you can zoom in to various regions in the spectrum. Left click while the cursor is in on the frequency analysis box will tell you the frequency of that cursor point.
9. Keep in mind that you will be plotting length versus frequency for the fundamental and some overtones. You will be trying to fit a line to these plots and you will be trying to predict this line. Make sure you have good enough quality data and enough data points (recorded notes) that you will be able to see line going through data points on this plot. Try to get a feeling for how certain your measurements are while you are taking data.

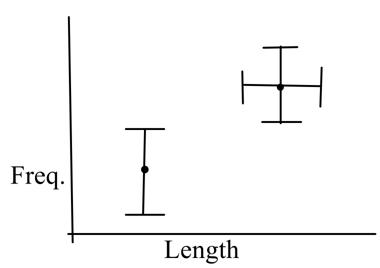
Part III – Voice

1. You will now use the same setup with which you analyzed the whistle to look at the spectrum of your own voice.
2. Record yourself singing a vowel sound.
3. Examine the spectrum of your voice. Does your voice contain multiple tones or frequencies? Are they integer multiples of one another?
4. Record yourself singing different notes. What is the difference between the spectra?

5. Compare the spectrum of the singing voice to that of the whistle.
6. Look at the waveform of your voice. You can blow this up in Adobe Audition by clicking and dragging on the x-axis of the display in waveform view. Is the waveform periodic (does it repeat?). Voice is excited by the periodic beating of vocal chords in the larynx. These aren't actually chords or strings but flaps that beat back and forth. While the motion is nearly periodic it is not a sine wave so the spectrum contains many overtones. Irregularities in the voice are caused by the uneven and non-periodic motions of the vocal chords.
7. Start singing low and then slowly raise your voice past the point where you cross into falsetto. Look at this in the spectrum and see what happens!
8. Record yourself saying different vowels sounds. In spectral view note the strength of the overtones in the ear's most sensitive region 1-5kHz and see if you can how their strength varies for different vowel sounds.

DATA ANALYSIS

1. Using graph paper or a plotting software package, plot a graph of the fundamental frequency produced by your whistle versus the length or the inverse length of the pipe. Physicists often use error bars to account for uncertainties in measurement. Using the difference in Hz between two nearby measurements on the screen of the computer, add error bars to each of your points. You can use vertical and horizontal error bars.
2. There is a sample Excel file here http://astro.pas.rochester.edu/~aquillen/phy103/Handouts/demo.xls if you would like to look at an example of plotting frequency vs length and a curve along with the data. This file uses a single error bar size for all points. The constant of proportionality in Equation 1 is chosen from one of the points. This file plots frequency vs length but you could also plot frequency vs 1/length.
3. Consider plotting a function based on equation 1 on your plot. Are your measurements consistent with frequency proportional to the inverse of the length? To plot a line you will need to estimate or set the constant of proportionality in the equation.
4. Add to the plot, in a recognizably different line, points and error bars for each of the two harmonics you observed for each note.
5. Consider the relation between the fundamental and the harmonics. Are the harmonics exactly integer multiples of the fundamental? Are they near integer multiples? Which integers?
6. For those who are interested in something more challenging: If Equation 1 fails can you find a better equation to fit the data? You could try adding a constant to the length and using a modified or effective length $L_e = L + 0.6D$. The extra term is known as the "end correction." Here D is the diameter of the pipe. Do you think the approximation given by equation 1 is worse when the length of the whistle is short or long?



On the left are examples of data points with error bars. The vertical line going through the data points are twice as long as the measurement error in frequency with which you obtained the point. Notice that the line is centered on the data point and the ends of the line are marked on both the top and bottom, so the error bars for the lower point look like a capital "I." You can also have horizontal error bars (as shown on the point on the upper right) as well as vertical ones representing uncertainties in the length measurements as well as the frequency measurements.

QUESTIONS

1. What kind of relationship did you find between frequency and length of the air column inside the whistle? How might this discovery relate to the equation at the beginning of the lab? Justify any observed deviations from the given equation. Is this equation accurate or not? Does it make sense to plot frequency vs length or frequency vs 1/length to test this equation?

2. For the whistle, what relationship could you determine existed between each fundamental and its two harmonics? Was there a ratio between each tone and its first and second visible harmonics?
3. Compare the spectrum of the singing voice to that of the sliding whistle. Do both have integer ratio harmonics? How many harmonics are visible in each spectrum? Can you see variation in the strength of the harmonics when different vowels are recorded?

Physics of Music

Lab 4 – Making a Fretted Monochord Using the Tempered Scale

EQUIPMENT and MATERIALS

- Guitar pegs (Economy guitar tuners)
- Wood board 8' x 1.5" x ¾" (hardwood) or something thicker if pine
- Guitar strings
- #6 flat washers to hold string end
- Pieces of wood, plastic and rubber to make bridge and nut pieces

- 1" diameter drills, hand drills
- Miter boxes, files, rulers, clamps, vices
- Fret wire, Fret saw (that has width the same as needed for the fret wire)
- Snippers for cutting fret wire flush to the neck.
- Digital Tuners
- Small screwdrivers, extra tuner screws
- Utility knifes
- Hand saws
- Example monochords from last year

Materials 1 monochord per lab group. I ordered the fret wire, saws, snippers and economy tuners from Stewart McDonald. I am still not sure what the best materials are for the bridge and nuts. Popular seems to be ¼"x ¼" wood (pickup at hardware or hobby shop) and same in plastic (ordered from Pastruct?). Also slightly smaller squares are good in plastic. The ¾" poplar does bend affecting the pitch.

Warning: use safety glasses when using power tools. Wear protective eyewear when near an operating drill press. If you are drilling and other people are watching the drill, please make sure they too are wearing protective eyewear.

INTRODUCTION

In this lab we will construct a monochord. We will use the tempered scale to calculate the location to place frets along the neck of the monochord. After we make a working fretted monochord, we will measure the accuracy of our scale. Half notes in the tempered scale have frequency that differ by a factor of $2^{1/12} = 1.05946$. For a string the fundamental mode frequency is proportional to the inverse of the string length.

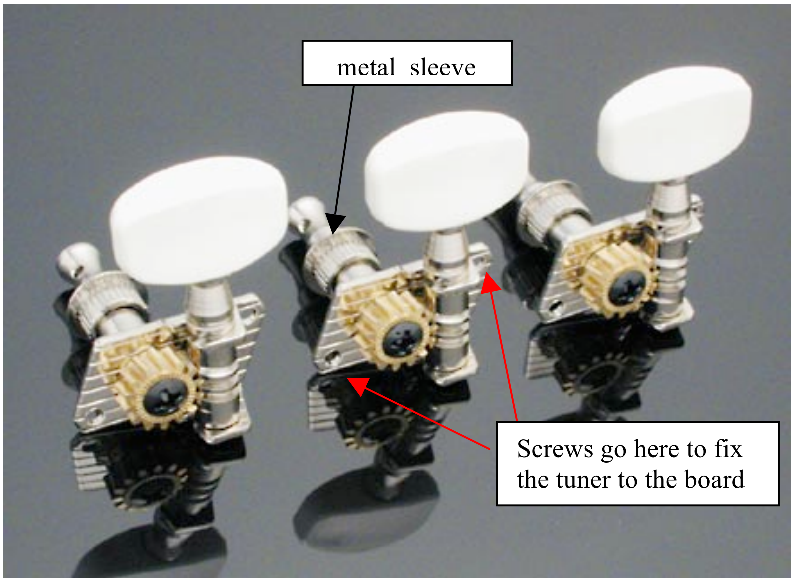


Figure 1: Our monochord will use one of these economy guitar tuners so that we can adjust the tension on the string. We will mount the tuners perpendicular to their normal orientation so that we do not have to have the headstock (see Figure 2) at a different angle than the guitar neck.

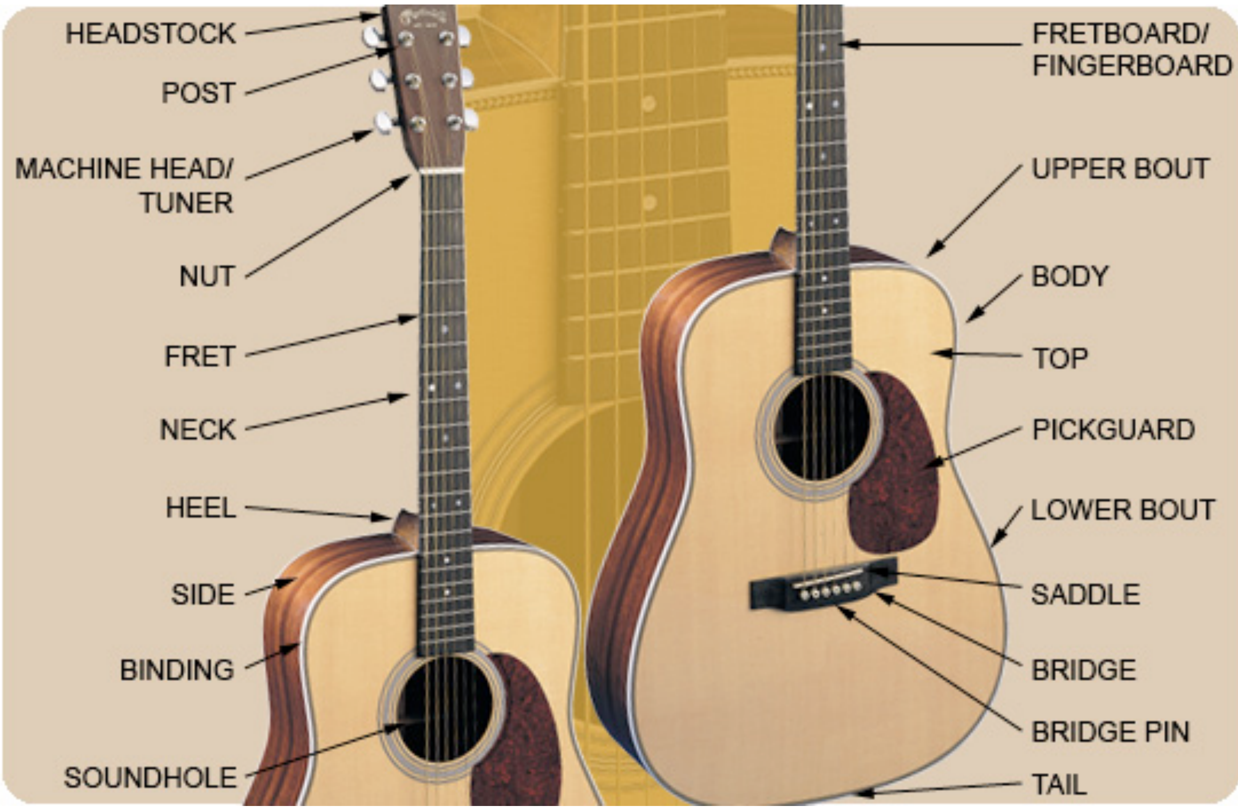


Figure 2. Parts of an acoustic guitar.

CONSTRUCTION

1. **Cut a board.** Cut a piece of 1 ½ x ¾" (actual size) hard wood to a length of about 2' 8". Guitar strings are only 3' long. Please use a hand saw, not a fret saw. The fret saws are a specific width to fit the fret wire and are best used for delicate cutting, not sawing boards in pieces.

2. **Installing the tuner**

Drill first a 11/32" and then a 1" hole as shown in Figure 3. Center the 1" hole so it will be centered on the center of the tuner. Insert the metal sleeve into the 3/8" hole face that is in the 1" hole edge. Drill 2 very narrow holes for the 2 tuner screws. Install the tuner. Screw the two holding screws into the tuner.

A note on drilling: Remember to put a piece of scrap wood below the piece you are drilling into when you are using the drill press. Otherwise you will wind up drilling into the metal table. Start by drilling a small and precisely located hole before you drill a large hole. **Use safety glasses**. Be aware that the wood can catch on the drill and start spinning. It is good to drill large holes with the drill press. You can start a large hole by drilling a small well positioned hole first.

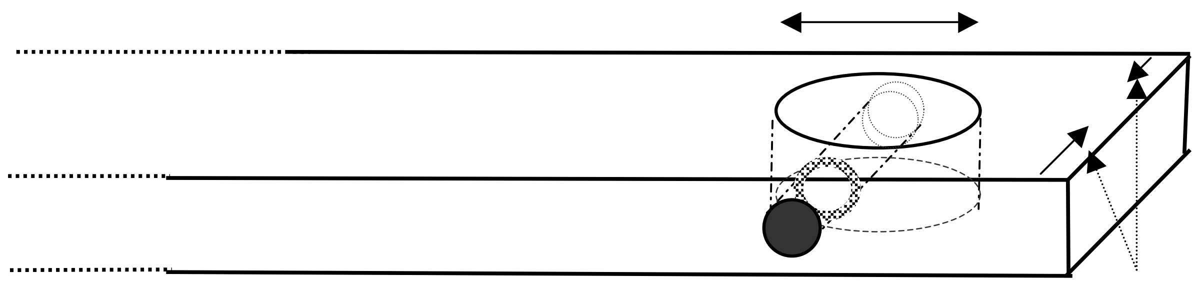


Figure 3 Showing 2 holes needed to install the tuner shown in Figure 1. The metal sleeve shown in Figure 1 is inserted at the location of the orange ring so that the tuner does not wiggle.

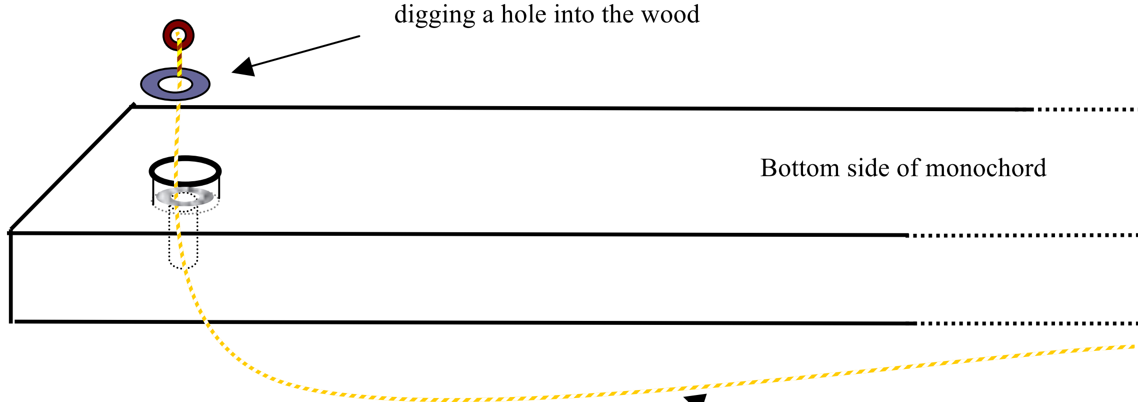


Figure 4: Two more holes must be drilled so that the end of the string can be held.

3. Fixing the end of the string

First drill a hole small enough that the string can pass through the hole (e.g., about 5/64" diameter) but not the round bead at the end of the string. Then drill a hole on the backside of the board just large and deep enough to hold the washer (about 7/16"; see Figure 4). The washer is there to spread the weight from the end of the string out over a larger area. The washer and bead on the end of the string (diameter ~4mm) are sunk into the backside of the monochord so that the board can lie flush against the table.

Figure 5 A fretted guitar neck. Notice the frets are closer at the bottom.

4. The frets

Mark the location where you would like to put the bridge and measure the distance (L) that the string will have between the nut and the bridge (see Figure 2).

Using the tempered scale calculate the location of the frets. The first fret should be at a distance $L/2^{1/12}$ from the bridge, the second fret $L/2^{2/12}$, the third $L/2^{3/12}$ from the bridge, etc….. The frets should be getting closer together as you move away from the tuner.

Using the fret saw and miter box, saw narrow groves into the neck at your marked fret locations keeping the saw perpendicular to the neck. Carefully tap or push the fret wire into each grove. Snip off the ends of the fret wires so the ends are flush with the board edge. File the edges of the frets so the neck edges are smooth. The fret saw should exactly match the width of the fret wire. Try to saw smoothly. If the saw bends while you are sawing the slot cut will be too wide and the fret wire will tend to fall out of its groove.

5. String your monochord

Thread the string and tighten it. The pitch for your monochord will likely depend on the type of string that you chose.

6. The nut and bridge

The nut is the piece of wood or plastic that raises the strings above the neck on the side of the string opposite the tuner (see Figure 2). The bridge is the piece of wood or plastic that raises the strings above the neck on the side of the string opposite the tuner.

I found pretty good luck with ¼" tall wood bridge and 1/8" plastic nut with a groove for the string. You can make the bridge higher than then nut. If the nut is too high it will be hard to press down the strings to the fingerboard. Some adjustment for height angle and grooves is required to get rid of buzzing. Also you can use soft rubber outside the bridge or nut to damp the buzzing. The height of the strings must be kept low so that the frets can be used without too much effort. Cut narrow pieces of wood or/and plastic that are the width of the monochord neck. Experiment with shape and height until you are happy with the sound of your monochord and how easy it is to play. You could cut a groove in the nut to hold the string.

Available in the lab for bridge/nut materials: Hard rubber (durometer 80). Rubber foam with adhesive backing for damping vibrations on the string between the nut and the tuner. Plastruct plastic squares and triangle stock. Hardwood square stock.

MEASURING THE FREQUENCIES OF THE NOTES

Frequencies of Notes in the Tempered Scale 4^{th} octave	
Note	Frequency (Hz)
C4	261.63
C# (D♭)4	277.18
D4	293.66
D# (E♭)4	311.13
E4	329.63
F4	349.23
F#(G♭)4	369.99
G4	392.00
G#(A♭)4	415.30
A4	440.00
A#(B♭)4	466.16
B4	493.88
To predict the notes in the octave above this multiple the above frequencies by two. To predict the notes in the octave below this, divide the above frequencies by two.	

It is easiest to use the digital tuner to measure the frequencies of the notes, though you could also record your monochord and measure frequency using our software. Using the digital tuner (that measures frequencies rather than tunes the string) write in your notebook the notes (C or A#) and number of cents that each note is off. You can use the list of pitch frequencies (given below) to determine the actual frequencies of the notes. For example if you recorded C4+15, do the following:

You know by the table on the left that the C note is supposed to be 261.63 Hz. You know that you are off by +15 cents. You can calculate:
15 cents = 1200 Log $_2$[f/261.63] where the logarithm is base 2
Or 15 cents = 3986 Log $_{10}$[f/261.63] where the logarithm is base 10.

$$(2^{\frac{+15}{1200}}) \times 261.63 = 263.9 \text{ or}$$

$$(10^{\frac{+15}{3986}}) \times 261.63 = 263.9$$

So, the actual frequency of the note is 263.9Hz

Note: 100 cents corresponds to a half step. 50 cents corresponds to a quarter tone. A quarter tone error is considered vastly out of tune for most musical instruments.

QUESTIONS AND ANALYSIS

How accurate is your monochord? Do you think you achieved a good or bad scale?
Could a good musician play music with this monochord? Alone or with other instruments?
Did you use a loose or tight string? Does the string tension affect your scale?
Did you choose a light or a heavy string?
How do you think you could make a better instrument?
What kind of bridge and nut did you find worked the best?
Does your monochord stay in tune?

Take your monochord home and bring it back for the pizeo lab. If you would like to leave it in the lab between now and then, please write your names on the back.

Physics of Music
Lab 5 – The Sonometer – The Resonant String

EQUIPMENT

- Pasco sonometers (pick up 5 from teaching lab) and 5 kits to go with them
- BK Precision function generators and Tenma oscilloscopes
- Digital Tuners
- Sets of wires for the sonometers (box of guitar strings with solderless leads attached)
- Adaptors so output of sound sensors can be connected to preamps
- Preamps connected to computers
- Two of ¼" guitar cables are needed per set up so that the output of the sonometer detectors can be put into the preamps and the output of the preamps can go into the audio input of the computers
- BNC T-s so the sonometer detector signals can simultaneously be seen on the oscilloscope
- Rubber squares to prevent power out of the preamps going into the sonometer sensors

Sonometer sensor warning. Please keep the microphone power in the preamp off if you are sending the sonometer sensor signal through the preamps. One year some sonometer sensors were damaged but we are not sure why.

INTRODUCTION

Many musical instruments, such as guitars, pianos, and violins, operate by creating standing waves in strings. This process has much in common with the creation of standing waves in tubes, as you may examine in more detail in other labs. The frequency of mode *n* for a perfect string that is fixed at both ends is given by the following formula:

$$f_n = n\frac{v}{2L} \qquad \text{(Equation 1)}$$

This formula predicts a relationship between the length L of the string and the frequencies f_n at which the string will resonate. Here n is an integer. The lowest or fundamental mode would have n=1. The velocity of the sound wave, v, is set by the tension and weight of the string. The velocity of waves along a string is given by the following equation:

$$v = \sqrt{\frac{T}{\rho}} \qquad \text{(Equation 2)}$$

In this equation, T is the tension in the wire and ρ is the linear density of the string. The linear density is the mass per unit length of the string.

If you know the linear density of and tension on a string, you can calculate the frequency at which the string will vibrate for a given length (like the length of a guitar neck or of a piano backboard). In our case, we will mount the string on a sonometer. We can vary the tension on the string and see how it changes the frequency at which the string will resonate.

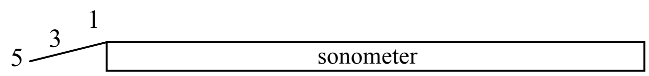
The sonometer also allows us to "drive" the string into vibration (without plucking the string) through the use of a magnetic field. This permits us to test which harmonics can themselves be made to ring within a certain length and type of string. The sonometer will also be used to analyze those parts of the string that are "louder" than others. By replacing the velocity in equation 1 with that given in equation 2 we find the following:

$$f_n = \frac{n}{2L}\sqrt{\frac{T}{\rho}} \qquad \text{(Equation 3)}$$

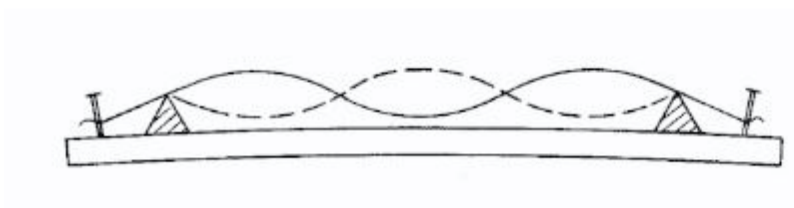
Note that if T is in units of mass times acceleration or g m/s^2 and ρ is in units of g/m then $\sqrt{T/\rho}$ is in units of m/s. This makes sense because it is a speed. If you then put L in units of m (meters) then f_n (using the above fomula) will be in units of Hz. The formula for the tension applied to the string is:

$$T = (notch\#)mg \qquad \text{(Equation 4)}$$

where m is the mass on the end of the string and g=9.81m/s is the gravitational acceleration. Notch number one is the notch closest to the sonometer string, while notch five is that farthest from the sonometer string.



The sonometer comes with two rectangular bars each with a wire. One is the wave driver and is connected to the wave generator. This one will vibrate the string at the frequency set by the function generator. The other bar is a sensor. It is like a guitar pickup. It senses the motion of the string turning it into a voltage that can be seen with the oscilloscope or turned into numbers using the microphone input to the computer.



The above image shows a mode with n=3. Note that there are two nodes where the string is not moving. Equation 1 predicts that this mode has 3 times the frequency of the fundamental and 1/3 the wavelength of the fundamental or lowest mode. A node is where there is little motion; an antinode where there is maximal motion.

The modes of oscillation have frequencies given by equation 1. Equation 1 shows that the frequencies of the modes of oscillation are integer multiples of the lowest or fundamental frequency (that with n=1).

$$f_n = n\, f_1 \qquad \text{(Equation 5)}$$

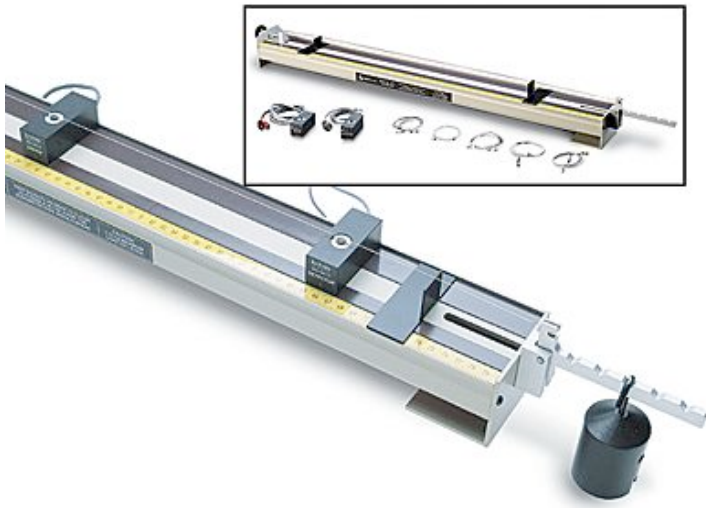
The wavelength of the lowest or fundamental mode, λ_1, is twice the length of the string. The wavelengths of the higher frequency modes are that of the longest one divided by n.

$$\lambda_n = \lambda_1/n \qquad \text{(Equation 6)}$$

Each mode has n-1 nodes.

PURPOSE

The purpose of this lab is to look at the relationship between mass and frequency and tension in vibrating strings. We will try to use this relationship to understand some of the decisions made when choosing strings for different instruments. We also will explore how the string vibrates more easily at certain frequencies that we called modes. The spectrum of a plucked string will be related to its modes of vibrations.



The Pasco sonometer showing weights sound sensor and driver.

PROCEDURE

Part I – Sonometer-Light and Heavy Strings and String Tension

In this part of the lab we will explore how changing string density and tension affects pitch.

1. Choose a few guitar strings to investigate. Place a string on the sonometer . Tighten the string by placing weights on the end with the notches. By using the weights you can measure the tension. If you tighten the string by screwing in the knob at the end of the sonometer you will not be able to measure the string tension with the weights.
2. Measure the pitch or the frequency of the fundamental mode. The easiest way to do this is send the output of the sonometer detector/sensor into the preamp and then record its output with Adobe Audition. To send the output of the sonometer into the preamp you need to use an adaptor that has one end a BNC and the other end a ¼" connector. To record you will need to adjust the preamp amplification. After recording you can measure the frequency of the fundamental mode using the spectral analysis window. Note you will need to adjust the size of the FFT window and you might experiment with choosing the type of FFT window function (see under advanced options).

Warning: Do not record in Spectral view as the software will not be able to keep up and you will miss part of the recording (you would see this as vertical bars in the spectral view that should not be there).

Note: If you put the FFT on a lower size (less than 8916) then the spectral analysis window will update in real time. Right click on the level bar at the bottom and choose "Monitor record level" to have it update in real time!

3. Measure the pitch of a plucked string. Increase the tension on the string (the weight on the end of the sonometer). You should find that the pitch rises.
4. Measure the pitch of two different strings under the same tension. Consider why you might choose lighter strings for a soprano instrument.
5. Try increasing the string tension of a **light thin** string so that you raise the pitch of a string by an octave. How much do you need to vary the string tension to raise the pitch by an octave? Compare your measurement to what you might expect by studying equation 3. Equation 3 predicts that frequency is proportional to the square root of the tension. If you raise the pitch by an octave that means the frequency of the fundamental is twice as high. Note: there are extra weights in the cabinet.
6. Place a light thin string (not a bass one) string on the sonometer and reduce the tension. Try to set it up to make a low note. What happens when you reduce the tension on the string to extremely low levels? Consider why one would choose heavier and longer strings for bass string instruments.

Part II – Sonometer-The Resonant String

In this part of the lab we will excite the string at different frequencies and look at the amplitude of the motions on the sstring. At some driving frequencies we will see a large amplitude response on the string. At these frequencies the string is said to be in resonance.

1. Set the oscilloscope on dual mode. The trigger should be set to the channel associated with the driver. The sound sensor is input into the other oscilloscope channel so you should be able to see both signals. If you are not getting a signal, check the DC offset on the function generator. Position the driver about 5 cm from the bridge and the detector near the center. Attach a weight mass to the end of the sonometer.
2. Use the instrument tuner to estimate the starting frequency of resonance (The frequency of resonance occurs when the string is plucked.) Set your function generator approximately 100 Hz below the frequency of resonance. The function generator will attempt to magnetically "drive" the string at the rate you have set. Slowly increase frequency until you find the first point at which the string resonates. You will know that the string is resonating because you will hear the string vibrating and you will observe a change in the waveform displayed by the pickup on your oscilloscope. Also you may see a peak in the spectrum shown in Adobe Audition if you have set it to monitor the record level and the length of the FFT is less than 8196. Remember, the pickup (aka sound sensor) and "driver" both create magnetic fields, so if you see a perfect sine wave coming from the pickup, you know that you are seeing traces of the driving signal. The real resonance may be a more complex waveform (I mean it might be excited at more than one frequency at once). Record the frequency of resonance using the number shown on the function generator.
3. After you find a resonance, experiment with moving the sensor. Are there locations where you detect a stronger signal than others? Can you find nodes and anti-nodes? See the above illustration.
4. Try going in and out of resonance while looking at the spectral analysis window in Adobe Audition while the spectral analysis window is updating in real time. (Make sure the FFT size is not high). See if you can see spectral peaks appearing and disappearing as the string goes in and out of resonance.

Note: We have found that often a frequency twice that of the driving frequency is excited on the string. We are not sure why this is true.

5. Continue to increase the frequency until you find several more resonating frequencies and record these on your data sheet. These resonating points will correspond to the string vibrating with two, three, four or higher numbers of nodes respectively.

DATA ANALYSIS AND QUESTIONS

1. Discuss your best guess on what integer modes you excited on the string with the frequency generator and wave driver. Base your interpretation on the frequencies of modes excited (they are integer multiples of what base frequency?) and how many nodes and antinodes you might have seen by moving the sound sensor.
2. Do you confirm that the frequency of the fundamental mode is proportional to the square root of tension?
3. When you varied your pickup (sound sensor) location, did you see any difference in output amplitude? Can you explain this using illustrations such as this one?

4. Knowing what factors go into changing the resonant frequencies of a string, discuss what choices for string tension, length and weight you would make for the design of different types of soprano and bass stringed instruments.

Physics of Music

Lab 6 – The Sonometer – String Linearity and Timbre change after plucking

EQUIPMENT

- Same as previous lab but we no longer need the function generators or the sonometer drivers.

Sonometer sensor warning. Please turn off the microphone power in the preamp if you are sending the sonometer sensor signals through the preamps.

INTRODUCTION

The frequency of mode n for a perfect string that is fixed at both ends is approximately given by the following formula:

$$f_n = \frac{n}{2L}\sqrt{\frac{T}{\rho}} \qquad \text{(Equation 1)}$$

This formula predicts a relationship between the length L of the string and the frequencies f_n at which the string will resonate. Here n is an integer. The lowest or fundamental mode would have n=1. In this equation, T is the tension in the wire and ρ is the linear density of the string. The linear density is the mass per unit length of the string. The above formula predicts that each harmonic is an exact integer multiple of the fundamental. A wave equation that predicts Equation 1 is said to be linear. Equation 1 predicts that f_n is a constant times n.

Non-linearity: Equation 1 predicts that each overtone is an integer multiple of a fundamental frequency. While this equation is a good approximation, it does not succeed in predicting the exact frequencies of the overtones, particularly for thick strings. In this case the string is said to be "non-linear". The reason for this is that the wave equation describing motions on the string contains higher order or non-linear terms and so would give corrections to equation 1. In this lab we will test this equation by trying to measure as many overtones as we can and see if their frequencies deviate from those predicted by equation 1. String non-linearity is particularly important for instruments with heavy base strings. For example, the overtones on a piano are systematically sharp compared to integer multiples of the fundamental. In this case f_n is a constant times n + another term. We could write for example $f_n = An + Bn^2$ with A and B constants. For the piano the constant B is positive and so the high overtones have a higher frequency than predicted with B=0. Because our hearing is more sensitive to overtones for the base notes, base strings are purposely tuned flat so that their overtones are in tune with other notes on the piano.

Timbre changes: After a string is plucked many modes of oscillation are excited. However each mode or overtone may decay at a different rate leading to a change in the timbre of the sound. We can try to measure this affect by comparing the strength of one of the higher overtones to the fundamental at different times after plucking. A string that has strong higher overtones at longer times after plucking is said to have a "bright" sound. A strong with higher overtones that quickly decay after plucking is said to have a "soft" sound. The decay rate of overtones is affected by string composition, structure and tension. In this lab you have the opportunity to test out steel and wound phosphor bronze and nickel plated strings. Phosphor bronze strings are considered softer or less bright than steel strings. We can search for differences in timbre between the two string types. Nickel/steel strings are often chosen for electric guitars. Heavier strings wind up under higher tension for the same pitch. A string under higher tension is harder to play (you need more force to push the string down). The nickel/steel string set are lighter than the phosphor bronze set so they are intended to be under lower tension and so easier to play. There may also be differences in the time the note is sustained after plucking (how long the note is loud).

Part I – Sonometer-String Linearity

The above equation 1 implies that all modes are exact integer multiples of the fundamental. To test whether this is a good approximation we will measure the frequencies of as many modes as we can and see if they are indeed integer multiples of the lowest mode.

1. Connect the output of the sound sensor to the preamp. To do this you will need to use an adaptor that has a BNC connector on one end and a ¼" connector on the other. Adjust the preamp so you can record sounds through it onto the computer. Pluck the string while recording the output of the string sensor through the premap. Measure the frequency of as many harmonics as you can. I found that a heavy string allowed me to measure more overtones because its fundamental frequency is lower. Also a string under higher tension tends to have stronger higher overtones. The higher overtones are strongest just after plucking.
2. Plot the n versus the frequency of each harmonic. Do these lie on a line as would be predicted by equation 1? Equation 1 is based on a linear model for wave propagation on a string. If you find that Equation 1 fails to perfectly predict the frequencies of all the modes then you have found that the waves on the string are "non-linear". You will probably need to measure at least 30 overtones to see the non-linearity of the string.

DATA ANALYSIS

1. Plot the frequency of overtones as a function of overtone number, n. Try to plot as many as you can (>30). Does a line fit the data points? To figure this out plot a line that goes through the first 10 overtones and see if the higher overtones lie on the line. Are the high overtones above or below the line? Is equation 1 a good approximation? If equation 1 fails to predict the high frequencies can you propose a better equation that would fit your data?
2. Consider a bass instrument with a string that is non-linear. How would you tune it so that you can play in harmony with soprano instruments? The octaves on a piano are stretched to take this into account --- the deviation between expected and actual tuning of strings on the piano is known as the Railsback curve.

Part II – Timbre change after plucking and how timbre depends on tension.

1. Pluck a string while recording through the preamp. Compare the spectrum of the sound just after plucking and a couple of seconds after plucking (keep note of the times!). Does the shape of the spectrum (strength of frequency of overtones compared to one another) change in time? You may wish to measure the amplitudes of the overtones at the two different times. Measure the strengths of the overtones in dB using the spectral analysis window.

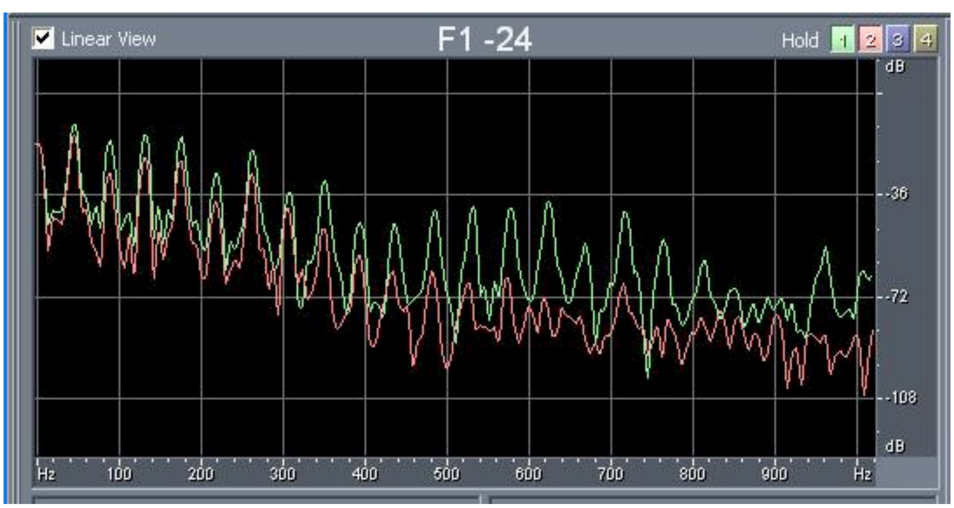


Figure 1. This shows the spectrum of a plucked string just after plucking and 3 seconds after plucking. I used the hold buttons on the upper right of the spectral analysis box to display both spectra on the same plot. Note that the strength of the fundamental is not much decreased after 3 seconds but the strength of the higher overtones has dropped by 30dB! After 3 seconds we would describe the sound as duller or less bright.

2. Note that the spectral tool in Audition measures amplitudes in dB (decibels). The decibel is the 20 times the log10 RMS (root mean square) plus a constant. Adobe Audition does dB with respect to the maximum volume possible before clipping, so 0dB is near a signal that is clipped.
3. Timbre variations depend on the type of string and its tension. We have two sets of guitar strings, one intended for an electric guitar (nickel plate wound) and one intended for an acoustic guitar (phosphor bronze wound). The acoustic guitar strings are heavier and intended to be played at about twice the tension of the electric guitar strings. Choose one string from each of the sets. If you choose an A string from the phosphor bronze set choose the A string from the nickel plate set. People had trouble last year with the low E strings becoming stuck in the sonometers, so you might not want to choose the low E strings. Adjust the tension of the A string (if you chose that string) on the sonometer so that the string sounds an A when it is plucked. Record the sound and wait for it to decay for at least 10 seconds. Then record the second A string, again adjusted to sound an A. If you chose G strings then adjust the tension so a G is sounded.
4. Compare the spectra (timbres) of the 2 strings just after they are plucked. Which string has stronger higher frequency overtones?
5. Compare the spectra of each string just after plucking, one second after plucking, 3 seconds after plucking and 6 seconds after plucking. Do the decay rates of the overtones differ between the two types of strings?

DATA ANALYSIS

6. Show a plot that shows the strength of the overtones at different times after plucking for a string at two different tensions or two different types of strings. Do you see changes in volume of the overtones as a function of time after plucking?
7. Did you observe differences in the strengths of the overtones in the two types of strings?
8. Did you observe differences in the decay rates of the overtones in the two types of strings?
9. The second part of the lab can be done qualitatively with figures instead of tables with lists of numbers.

QUESTIONS

1. Piano strings are non-linear and the high *n* overtones are systematically higher than predicted by equation 1. Discuss your measurements of the frequencies of overtones and the accuracy of equation 1. Consider what this means for the tuning of strings on a piano or a bass instrument that is played in harmony with soprano instruments.

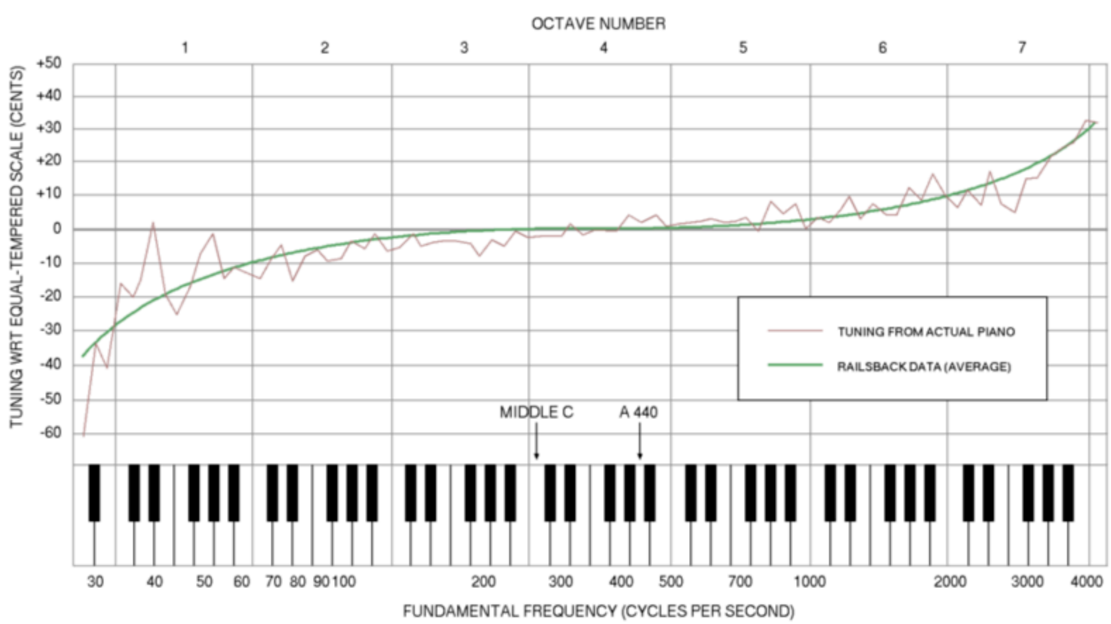


Figure 2. The Railsback curve, indicating the deviation between normal piano tuning and an equal-tempered scale. The base notes are purposely tuned flat so their overtones are in tune with other notes. The soprano notes are purposely tuned sharp.

2. Two sets of strings are available in the lab: 1) nickel wound super light gauge intended for low tension and use on electric guitars. 2) phosphor bronze round wound strings intended for use on acoustic guitars but at twice the tension of the other set. Since both strings are meant for the guitar and so meant to play the same notes, the increased tension for the acoustic guitar strings implies that their linear densities are higher than the nickel wound strings. Based on your experiments with timbre changes do you have ideas on why a musician might prefer the phosphor bronze set for an acoustic guitar but the nickel wound set for the electric guitar? Which ones are better for playing sustained notes and which ones for plucked notes? Which set is brighter (stronger higher frequency overtones)?

Physics of Music

Lab 5 or 6 Alternate—Amplifying the Monochord with a Piezo-electric Contact Pickup

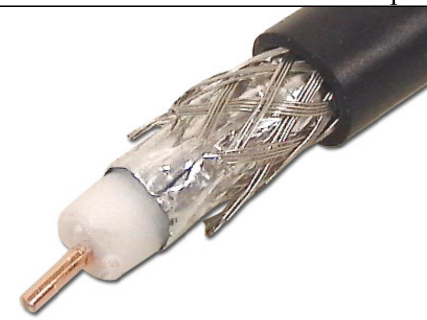
EQUIPMENT and MATERIALS:
Soldering irons + solder + heat guns, multimeters x5
Exacto-knifes, wire cutters, pliers, wire strippers, scissors x5
¼" instrument cables
Guitar amplifiers (e.g., Crate) right now we only have 2 of these, but we can put them out so people can play their amplified instruments. Instrument cables for these amps.
The usual recording setup with preamps, microphones etc.
Bobby-pins, clothespins, rubber bands, C-clamps, metal sheets, snips

Heat Shrink tubing
Piezo Film Tabs (ordered from Parallax.com)
¼" phone in-line mono jacks
conductive shielding copper tape, insulating electric tape
coaxial cable (I recommend something narrow and flexible, i.e., braided both shield and central conductor, and make sure you can solder easily to the shield, We have something called RG174/U 50 Ohm that seems nice)
Materials: for each monochord: cable, piezo tab and jack. Piezo ordered from parallax.com. Cable and jacks ordered from PartsExpress. Copper tape from Stewart-McDonald.

STUDENTS: You may want to bring the monochord you made previously!

INTRODUCTION:

Some recently developed and very popular instruments are electronically amplified. In this lab we will build our own contact pickups and amplify the monochord we built in the previous lab. Piezo electric materials are commonly used in mechanical-electrical devices such as speakers, buzzers and microphones. A variety of experimental instruments can be made by placing pickups on or in vibrating objects. Piezo-electric materials are crystals that respond to deformation by producing a voltage. When a vibration passes through the crystal it generates an alternating voltage. The signal can be sent to an amplifier and then a loud speaker. A piezo electric pickup is a cheap way to amplify a stringed instrument, making it possible to play the instrument without a sound-board. The output from a piezo-electric is weak (or high impedance) and so will not travel far. You can amplify the output with a guitar amplifier or use a preamp. The piezo-electric material is often placed on or under the bridge of the instrument. Contact pickups can have a good strong signal and they are not prone to feedback or picking up noise. However they also amplify surface sounds such as scraping or knocks as well as vibrations from the strings of an instrument. The vibrational spectrum picked up by a contact pick-up mounted at the bridge of a stringed instrument may differ from that emitted by the instrument in the air. This is also true of magnetic pickups. However people tend to like the sound from well-designed magnetic pickups placed under strings and often do not love the sound from a contact pickup placed under a bridge. Consequently most electric guitars use magnetic pickups not piezo-electric ones.

Table 1: Components for our piezo-electric pickups	
	Coaxial cable showing inner copper conductor and an outer braded metal shield. There are two insulators. Here the outer one is black and the inner one is white. The inner insulator separates the inner conductor from the metallic shield. The outer insulator protects the braided metallic shield.
	¼” phone in-line mono jack. On the bottom is a metal interior sleeve which screws into the black outer sleeve. You can see two terminals or leads on the metal part. The inner wire of the coaxial cable is soldered to the smaller of these two leads. The metal shielding of the coaxial cable is soldered to the larger of the two leads.
	Piezo-electric film tab. The leads are not very big or long. We can solder to the leads but should be careful not to heat up the tab too much or we will damage the film. To protect the film we can put a clip in between the spot we are soldering and the rest of the tab to act as a heat sink slowing heat conduction to the tab. See Figure 1, below.

Soldering:

To construct our pickup we need to solder one end of a coaxial cable to the piezo tab and the other end to the jack. The coax cable has two conductors, an inner one and a shield. Each will be soldered to a lead of the piezo tab and a lead of the jack.

Warning: Soldering irons are very hot. Do not touch the tip!

Is your soldering iron ready? First check that the soldering iron tip is hot. You ***don’t*** want to touch the tip with your finger to find out! Instead touch the end tip of the iron with the end of some solder. If the iron tip is hot you can ***tin*** the tip by melting some solder onto the end of the iron. Brush the solder off with either a wet sponge or the wire sponge. The tip of the iron should be thinly covered with melted solder and so should look shiny. Later on if you have trouble getting good joints you can consider tinning the wire and leads (particularly the coaxial shield) before you solder them together.

Soldering a wire to a lead: Heat the lead and wire with the soldering iron, then touch the solder to the wire and lead. The solder should wick into the cracks between the lead and wire. Take away the soldering iron, letting the joint solidify without moving it. If the joint looks cracked because you jiggled it when it was solidifying, the connection may not be good. Soldering requires about 4 hands: one for the iron, one for the wire, one for the solder and one to hold the object with the leads. One can make do with fewer hands with clips and clamps. You can also twist or thread the wire onto the lead before soldering.

Here we will not only be trying to make good conducting joints but also trying to solder fast enough that we don't damage the piezo films. Before soldering, place an alligator clip at the base of the lead to the piezo film. Then solder the end of the lead (see the picture below). The alligator clip serves as a heat sink reducing the conduction of heat along the lead. The soldering iron touches the lead just to the right of the alligator clip end in the figure below.

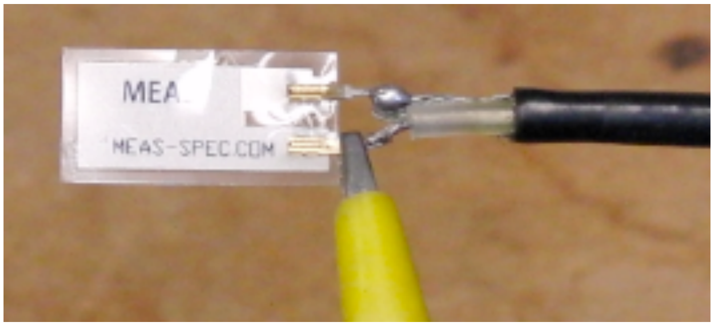


Figure 1. Using an alligator clip as a heat sink to protect the piezo film from getting too hot while soldering.

CONSTRUCTION:

The recipe we follow here to construct the pickup is essentially that by David Fittel
http://fittell.id.au/piezo/

1. Cut or find an approximately 6'' length of coax cable. I find best results with a very flexible narrow coax cable. Strip about ½" of the outer insulation from both ends. Unwrap the braided shielding and strip about ¼" inch of the inner insulator from both ends.
2. Soldering the cable to the jack:
    a. Cut a small piece of heat shrink tubing, place it over the inner wire of your coax cable. Then solder the inner wire to the inner lead of the jack. See the above notes about soldering! Some fo the jacks have two leads. You want to push them together with a pliers and solder to both.
    b. Solder the shield of the coax cable to the outer lead of the jack.
    c. Push the shrink tubing over the smaller lead. Shrink the tubing with the heat gun so that the inner lead and wire are well insulated and will not touch the shielding when you assemble the jack.

Figure 2. The jack showing some black heat-shrink tubing.

    d. If your jack has a plastic outer case you can additionally shield the jack by wrapping it with copper tape. Wrap in such a way as to touch the shielding of the cable but make sure the inner conductor is not shorted to it! If your jack has a metal case then no extra shielding should be necessary. If you didn't use heat shrink you can insulate with tape.
    e. Screw together the jack. The jack end of your pickup should now be finished.

3. Soldering the coax cable to the piezo tab:
    a. Clip an mini-alligator clip to the tab lead just inside where you want to solder it. The clip should act like a heat sink, protecting the tab from too much heat.
    b. Delicately solder the inner wire of the coax cable to one lead of the piezo tab. Try to solder fairly quickly. If your coax cable is flexible you can loop the connector around the pins before you solder.

c. Move the alligator clip to the other lead and delicately solder the shield of the coax cable to the other lead of the piezo tab. You may want to tin the shield before you solder it to the piezo tab lead.
d. Cover the lead that goes to the inside wire of the coax cable with insulating electrical tape to make sure that it never touches the other lead and conductive tape that we will use to cover and shield the entire tab.

Figure 3. The lead connected to the inner wire in the coax cable has been covered with insulating electrical tape. However the other pin is bare. When you cover with copper conducting tape, the inner wire will not be shorted to the shield, and the tape will touch the shield. This way shield can cover the eintre tab.

4. Place the end of the piezo film under the bridge to your monochord or clip it to some other object that you want to amplify. If you are amplifying a monochord you will have to loosen the monochord string to do this.
5. Check out the pickup to see if it works! Connect the output to a guitar amplifier or to the preamp. If your pickup does not amplify the monochord, check the contacts and leads for lack of continuity and shorts with a multimeter. Lack of continuity means you have bad or broken connections. A short is when there are stray wires or solder connecting the two leads of the jack, piezo tab or inner wire and shield of the cable. To check for either of these you set the multimeter on `ohms' (symbol Ω) so you can measure resistance. Zero resistance is good continuity (or a short if you are looking across the two leads). Infinite resistance is a broken connection.
6. If the pickup works you can use conductive copper tape to cover and so shield the tab. I found by measuring the noise before and after that it is a very good idea to cover the entire thing with shielding tape.

Figure 4. The piezo tab is shielded with conductive copper tape and is placed under the bridge of the monochord. The bridge here is a sandwich of rubber and plastic.

<u>EXPERIMENTATION:</u>

Choose one of the following experiments.

1) Comparison of amplified to non-amplified monochord. Compare the sound of the plucked monochord amplified through the pickup to that picked up from a microphone near the string. Compare the sound quality, the spectrum at different times after plucking and the shape of the envelope of the sound. Is there a difference in the sustain? Is the sound brighter or duller?

2) Exploration of piezo placement on the monochord. In Figure 4 above I was trying a bridge sandwich with wood on the bottom, then rubber, then the piezo tab pickup then the plastic and then the string on top. Experiment with different types of bridges (you could put the pickup under the rubber instead of on top of

it, or use a larger piece of softer rubber). Or try placing the pickup in a different spot than in a bridge sandwich. Compare the sound quality, spectrum and sound envelope for your different piezo placements.

3) Making Piezo music. Put together an experimental object/instrument to amplify with your contact pickup. The pickup must be firmly attached to something vibrating. Explore how varying your design changes the sounds from your creation. I had fun with clothespins, bobby pins, metalsheet, C-clamps and rubber bands.

On displaying two spectra at once in Audition: The hold buttons in the spectral analysis window in Audition will let you look at and show two spectra at the same time. If you press a hold button it will keep one spectrum displayed. You can then move to a different time in the waveform window, click, and then look at the two spectra together in the frequency analysis window.

On comparing the timbre of two plucked notes: The spectrum of a plucked string depends on time, as well as how you plucked the string and where you plucked the string and the pitch of the string. To compare the spectral characteristics of two different plucks you need to try to match these quantities. You also need to understand how differences in plucking and time after plucking affects the spectrum. Suppose you compare two different sounds and find the spectrum of the second has weaker high overtones at about 3kHz. The strength of high overtones in the spectrum of a plucked sound decay more rapidly than the low ones (see the second sonometer Lab). So you must be sure that you have not looked at the second sound at a later time after plucking. The strength of the overtones also depends on where you plucked the string and how hard you plucked. You could try to pluck the string the same way every time.

Physics of Music

Lab 7 – Constructing a PVC Flute

EQUIPMENT

- PVC pipe The instructions are for ¾'' diameter PVC 480 PSI or 200 PSI. The thickness of the PVC depends on the PSI rating.
- Corks or dowels that fits into the end of the PVC pipe (#9 corks for ¾"ID pipe 480 PSI, ½" dowel for ½"ID pipe). Note 200 PSI PVC requires #10 corks or Diam II Wine Corks 23.5mm diameter which I picked up at a home brewing place)
- Rulers, in cm
- Tools: power hand-drills, drill bits, hacksaws, round and flat files, hammers, center punches, matt knifes
- Dremel tools
- Mini vices
- Protective eyewear
- Tuners for measuring frequency
- Plumber's goop for sealing the ends. Or wood filler. Or glue-guns.
- Sandpaper
- Mirrors, antiseptic mouthwash

Note: two figures in this chapter are from Hopkin's book "Musical Instrument design."

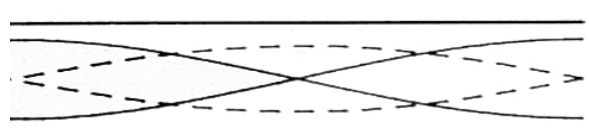
Materials: Every student should make their own flute.

Warning: If you share your flute or borrow somebody else's sanitize it first! Use the disinfecting mouthwash or wash the flute in a bathroom sink before you blow into it!

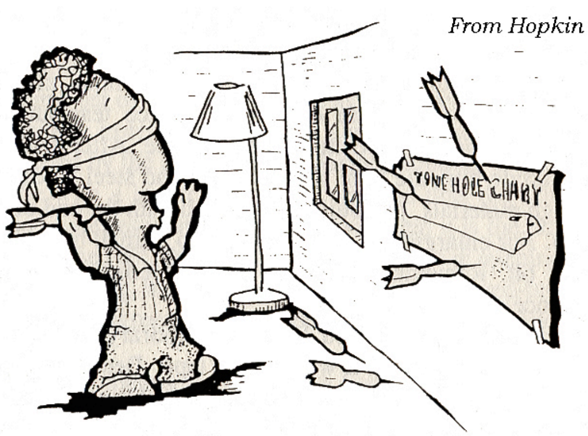
Warning: Use protective eyewear when you are near operating drills. When you are drilling make sure that everybody watching the drill is wearing protective eyewear.

INTRODUCTION

Here we will build our own PVC pipe flute. PVC looks nasty but I have found that the tone of the flute has a lovely soft bamboo like sound. It is possible to make a beautiful instrument with PVC. The challenge is to make it playable, and in pitch. A flute when blown can be considered a vibrating resonant column of air with two open ends.



From Hopkin



The Scientific Approach to Tonehole Placement.

When constructing your flute, you will need to understand where your holes should go. You can use the following approximate relation

$$L_1 f_1 = L_0 f_0 \qquad \text{(Equation 1)}$$

where f_0 and f_1 are frequencies corresponding to the fundamental tones for pipes of lengths L_0 and L_1, respectively. Lengths are measured between the mouth-hole and the first open finger hole.
The above equation is consistent with

$$f_0 = \frac{v}{2L_0} \qquad \text{(Equation 2)}$$

predicted for the fundamental mode of a pipe with two open ends where v is the speed of sound.

If you wish to make a flute in G or F, the hole placements are estimated below for you in the Tables. They should approximately be consistent with the above equation. In your report you could compare the frequencies predicted from the above two equations and those you actually measure. Unfortunately the above equations do not exactly predict the actual notes played by a flute. The diameter of the tube increases the effective length of the resonating tube (this is known as the end correction). A larger diameter hole decreases the effective length of the resonating tube so you can raise the pitch of a tone by enlarging a hole. Large tone-holes produce a larger volume of sound. Larger holes should also contribute to better tone quality by providing a louder tone richer in overtones or partials.

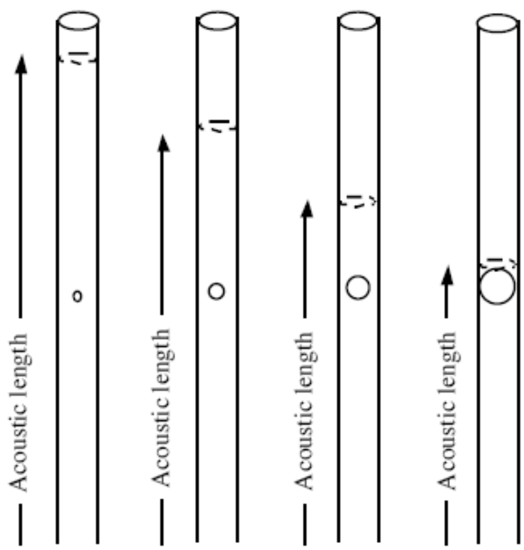
The end correction: A cylindrical tube with two open ends actually behaves as if it were slightly longer than its length. A better approximation to the actual fundamental frequencies of a tube can be made with the following formula.

$$f_0 = \frac{v}{2L_e} \qquad \text{(Equation 3)}$$

where v is the speed of sound. L_e is the "effective length of the tube, measured to be

$$L_e = L + 1.226R \qquad \text{(Equation 4)}$$

where L is the actual length of the tube and R is its radius. Note the above two equations (while an improvement from equation 2) still won't predict exactly the frequencies of a tube with multiple holes in it. Equation 4 above has taken into account the two open ends.



A hole drilled on the side of a pipe changes the effective acoustic length of the pipe. The larger the hole, the closer the acoustic length will be to the hole position. Figure from www.tufts.edu/as/wright_center/physics_2003_wkshp/book/

Pitch measurement: Pitches are commonly measured with respect to the frequencies of the tempered scale with a concert A of 440Hz. These frequencies are listed in the table below and in Appendix C. Tuners usually give the nearest note on the tempered scale and the difference between this not and the one you played. This difference is given in **cents**. Cents are defined in the following way: There are 100 cents in each half tone, and twelve half tones in an octave. So there are 1200 cents in an octave. An octave corresponds to a frequency change of a factor of two. In other words a second note that is an octave above a first note has twice the frequency of the first. Consequently 1 cent corresponds to a factor of $2^{\frac{1}{1200}}$. If you are sharp by +21 cents you multiply the frequency of the nearest tempered scale note by $2^{\frac{+21}{1200}}$ to calculate the actual frequency of your note. If you are flat by 18 cents you would multiply the nearest tempered scale note by a factor of $2^{\frac{-18}{1200}}$.

PURPOSE

The purpose of this lab is to relate your knowledge of physics and music to the construction of a musical instrument. It is quite difficult to make a flute that is easy to play and that can play notes that are in tune. In this lab you may discover that the simple numerical estimates for pitch (that given by the above equations) are not exact. By creating this instrument we can perhaps gain respect and admiration for the design and redesign effort that goes into many musical instruments.

On making accurate and clean holes in PVC: Before you begin you can mark each tonehole position accurately with a center punch. It is good to start by drilling a small hole. You only want to drill through one side of the PVC tube. Don't push the drill in too deeply as you don't want to drill through the opposite side of the PVC tube! After you have made a small hole, you need to widen it. Widen it using a somewhat larger drill bit. When drilling larger holes it is best to **drill slowly into the PVC!** Slowly widen the hole progressively using larger and larger drill bits, never skipping more than 1 size. The PVC will be less likely to jump or vibrate as you drill and your hole edge is less likely to become chipped when you drill slowly and enlarge the hole drill size by drill size. Afterwards de-burr the edges of the hole with a matt knife or sand paper.

Everyone should make his or her own flute! PVC is cheap. If you don't like your flute, throw it out and try again!

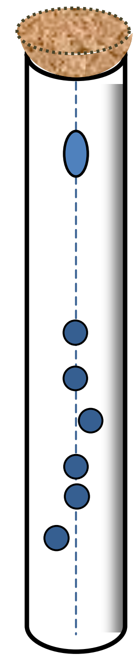
We have two different thicknesses of ¾" diameter PVC pipe, the thicker 480PSI (pounds per square inch pressure) and the thinner 200 PSI. I have made two tables, one for a G flute and the other for an F flute. These tables could be improved.

Measurements for a side blown G flute made with ¾" diameter PVC pipe 480 PSI				
Hole Number	Measurements in cm from the end near the blow hole	Measurements in cm from the blow hole	Intended note played when all holes are covered up to this one	Hole diameter in inches
END	0 (#9 cork in this end)	-3.0		
0	3.0 (center of blow hole)	0.0		3/8" but widened to an oval with file to ½"
#1	18.2	15.2	F#	11/32"
#2	20.8	17.8	E	11/32"
#3	23.8	20.8	D	11/32"
#4	27.8	24.8	C	11/32"
#5	29.7	26.7	B	11/32"
#6	33.5	30.5	A	11/32"
END	41.3 (open end)	38.3	G	11/32"

Measurements for a side blown F flute made with ¾" diameter PVC pipe 200 PSI				
Hole Number	Measurements in cm from the end near the blow hole	Measurements in cm from the blow hole	Intended note played when all holes are covered up to this one	Hole diameter in inches
END	0 (#10 cork in this end)	-3.0		
0	3.0 (center of blow hole)	0.0		3/8" but widened to an oval with file to ½"
#1	18.8	15.8	E	11/32"
#2	21.7	18.7	D	11/32"
#3	25.1	22.1	C	11/32"
#4	29.3	26.3	B♭	11/32"
#5	31.3	28.3	A	11/32"
#6	35.3	32.3	G	11/32"
END	43.6 (open end)	40.6	F	11/32"

PROCEDURE

1. With a saw cut your PVC tubing to the length given in the above tables that depends on whether you have 200 or 480 PSI PVC. The length is the last number in the second column. De-burr the ends.
2. Tap your cork into one end of the tube. Make sure it is a tight fit. It the cork has holes in it and the plugged end leaks you can plug the small holes with goop (like silicone caulk). You may need to trim the cork with a knife so that it sticks in only about 1/3".
3. With a pencil and a ruler mark a line down the tube. You will place most of holes along this line. See the figure to the right showing hole placement and the center line.
4. To form the blowhole, make a 3/8 inch hole in the side of the tube one 3cm from the end of the PVC tube that has the cork in it. Start by drilling a much smaller hole and slowly widen it. After the hole reaches 3/8" diameter, file the hole out so that you make an oval shape, about ½ of an inch in the long dimension, with the long dimension along the line of the tube. You can use a file or a Dremel tool to enlarge the whole. Smooth and de-burr the edges of the blowhole hole.



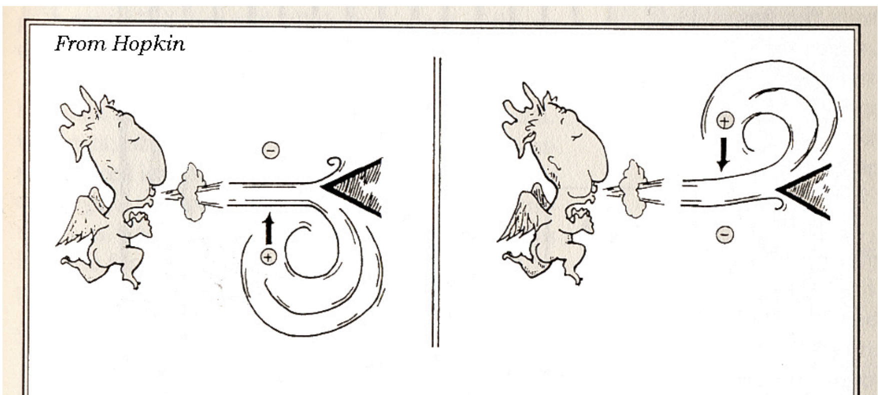


FIGURE 6-2: Edgetone mechanics. At the start of the tone, the major part of the air flows to one side of the edge. This results in a pressure buildup on that side, which deflects the air stream and forces the majority of the air flow over to the other side. This causes a pressure buildup there; the process then reverses and the cycle repeats.

5. Practice playing your flute. You need to blow so that your air-stream hits the opposite edge of the mouth-hole. Place the flute against your lips and turn the flute back and forth while blowing until you can hear a breathy note. Adjust the angle of the flute and the way you blow until the note becomes stronger and purer. It may help to look in a mirror while blowing. If you share your flute or borrow somebody else's sanitize it first! To play more expressively try blowing a vibrato.

6. Measure the pitch of the flute lacking any finger holes. To measure the pitch, play your flute while looking at a tuner. Or you can play your flute while recording and then use the frequency analysis tool in Adobe/Audition as a tuner. You may need to adjust the length of the FFT for a pitch to be estimated. The pitch should appear on the top of the frequency analysis window. Here is an example of the format used by tuners: C4-10. The first letter is the nearest note on the tempered scale. The second note is the octave (4 is that begun by middle C on the piano). The last note is the number of cents the note is above or below the

pitch of the note. In this example the note is -10 cents below C4. Write down in your notebook your pitch measurement. How does the pitch of the note depend on how hard you blow into the flute?

7. Mark the positions of the 6 finger holes. Note the hand position for playing the flute in the picture below. You might want to mark hole #3 and #6 offset from your marked center line, as shown in the diagram above, so that is easier for your fingers to reach the holes.
8. Choose one finger hole to drill last. Drill out the rest of the finger-holes, that is only 5 of them. After you have drilled only 5 of the holes, practice playing your flute. First play the highest note (no fingers down). Then slowly block each hole till you reach the lowest note. If one hole is leaking even a little bit you will not be able to play the lowest notes. Measure the pitches of each note you can play. Did the pitch of the lowest note change after you drilled out 5 of the finger holes? (You should have measured the pitch of the flute prior to drilling any finger holes).
9. Drill out the last finger hole. Again measure the pitches of all the notes. Did the pitches of the notes change after you drilled the last hole?
10. Now you have a complete flute. You can attempt to improve its tuning. If a hole is filed so it is moved up
11. the flute toward the blow hole, the pitch of its note will rise. If a hole is enlarged, the pitch of its note will rise. If the cork is pushed further into the flute, the blow hole enlarged or filed so that it extend further from the cork, the pitch of all notes will rise. If the notes are within a quarter tone of the desired note you are doing great!
12. Convert all your pitch measurements to frequencies in Hz. Note: if you use Adobe/Audition you can directly measure the frequencies by clicking on the fundamental peak in the frequency analysis plot. If you used a tuner you might have recorded C4+15, do the following: You know by appendix C or the table given here that the C4 note is supposed to be 261.63 Hz. You know that you are off by +15 cents. You can calculate:

    15 cents = 1200 Log $_2$[f/261.63] where Log is base 2
    Or 15 cents = 3986 Log $_{10}$[f/261.63] where Log is base 10.

$$(2^{\frac{+15}{1200}}) \times 261.63 = 263.9 \text{ or}$$

$$(10^{\frac{+15}{3986}}) \times 261.63 = 263.9$$

So, the actual frequency of the note is 263.9Hz.

13. Practice playing your flute and see if you can get a clear tone at each note.

Frequencies of Notes in the Tempered Scale 4^{th} octave

Note	Frequency (Hz)
C4	261.63
C# (D♭)4	277.18
D4	293.66
D# (E♭)4	311.13
E4	329.63
F4	349.23
F#(G♭)4	369.99
G4	392.00
G#(A♭)4	415.30
A4	440.00
A#(B♭)4	466.16
B4	493.88

To predict the notes in the octave above, multiply above frequencies by 2. To predict the notes in the octave below, divide the above frequencies by two.

To predict the notes in the octave above, multiply above frequencies by 2. To predict the notes in the octave below, divide the above frequencies by two.

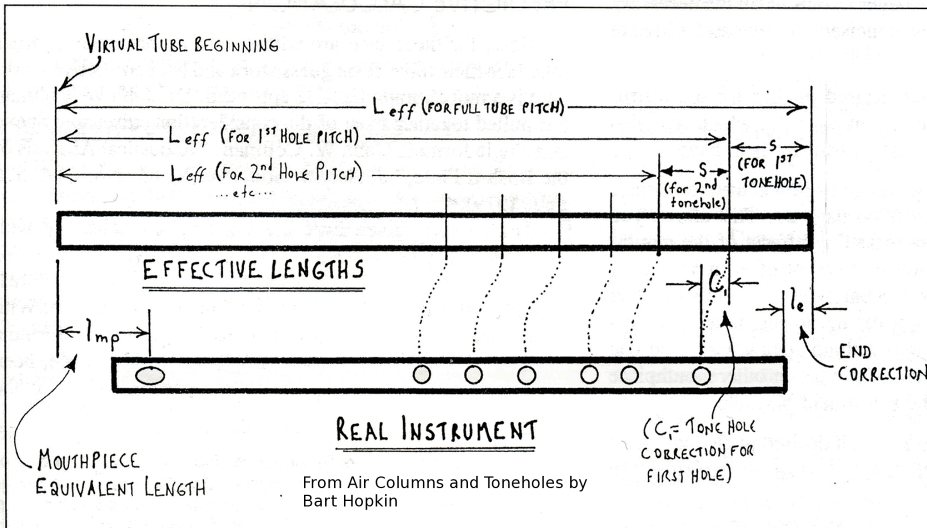
DATA ANALYSIS

1. Use equations (1,2) to determine the theoretical hole placement for your flute. Make a table with the theoretical hole placement (found with the equation) in one column and the actual hole placement (where the holes actually are on your flute) in another column.
2. Make a table listing the theoretical frequencies of the scale of your flute (based on the length), the actual frequencies you obtained for the notes in your scale, and the number of cents you were off for each note. Consider making a plot or graph that shows these measurements.

An example: the pitch of the flute, f, is proportional to $1/L$ where L is the length. So $f = A/L$ where A is a constant of proportionality. Measure the length between the mouth hole and end of the flute (L), and the frequency of the lowest note you can play. Now compute the constant of proportionality, *A*. Then measure lengths between the mouth hole and first open hole for different notes. Use the constant *A* and these other measured lengths to predict the frequencies of other notes. Compare these predicted frequencies to the actual ones you measure when you play these notes. A more sophisticated treatment could also consider the end correction (equations 3,4) or/and equation 5 which computes additional corrections for each tonehole.

A number of things affect the sounding pitch of a note.

- The closer to the mouth piece a hole is, the higher the pitch.
- The larger the hole, the higher the pitch.
- A larger hole in a thicker barrel is similar to a smaller hole in a thinner barrel. The depth of the hole affects the pitch.
- Additional open holes below the first open tonehole will raise the pitch. The smaller the first open tonehole is, the more it will be affected by the open holes below it.
- Closed holes above the first open tonehole can affect the pitch played, however they can either raise or lower the pitch



From Air Columns and Toneholes by Bart Hopkin

FIGURE 14

A REAL FLUTE AS IT RELATES TO AN IMAGINARY TUBE SHOWING THE EFFECTIVE TUBE LENGTHS. The seven vertical lines on the upper tube indicate effective lengths (L_{eff}) corresponding to the six toneholes and the full tube length on the real instrument shown below. Also shown are the end correction, the mouthpiece equivalent length (l_{mp}), the distance *s* between effective lengths, and the tonehole correction C (distance between the effective length and the actual tonehole location) for the first hole.

The flute can be modeled in terms of a virtual flute with effective lengths for the entire thing and corrections for the end, mouth piece and each hole. As explained in Bart Hopkin's book `Air Columns and Toneholes' (see above figure) and referring to J. W. Coltman's paper `Acoustical Analysis of the Boehm Flute', J. Acoust. Soc., 1979, 65, 499-506 an estimate for the tone hole correction is

$$C = \frac{s}{2}\left[\sqrt{1 + 4\left(\frac{t_e}{s}\right)\left(\frac{d_t}{d_h}\right)^2} - 1\right] \qquad \text{(Equation 5)}$$

where

C	tone hole correction
d_t	internal diameter of the tube near the tonehole
d_h	diameter of the tonehole
t_e	effective thickness of the tonehole, $t_e = t + 0.75\, d_h$
t	wall thickness
s	the distance between the effective length (L_{eff}) for the tonehole in question and the effective length of the next hole below or the entire tube in the case of the first open hole

QUESTIONS

1. Is your flute accurate? What would you have done differently to increase the accuracy if you were to make a better flute?
2. Are there trends in the differences between predicted and actual frequencies? Do your measured frequencies tend to be higher or lower than the predicted values. Are the high notes further off in pitch than the low notes?
3. Equation 1,2 given here provide pretty accurate predictions. However differences between predicted and measured pitches, even though they are small, can still be intolerable, particularly to musicians. Comment on the required frequency accuracy of a musical instrument and whether the frequencies predicted in this lab are sufficiently accurate to create a good instrument.
4. What was the most difficult part of constructing the flute? Can you think of ways to make construction easier?
5. You may have measured the pitch of the flute prior to drilling any finger holes. Compare this pitch to that you measured after you drilled 5 or 6 finger holes but are playing the lowest note. Did the pitch change significantly?
6. You may have measured the pitches of some notes when 5 finger holes were drilled and then again when 6 finger holes were drilled? Did the pitches of any of the notes change? Which notes changed the most?
7. If you have time or interest consider using equations 3-5 or checking to see if they better match your measurements.

Physics of Music

Lab 8 – Resonant Tones in a Column of Air – The Didgeridu

EQUIPMENT

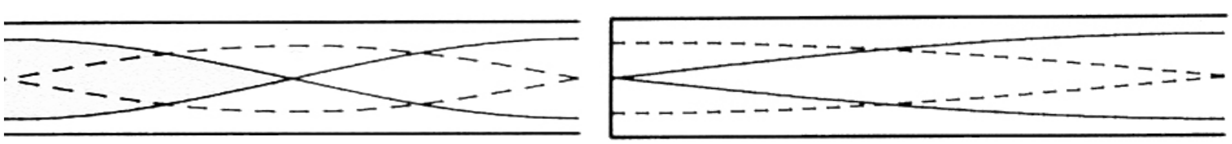
- Skill saws x5
- Heat guns x5. Heat strip. Leather gloves.
- Glue guns
- PVC 1 ½” OD in 10 ft lengths. Enough for each group to make a didgeridoo.
- Hose clamps or rings (large for cone ends, can extend to ~5”) and 2” for mouthpieces
- Mouthpiece: Beeswax or/and 1 ¼” to 1 ½” flexible PVC drain connectors.
- PC and software and microphones
- Sine wave generators and open speakers, banana plugs
- Measuring tapes
- Masking tape
- Antiseptic Mouthwash
- Plastic cutters, plastic glues
- PVC sheet for making bells
- Files, flat screw drivers

American Valve gray thin PVC drain connectors that I found at Lowes worked great. The regular black kind (much easier to mail order) were thicker and I like them less but they were okay after trimming and/or squeezing them smaller.
Materials: one didgeridoo per lab group

Health warning. PVC is inert and safe except when heated to high temperatures. The C is for chloride. PVC fumes are poisonous. Be careful not to overheat the PVC with the heat guns.

INTRODUCTION

All wind instruments operate on the same basic principle. They all have a method of creating vibrations. For example, a French horn player buzzes her lips and a clarinetist makes a wooden reed vibrate. All wind instruments have a characteristic length that determines what notes or vibrations can be produced by the instrument. Finally, they often have some type of device to project the sound that they produce. It is the second item in this list that we are concerned with here – the characteristic length. Only certain resonant frequencies can be produced in a tube of a particular length. Understanding the boundary conditions of a column of air will determine these frequencies.



Above are shown the fundamental modes of a column of air with two open ends (left) and one open, open closed end (right). We can calculate the allowable frequencies of a column of air by using the formulas:

$$f_n = n\frac{v}{2L} \text{ (open-open)} \quad \text{or} \quad f_n = (2n-1)\frac{v}{4L} \text{ (closed-open)} \qquad \text{(Equations 1)}$$

In the above formulas, f is the frequency, v is the speed of sound in air, and L is the tube length. The allowed frequencies are known as harmonics of the tube. The velocity depends on the temperature in the room (approximately 331.4 m/s).

In this lab will construct a digeridu. We will measure its fundamental mode frequency. We will use broad band noise to measure the frequencies of many modes all at once. We will compare the fundamental frequency of a pipe of the same length but a flared end to one without a flared end. We will look at the spectrum of the played instrument.

Notes on a making a digi with PVC: Flared digis are reported to be both easier to play and louder than non-flared digis. The flare provides better coupling with open air allowing more power to be radiated by the instrument. Horn instruments are flared for the same reason. The frequencies of resonance modes in the digi is affected by its shape, so a flared digi sounds different than a cylindrical one. Flaring PVC presents a challenge for us in the lab. PVC when heated becomes soft and stretchy, however while you can swallow ground PVC and it won't really hurt you, **PVC fumes are poisonous**. So please be careful when using the heatguns on the PVC as our lab is not well ventilated. Most instructions for making PVC digis involve stretching the end of the pipe which requires a lot of heat and much energetic stretching (and I didn't have much success with this).

Natural digis made from termite eaten eucalyptus branches have a rich texture on the inside, consequently some digi makers add imperfections (dings and bumps and bends) to the PVC. I am not sure if this adds to the tone quality (might make a good experiment or project to find out).

PURPOSE

The purpose of this lab is to examine the resonant frequencies of a tube with two open ends and one with one open and one closed end. We will also build didgeridoos out of PVC.

PROCEDURE

Part I – Measuring the resonant frequencies of a pipe using a sine wave generator and an open speaker

1. Cut an approximately 4' to 5' ft length of 1 ½" diameter PVC. For example, I have a plastic digi that plays and E at 4' 5". If you make it too long the low note of the digi will be harder to play! If you make it very short then you won't get a deep sound. Place the open speaker up against one open end. Place the microphone outside the other end. Start recording onto the computer, keeping Adobe Audition in Waveform view. Connect the sine wave generator to the open speaker. Turn on the sine wave generator and set the frequency to something low like 50 Hz . Now slowly advance the frequency while watching the recording in waveform view. You should see the amplitude of the recorded sound increase as you approach resonances in the tube. You may need to adjust the preamp or speaker volume so that the recorded sound is not clipped during or too faint during resonances. Find a volume peak and measure its frequency. This would be a frequency of a resonance in the tube. Advance the sine wave generator in frequency and find another resonance. Measure its frequency. You can do this between 1 and 800 Hz. The sine wave generator will not generate frequencies past 800 Hz.
2. Do the frequencies of your measured resonances agree with equation 1 for the open/open pipe or the open/closed pipe? You will need to know the length of your pipe in meters and look up the speed of sound.

Part II – Measuring the resonant frequencies of a pipe using a noise source

3. Now we will measure tbe resonant frequencies of the tube all at once using a noise source. We will create a noise sound file, play it using other software on the computer while simultaneously recording with Adobe Audtion. First we need to create the noise sound file. In Adobe Audtion create a new sound file. Use the Generate function to generate 10 seconds of noise. Check the spectrum of the noise and confirm that it is flat or broad band; in other words no strong single frequencies stands out and there is approximately equal power at all frequencies.
4. Save the noise file as an mp3. Play it using software on the computer other than Adobe Audition; I found Microsoft media player worked fine. Set media player so the sound repeats over and over.
5. Move one of the smaller computer speakers so that it is facing an open end of the pipe. Record with the microphone outside the other end of the pipe. Take a look at the recorded spectrum (shift to spectral view). Is the spectrum consistent with your measurements in 1 and equation 1? Note the small speakers connected to the computer cut off at about 100Hz (that's why there are woofers). This makes it difficult to measure the lowest resonant frequencies of the pipe using a noise excitation and the small speakers. However you should be able to see higher frequency modes in the spectrum.

Part III – Making a didgeridoo – the flared end.

We have two different ways to flare the end.

A. Cutting slats in the end of the pipe with a skill saw. The ends are taped to a fixed ring. The flare is fixed using the heat gun. The result is a soft somewhat floppy end but a reasonable sounding and loud instrument. Using slats we can make a nice bell shape, however I have noticed that musicians tend to prefer more smoothly tapered digis.

B. **Experimental:** Making a cone shaped piece of sheet PVC plastic and gluing it to the end of the digi.

1. **Making a Flared end with slats:** Mark a 1ft length at the end of the pipe. Mark 6 lines along the pipe (parallel to the pipe, see the figure below) evenly spaced in angle along the pipe. Use the skill saw to saw slits along the 6 lines. I find best results using a blade with many small teeth. Place a glass bottle in the end and use the heat gun to soften the PVC so the slats flare. Tape a metal ring or a hose clamp to the end so the slats are held fixed. Set the outer ring diameter at 4" or so. Tape paper around the flared end so the end of the pipe is effectively flared. The heat gun is used to set the PVC slats in the flare rather than allow you to stretch the PVC (much less heat is required).

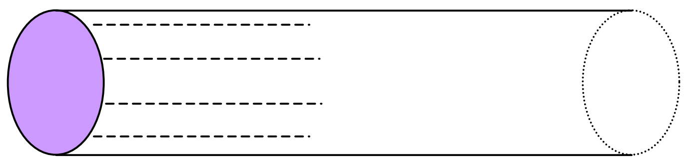


Figure 1. Showing how to cut slats in the end of the PVC pipe if you are using slats to flare the end.

2. **Making a flared end by constructing a plastic cone.**
Cut out the design shown in Figure 2 from a plastic sheet of gray 1/8" PVC.

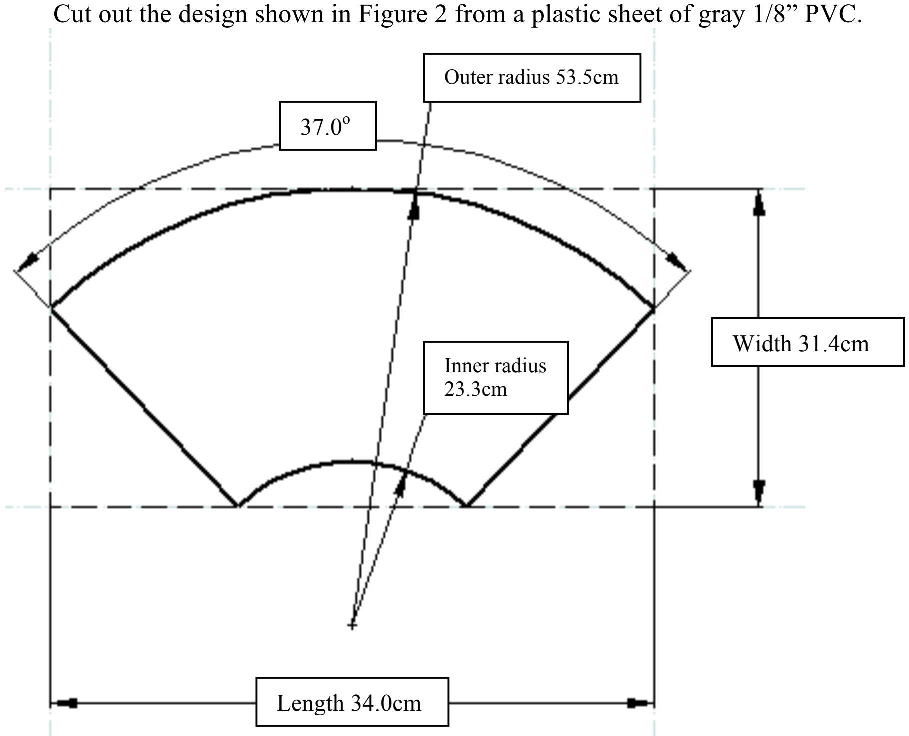


Figure 2. A template for a 30cm long cone with inner and outer radii of 5.5 and 2.4cm.

I learned how to make plastic cones from the following helpful web site: http://www.polyfab.biz/plastic_cones_fabricate.htm They have a nifty .xls calculator that helps you design a cone of any dimensions.

Cutting plastic: To cut a straight edge in very thin (say 1/16 ") plastic: score the plastic along the cutting line with a sharp knife using a ruler. Clamp to table along cutting line. Sharply bend the plastic until it breaks along the line. To cut a curved edge or a straight edge in thicker plastic: clamp to table top. Use skill saw with a blade with many small teeth. File edge smooth.
Bending plastic: Use heat until it softens then slowly bend it. We have two ways to heat the plastic. You can use a heat gun or you can use the long linear heat strip. It takes a couple of minutes for the plastic to soften. Hold until cool to hold shape. Bending this into a cone will take a while. PVC has a lower working temperature than Plexiglas.
Gluing plastic: Primer and all purpose cement. To use glue surface must touch. If the surfaces don't touch you can use a glue gun.

IV – Making a didgeridu – the mouthpiece.

We have three different proposed ways to make the mouthpiece.

A. You can stick a 1 ½" to 1 ¼" flexible PVC drain connector onto the end and blow (this is the easiest!).
B. You can mold a mouthpiece with beeswax . This is the most artistic and adjustable method and beeswax smells nice.
C. You can heat the end of the pipe while tightening a hose clamp on the end. Saw off the pipe at the thinnest point and file the edge smooth. Traditional wood digis have hard mouthpieces so this could be the most "authentic."

1. **The mouthpiece – using a 1 ½" to 1 ¼" flexible PVC drain connector.** File the end of the PVC pipe so that it's smooth. Push the drain connector onto the end. I found the American Valve (thin, gray) drain connectors worked well without any adjustment. However the regular kind (thicker and black) of drain adapter seemed too large for me. I improved these by trimming the end with a knife and by squeezing the end to making it smaller. To shrink the end I squeezed it thinner with a hose clamp placed on the very end while heating it with a heat gun.
2. **The mouthpiece – using beewax:** Soften beeswax with the heat gun until it is malleable. This will take about 5 minutes. If you heat the beeswax so that it is warm but not melted it will be easy to deform. Roll/squeeze out a strip and apply to the non-flared end of the tube. Keep the mouthpiece from touching dirty surfaces after you have molded it. In particular you might want to keep all those PVC shavings from sticking to the end of your instrument!

Figure 3. An example of a beeswax mouthpiece. Note: it does not stick out past the outer diameter edge of the pipe, though it does extend inside the inner rim of the tube.

3. **The mouthpiece – using a hose clamp.** Tighten a 1” to 2” hose clamp to the end of the pipe. Heat the end of the pipe with a heat gun. As you heat it up slowly tighten the host clamp. Tighten the hose clamp until the end is about 1 ¼” diameter. Let it cool then remove the hose clamp. Saw the pipe at the thinnest point. File the end until it is smooth.

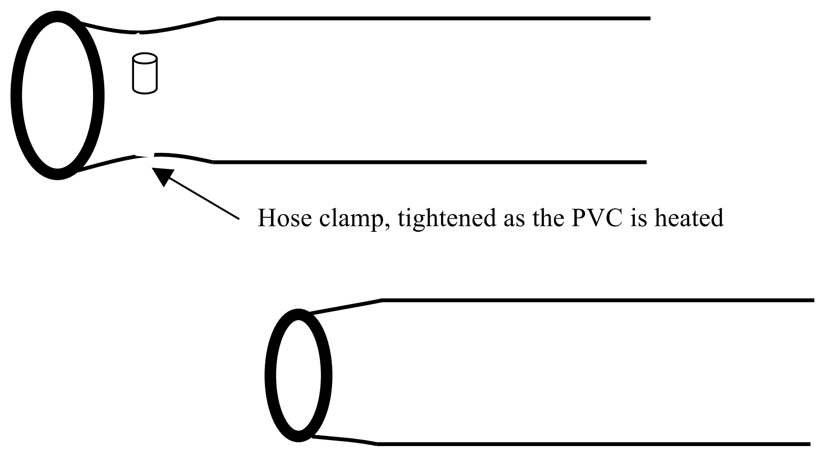


Figure 4. Showing how to construct a mouthpiece by squeezing the end of the PVC pipe.

Part V – Recording the digi!

1. Practice playing the digi (see hints on how to blow below). Use mouthwash to clean the end if you are sharing.
2. Record yourself playing the digi. Look at the spectrum of a played digi. Measure the frequency of the lowest overtone. Is this frequency what you expect based on that you measured for a cylindrical open pipe? (If your digi is longer or shorter than your original pipe then scale by length).
3. What are the frequencies of the overtones? Can you change the strengths and frequencies of these overtones by blowing in different ways? While the speaker drives a sine wave, when you blow into a digi the excitation might be described as something more similar to a triangle wave or something that contains high frequencies that are integer multiples of the fundamental.

How to play the didgeridoo (hints from the digistore!): The lip vibration is similar to giving someone a "raspberry". It can help to stick your bottom lip out a little more than the top lip. To improve the tonal quality of the drone it is important to try to tighten your lips a little after the drone is started. This will increase the pitch and really get the didgeridoo going! If you tighten up too much the drone will abruptly stop and you get a sound we call the "Blow Out". The secret to a good drone is starting loose and tightening up the lips until you almost Blow Out. If you ride the fine line of playing tightly with almost doing a "Blow Out" you can achieve a loud and intense drone. Check out the web site below for ideas on ways to vary your sounds.

http://www.didgeridoostore.com/soundsrhythms.html

DATA ANALYSIS

1. Record the frequencies of the resonant modes for the cylindrical pipe measured using the sine wave generator and broad band noise. Record the length of pipe so you can use equation 1. Record the frequency of the lowest mode for the flared pipe measured using the sine wave generator and open speaker.
2. Make a record of the spectrum of your played digi! Record the frequencies of the overtones in your played digi.
3. Use equation 1 to try and figure out whether the pipe is behaving like an open/open pipe or an open/closed pipe for the played digi and for the pipe excited by a speaker. (Hint: the open/closed pipe has a fundamental frequency that is ½ that of the open/open pipe. Is the fundamental frequency of the blown digi near that or near ½ that of the open/open pipe?)
4. Compare the frequency of the lowest mode of the cylindrical pipe to the flared pipe. Use your intuition gained on the edge effect when we made flutes to account for the direction of the frequency shift.

QUESTIONS

1. Did your measurements of resonant frequencies agree with Equation 1? For the open/open pipe or the open/closed pipe? Does this make sense? The speaker membrane moves air but does not make pressure variations so it should act like an open end.
2. How many resonant modes did you measure using a noise source. Why does broad band noise allow you to measure all the resonant modes at once? Do the frequencies you measure agree with Equation 1?
3. Did you measure a difference in the frequency of the low resonant mode of the flared pipe compared to the cylindrical pipe. Lower or higher?
4. Is the frequency of the lowest mode of the played digi the same as you measured with the speaker in the flared pipe? Consider the difference between excitation caused by your lips and that caused by the speaker. It's possible that the pipe acts like an open/closed pipe when blown but like an open/open pipe when excited by the speaker.
5. Are the overtones of the played digi related to the higher order resonant modes of the pipe or because the excitation (your lips) contains power at higher frequencies that are multiples of the lowest tone?
6. Did you succeed in making a sound with your digi? How could we make them better next year?

Physics of Music
Lab 9 – Room Acoustics

EQUIPMENT

- Tape measures
- Noise making devices (pieces of wood for clappers).
- Microphones, stands, preamps connected to computers running Audition.
- Extra microphone cables so the microphones can reach the padded closet and hallway.
- Extension cords.
- Key to the infamous *padded* closet

INTRODUCTION

One important application of the study of sound is in the area of acoustics. The complete acoustical analysis of a room is both difficult and time-consuming. The benefits of such an analysis, however, are tremendous for lecture halls, auditoriums, and libraries where the sound level is an important feature of the room. What we plan to do in this lab is give you an introduction to room acoustics by listening and recording the way sound travels in rooms. There are three rooms we can easily study near the lab: the lab itself, the “anechoic” chamber (i.e. padded closet across the hall, B+L417C, that isn’t actually very anechoic) and the hallway (which really carries echoes). Anechoic means no echoes. An anechoic chamber is a room built specifically for the purpose of absorbing sound. Such a room should be considerably quieter than a normal room. Step into the padded closet and snap your fingers and speak a few words. The sound should be muffled. For those of us living in Rochester this will not be a new sensation as freshly fallen snow absorbs sound very well. If you close your eyes you could almost imagine that you are outside in the snow (except for the warmth, and bizarre smell in there).

The reverberant sound in an auditorium dies away with time as the sound energy is absorbed by multiple interactions with the surfaces of the room. In a more reflective room, it will take longer for the sound to die away and the room is said to be 'live'. In a very absorbent room, the sound will die away quickly and the room will be described as acoustically 'dead'. But the time for reverberation to completely die away will depend upon how loud the sound was to begin with, and will also depend upon the acuity of the hearing of the observer. In order to provide a reproducible parameter, a standard reverberation time has been defined as the time for the sound to die away to a level 60 decibels below its original level. The reverberation time, RT_{60}, is the time to drop 60 dB below the original level of the sound. The reverberation time can be measured using a sharp loud impulsive sound such as a gunshot, balloon popping or a clap.

Why use 60dB to measure the reverberation time? The reverberation time is perceived as the time for the sound to die away after the sound source ceases, but that depends upon the intensity of the sound. To have a parameter to characterize a room that is independent of the intensity of the test sound, it is necessary to define a standard reverberation time in terms of the drop in intensity from the original level, i.e., to define it in terms of relative intensity. The choice of the relative intensity drop to use is arbitrary, but there is a rationale for using 60 dB since the loudest crescendo for most orchestral music is about 100 dB and a typical room background level for a good music-making area is about 40 dB. Thus the standard reverberation time is seen to be about the time for the loudest crescendo of the orchestra to die away to the level of the room background. The 60 dB range is about the range of dynamic levels for orchestral music.

What is a good reverberation time for a room? If you are using the room for lectures (speech) then a long reverberation time makes it difficult for the audience to understand words. However long reverberation times are desirable in churches for organ music. Reflective surfaces lengthen the reverberation time whereas absorption surfaces shorten it. A larger room usually has a longer reverberation time because it takes longer for the sound to travel between reflections. Rooms that are good for both speech and music typically have reverberation times between 1.5 and 2 seconds. The reverberation time is influenced by the absorption coefficients of the surfaces in a room, but it also depends upon the volume of the room. A small room would not have a long reverberation time.

Predicting the reverberation time and Sabine's formula.

Sabine is credited with modeling the reverberation time with the simple relationship which is called the Sabine formula:

$$RT_{60} = (0.16\,\text{sec/ m})\frac{V}{S_e} = (0.049\,\text{sec/ ft})\frac{V}{S_e} \qquad \text{(Equation 1)}$$

This formula relates the reverberation time, RT_{60}, to room volume and an effective area. You use the 0.16 sec/m coefficient if you are working in meters. You use the 0.049 sec/foot coefficient if you are working in ft. Here V is the volume of the room and S_e is an effective area. The effective area is calculated as follows

$$S_e = a_1 S_1 + a_2 S_2 + a_3 S_3 + \ldots$$

Here each area S_i has an absorption coefficient a_i. The effective area is a sum of areas, S_i , each with its own absorption coefficient a_i. These areas are the surfaces in the room (ceiling, walls, floor, people, etc…). Another way to write the effective area is with a sum

$$S_e = \sum_{\text{surfaces } i} a_i S_i$$

Note you must put the areas (S_i) in the same units as the volume V (meter^2 for area and meter^3 for volume or ft^2 for area and ft^3 for volume). When a sound wave in a room strikes a surface, a certain fraction of it is absorbed, and a certain amount is transmitted into the surface. Both of these amounts are lost from the room, and the fractional loss is characterized by an absorption coefficient, *a,* which can take values between 0 and 1, 1 being a perfect absorber and 0 being a perfect reflector. Absorption coefficients are unitless. The absorption coefficient is the fraction of the power absorbed in one reflection. Absorption coefficients for some common materials are given below. The absorption coefficient of a surface can depend on the frequency of sound used to measure it. For example, carpet is quit absorptive at high frequencies but not at low frequencies. A perfectly absorptive room would have an effective area that is equal to the total surface area of its walls, ceiling and floor. A highly reflective room would have an effective area that is smaller than its total surface area.

The Sabine formula works reasonably well for medium sized auditoriums but is not to be taken as an exact relationship. The Sabine formula neglects air absorption, which can be significant for large auditoriums. It also tends to overestimate the reverberation times for enclosures of high absorption coefficient. A better approximation for such enclosures utilizes an overall average absorption coefficient:

$$RT_{60} = \frac{(0.16\ \text{sec/m})V}{S_e\left(1 + \dfrac{S_e}{2S}\right)} \qquad \text{(equation 2)}$$

where S is the total surface area of the room. Note that this reduces the calculated reverberation time.

Some important acoustical characteristics to consider are:

- Liveness – This just refers to the length of the reverberation time. The longer the reverberation time, the more "live" the room is.
- Intimacy – This refers to how close the performing group sounds to the listener.
- Fullness – This refers to the amount of reflected sound intensity relative to the intensity of the direct sound. The more reflected sound, the more fullness the room will have.
- Clarity – This is the opposite of fullness. In general, greater clarity implies a shorter reverberation time.
- Warmth – This is obtained when the reverberation time for low-frequency sounds is somewhat greater than the reverberation time for high frequencies.
- Brilliance – This is the opposite of warmth.
- Texture – This refers to the time structure of the pattern in which the reflections reach the listener. The first reflection should quickly follow the direct sound.
- Blend – This refers to how well the mixing of sound occurs between all of the instruments playing.
- Ensemble – This refers to the ability of the members of the performing group to hear each other during the performance.

Some problems in acoustical design are:

- Focusing of sound – This refers to the undesirable effect that occurs when sound is much louder at one point in the room than surrounding points in the room.
- Echoes – To obtain good texture, it is desirable to avoid any particularly large single echoes.
- Shadows – These are quiet areas that can be produced when there are large overhanging balconies or other structure jutting into the room.
- Resonances – It is necessary to separate the resonances of a speaker and the box enclosure of a tuned-port system to provide the best sound.
- External noise – This refers to noise coming from outside of the room.
- Double-valued reverberation time – If a recording of a musical event is played back in a room that it was not originally recorded in, this phenomenon can result.

Absorption coefficients for common surfaces:

	Sound Absorption Coefficients at frequency					
Nature of Surface	125	250	500	1000	2000	4000
Acoustic tile, rigid mount	0.2	0.4	0.7	0.8	0.6	0.4
Acoustic tile, suspended	0.5	0.7	0.6	0.7	0.7	0.5
Acoustical plaster	0.1	0.2	0.5	0.6	0.7	0.7
Ordinary plaster, on lath	0.2	0.15	0.1	0.05	0.04	0.05
Gypsum wallboard, 1/2" on studs	0.3	0.1	0.05	0.04	0.07	0.1
Plywood sheet, 1/4" on studs	0.6	0.3	0.1	0.1	0.1	0.1
Concrete block, unpainted	0.4	0.4	0.3	0.3	0.4	0.3
Concrete block, painted	0.1	0.05	0.06	0.07	0.1	0.1
Concrete, poured	0.01	0.01	0.02	0.02	0.02	0.03
Brick	0.03	0.03	0.03	0.04	0.05	0.07
Vinyl tile on concrete	0.02	0.03	0.03	0.03	0.03	0.02
Heavy carpet on concrete	0.02	0.06	0.15	0.4	0.6	0.6
Heavy carpet on felt backing	0.1	0.3	0.4	0.5	0.6	0.7
Platform floor, wooden	0.4	0.3	0.2	0.2	0.15	0.1
Ordinary window glass	0.3	0.2	0.2	0.1	0.07	0.04
Heavy plate glass	0.2	0.06	0.04	0.03	0.02	0.02
Draperies, medium velour	0.07	0.3	0.5	0.7	0.7	0.6
Upholstered seating, unoccupied	0.2	0.4	0.6	0.7	0.6	0.6
Upholstered seating, occupied	0.4	0.6	0.8	0.9	0.9	0.9
Wood seating, unoccupied	0.02	0.03	0.03	0.06	0.06	0.05
Wooden pews, occupied	0.4	0.4	0.7	0.7	0.8	0.7

Data from Hall, 2nd. Ed, Table 15.1

Effective areas for people and seats in auditoriums in sabins

	120Hz	250Hz	500Hz	1000Hz	2000Hz	5000Hz
Audience per person in sabins	0.35	0.43	0.47	0.50	0.55	0.60
Auditorium seat, solid un-upholstered	0.02	0.03	0.03	0.03	0.04	0.04
Auditorium seat, upholstered	0.3	0.3	0.3	0.3	0.35	0.35

A sabin is a unit of acoustic absorption equivalent to the absorption by a square foot of a surface that absorbs all incident sound.

PURPOSE

The purpose of this lab is to explore the acoustics of different rooms. We can compare the characteristics of three rooms: the hallway, the lab (B+L403), and the padded closet (B+L417C) (aka "anechoic chamber" but its really not anechoic). Your lab group should use two of three of these rooms to measure sounds. You will measure the reverberation time of the two rooms and compare your measured value to those estimated using Sabine's formula.

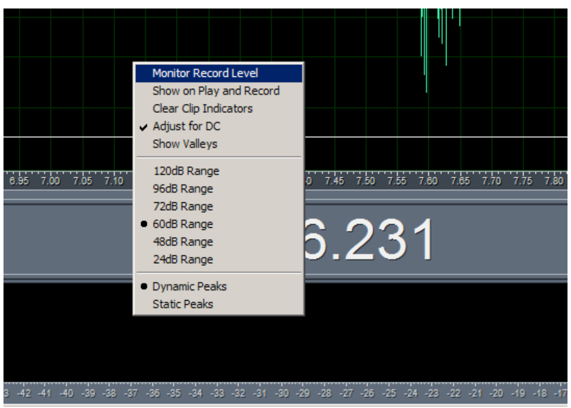
PROCEDURE

A. Listing

1. By listening compare the characteristics of the three rooms: the hallway, the lab (B+L403), and the padded closet (B+L417C). Snap your fingers, carry out a conversation, sing, clap some boards together. Describe the characteristics of these room in terms of the above vocabulary (brilliance, liveness, clarity, warmth, etc..).

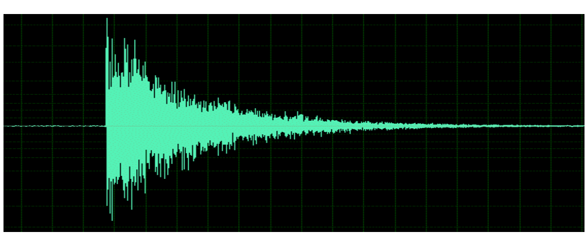
B. Measuring the Reverberation Time

1. Go to one of the two rooms that your lab group has chosen to study.
2. To record a quickly decaying sound you need to adjust the recording level so that the sound is not heavily saturated or clipped. Bring up Adobe Audition on your computer and make sure that the preamp is plugged in. You can look as the sound level by right clicking on the level meter (bottom of screen) and choosing "Monitor Record Level." Make a loud clap with two pieces of wood. Check that the sound is not saturated. Adjust the preamp to a good level. Record a few claps. Check that the sound is not heavily clipped. Play back your sound to hear what it sounds like. You want to hear the room not the clapper so make the claps distant from the microphone (on the other side of the room).



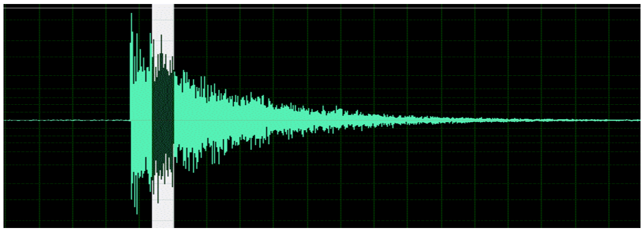
To show the sound level before recording select Monitor Record Level. To get this box right click on the sound level bar at the bottom of the screen. Adjust the preamp knobs so that a clap is well measured (not too faint) but is also not heavily clipped (or in the red zone).

3. Look at the waveform of your recorded claps. Expand the horizontal or time axis using the zoom button at the bottom of the screen (this looks like a little plus sign inside a circle). Take a look at the decay of the sound that you recorded.



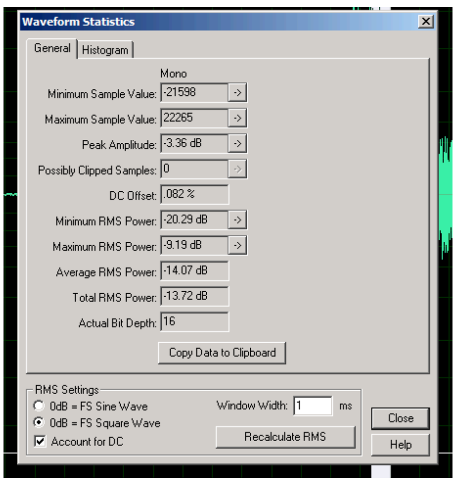
An example of a clap. Each spike is an echo from a different surface in the room. Large flat surfaces produce large strong echoes.

4. Are there any particularly strong echoes in your room? Can you see them in the waveform of your recorded claps?
5. Make sure the time format is set so that you can record the time with precision (to a precession of msecs). Click on "View" at the top of the program, select "Display Time Format" and choose "Decimal".
6. Measuring the loudness at a particular time. With a left click and drag, select a region of your waveform. Write down the start time of your selected region.



A selected region of the waveform.

7. On the top of the screen there are a bunch of options. Choose “Analyze”. Choose “Statistics”. Adjust the “Window width” to something small such as 1ms. Press the button that says “Recalculate RMS.” The window width is the time over which each power measurement is made. You should make the window width small enough that your selected region contains many windows. The Average or RMS power is the average of all the powers measured in each window. RMS stands for root mean square. Remember the power depends on the square of the signal. The RMS power is given in dBFS which is dB below full scale. This arbitrarily sets 0dB to be the largest number that you can measure. The beginning of the clap should be near 0dB (nearly clipped). As the sound decays you should measure lower values of the RMS power (that means negative values of dB).



The Waveform Statistics box. Using this box you can measure the loudness in dB at a particular time.

8. Record the power levels at different times after the onset of the clap. Use the RMS power as a measure of the power. Remember dB is decibel or 10 $\log_{10}$ (Power).
9. ANALYSYS: Plot dB level vs time. You should see points that lie along a line. If you don’t see this consider making your measurements in a different way and redoing your measurements. It may be that the first point is too high – if so consider ignoring it, but it you do explain that you are doing so in your analysis. If your plot has a series of points in a line then the power is decaying exponentially in time. Since dB is the logarithm of the power we expect a line on a plot of dB vs time. Plot a line that lies on top of your plotted data points and specify its slope. The slope of this line is related to the exponential decay rate of the sound and the reverberation time.
10. It is impractical to directly measure RT_{60} or how long it takes to drop 60 dB below the loudest level. This would correspond to a drop of a factor of a million in the power of the noise. What prevents us from measuring the sound level over such a large range? We can’t measure noise fainter than the ambient background noise. Also the analog to digital converter on the sound card in the computer doesn’t have enough bits to cover this large a dynamic range. How are we going to calculate RT_{60} if we can’t measure the sound when it is 60 dB fainter? We can assume that the decay of the sound follows a line when plotted as dB vs time. Using your plot of dB vs time, extend your line and estimate how long it would take for your sound to decay to 60dB below that of the loudest part. This time is then an estimate for the reverberation time or RT_{60}. This technique is called extrapolation.
11. Repeat your measurements for a second room.

C. Predicting the reverberation time using Sabine's formula.

1. Using the various measuring devices, measure the volume of the room. If your room is not rectangular, be sure to make enough measurements so that you are able to calculate the volume. Note: floor tiles in the US tend to be 1ft square.
2. Measure the surface area of all the different absorbing materials in the room. Remember to include yourself if you happen to be in the room! If you have a large number of similar objects, it suffices to measure the surface area of one of them and multiply. For example, if you are measuring the surface area of seats in an auditorium, measure one seat and multiply that value by the number of seats in the auditorium. There is no need to measure the surface area of everything in the room, but all of the larger objects should be measured. Don't forget to measure doorways, windows, and yourselves.
3. Calculate the affective area of your room, A_e, using absorption coefficients listed above or from the appendix of your lab manual. If a material in you room is not found on the chart, make a guess at what the absorption coefficient is based on what material it is most like on the list. For people use the table above that gives affective areas per person in sabins (units ft squared).
4. Predict the reverberation time using Sabine's formula. Predict the reverberation time for two different frequencies.
5. Does your predicted RT_{60} agree with the one you measured? A clap has a fairly flat spectrum (both high frequency and low frequency components). This makes it hard to do this comparison exactly. What other effects might account for any difference between your predicted and measured value for the reverberation time?

Additional Questions

What are the differences in the predicted and measured reverberation times from the two rooms that you studied? Did you measured and predicted reverberation times agree? Why or why not?

Notes:

To remove DC offset that might have been introduced in recording: Select the entire waveform. Choose "Effects". Choose "Filters". Choose "Scientific Filters". Do a High Pass filter with a cutoff of a low frequency like 5Hz. This should remove a DC offset. You can now set the vertical scale to dB and have a sensible zero level.

We always seem to predict reverb times using Sabine's formula that are vastly different than the ones we measure from the decay timescale. I still have not figured out why.

Physics of Music

Lab 10 – Copper Pipe Xylophone and Gongs

EQUIPMENT

- Copper pipes in a scale at ½" diameter
- Different diameter pipes with same lengths
- Mallets
- Weather-strip coated board Stands for the copper pipes
- Tuners
- Microphones, stands, preamps connected to computers running Adobe Audition.
- Band saw
- Jigs for cutting slots in copper pipes
- Pipe cutters
- Wire, wire cutters
- 3/4" diameter copper pipe for gongs (1 foot per gong, enough for one gong per station)

Safety warning: Wear goggles when using power equipment! Make sure others watching are also wearing protective eyewear.

Safety warning: It is possible to *loose* a finger if you let your fingers get near the blade on the band saw. We are using jigs to hold the pipe while cutting the slot in the copper pipe so that our fingers *never* get near the blade. Please remember to keep your hands away from the blade at all times. If you see a colleague using the band-saw unsafely don't just watch hoping that they won't hurt themselves – prevent the injury before it happens. Shut the band saw down and complain loudly until your colleague uses the band-saw safely.

INTRODUCTION

For a guitar string or a column of air, the pitch of the fundamental tone sounded is proportional to the length of the string or length of the column of air. However for other systems the pitch of the fundamental tone may depend on the length in a more complicated manner. For example the pitch of the fundament may depend on the square of the length or the square root of the length. In this lab we will experimentally measure the way that the fundamental tone of copper pipe depends on its length. We can write

$$f \propto L^{\alpha} \qquad \text{(Equation 1)}$$

where f is the frequency of the fundamental tone, L is the length of the pipe and α is a power that will be measured in this lab. The symbol $\propto$ means "is proportional to". For guitar strings and flutes, $\alpha \approx 1$ and the pitch of the fundamental tone is proportional to the length of the string or column of air.

If we take the log of the above equation we find

$$\log f = \text{constant} + \alpha \log L \qquad \text{(Equation 2)}$$

On a plot of $\log f$ vs $\log L$ the exponent α would be the slope of a line.

For a guitar string or a column of air, the overtones are integer multiples of the fundamental tone. However there are vibrating systems where the overtones are not integer multiples. This contributes to their timbre. Bells, drums and copper pipes are examples of instruments that have a complex spectrum of overtones. In this lab you will measure the frequencies of these overtones, f_n and their ratios to that of the fundamental or f_n/f_1. Here f_n refers to the n-th partial or overtone.

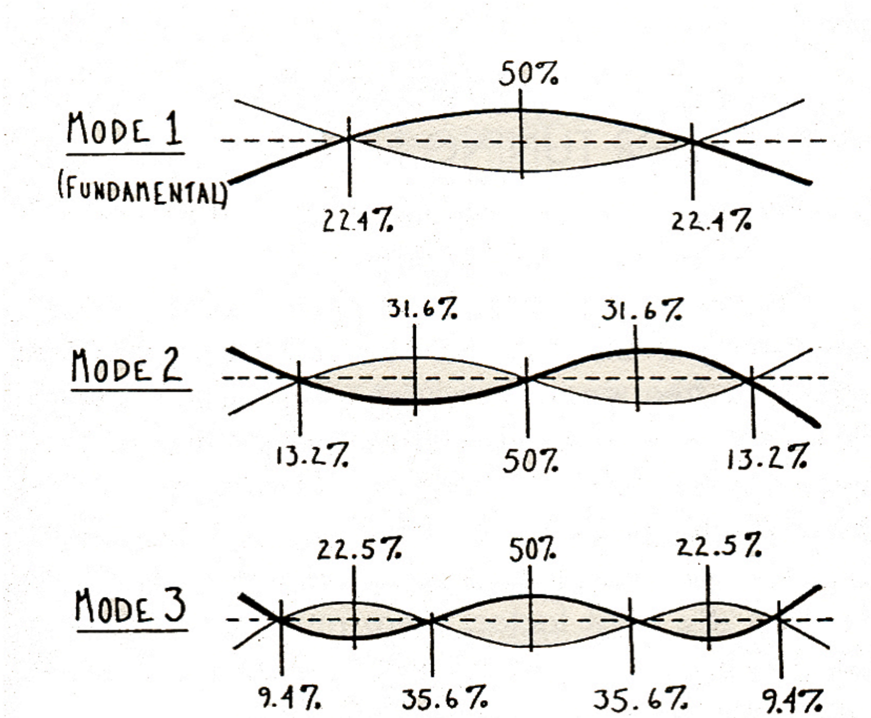


Figure 1. This figure shows motions for first three modes of a steel bar. The steel bar is not help fixed at either end. From Hopkins.

The modes excited in a copper pipe depend on the speed of sound in the pipe and the stiffness of the pipe. A different diameter pipe should have a different stiffness (harder or easier to bend) and so should have different frequencies of vibration. A copper pipe has bending modes similar to those in a steel bar shown above.

In this lab we will measure the frequencies of the fundamental bending mode for copper pipes of different lengths and determine experimentally how the fundamental model frequency depends on pipe length. Specifically we will measure the exponent α in Equation 1 or 2 above.

When a copper pipe is hit it moves with bending modes (shown above) that have frequency that depend on pipe length. However the pipe can also deform in other ways. For example two sides of the pipe could approach each other while the opposite sides move away (see Figure 2) . The frequencies of these modes would not depend on pipe length, though they would be sensitive to pipe thickness and diameter. In this lab we will look at the spectrum of a copper pipe to see if we can find mode frequencies that don't depend on pipe length.

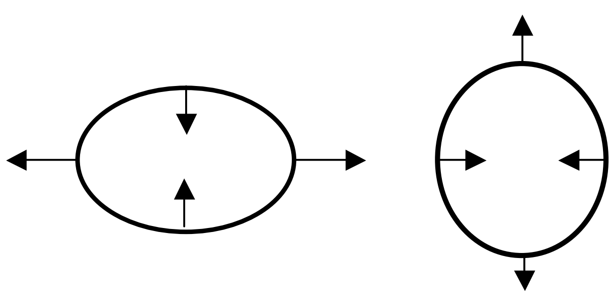


Figure 2. Possible vibration modes looking down the end of a copper pipe. This type of motion could correspond to a mode with frequency that does not depend on pipe length.

If a slot is cut in the end of the pipe then the ends of the pipe an also move away from each other, in a way similar to a tuning fork (see Figure 3).

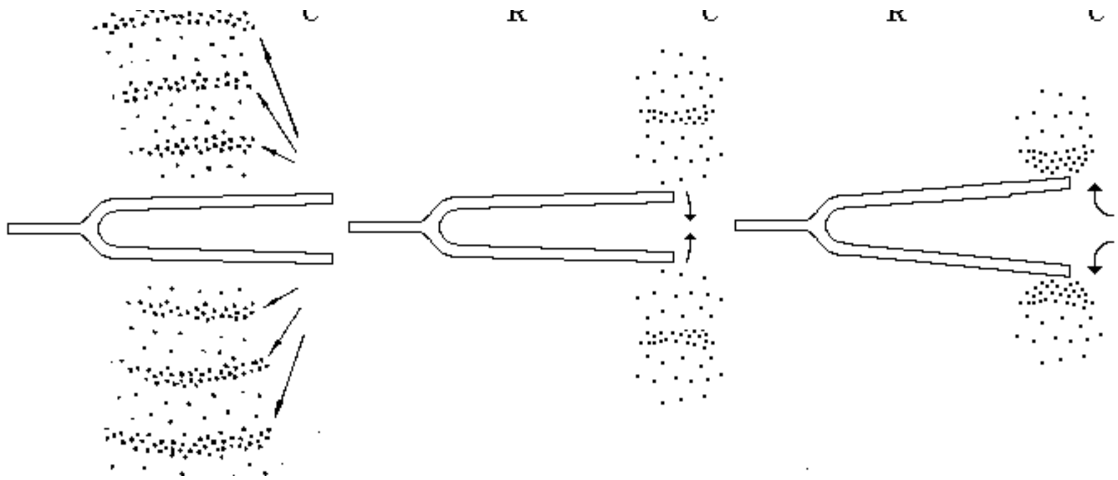


Figure 3. This figure shows motions for a tuning fork. We expect a lower fundamental mode frequency if the fork prongs are longer.

A copper pipe with a slit cut in the end has many possible modes of vibration leading to a rich spectrum and possibly a pleasing sound. In this lab we will look at how the spectrum of a copper pipe changes as a slit is cut into its end.

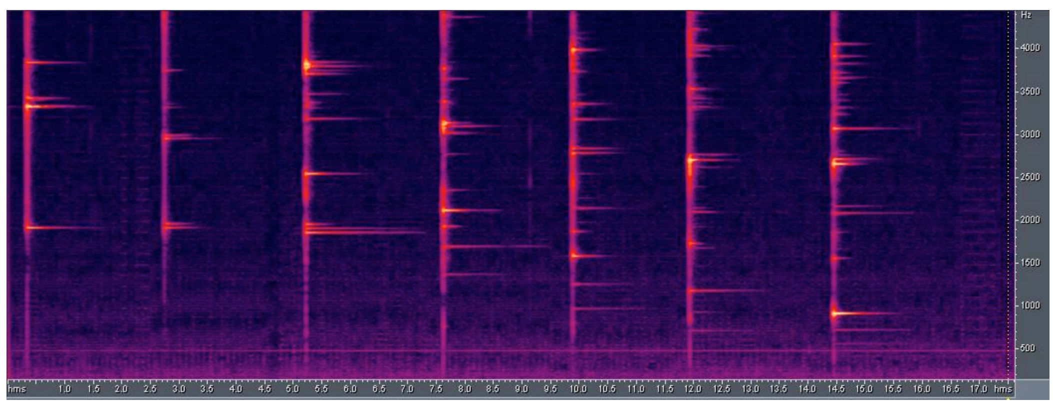


Figure 4. Spectra of a ¾" diameter, 9" long copper pipe. On the left no slit has been cut. From the left to the right each spectrum corresponds to the pipe with a 1cm longer slit. The last spectrum with the 6cm slit had a nice sound. Perhaps two modes coincide and the lowest mode is particularly strong as a consequence. Because I liked the sound I stopped extending the slit.

PROCEDURE

A. Pitch as a function of length.

1. Using the ½" pipes at different lengths set up as a xylophone playing a scale. Hit the pipes and measure the frequencies of their lowest tones.
2. What notes are played?
3. Measure the lengths of the pipes.
4. What is the relation between pipe length and pitch? Try making the following plots: Plot pipe length vs pitch for the 8 pipes. Plot log pipe length vs log pitch for the 8 pipes. On which of these plots do the points lie on a line?
5. Using equation 1 predict for different values of α the relation between length and frequency you would expect on each of your plots. On one of these plots α is related to the slope of a line. By measuring the slope measure α .
6. Does the frequency depend linearly on the length of the pipe? What is your best estimate for α in equation 1? Plot lines on top of your data points. The line that best goes through your data points should determine your best estimate (measurement) for α .

B. Pitch as a function of pipe diameter and material

1. Measure the fundamental frequencies for two pipes of the same material (copper) but different diameters.
2. Suggest a relationship between pipe diameter and fundamental frequency.
3. Measure the fundamental frequency for two pipes of the same length and approximately same diameter but different material (steel and copper). Are the frequencies higher or lower for a stronger material?

C. Structure of Overtones

1. Record the sounds from the pipes. Measure the frequencies of the overtones. Are they integer multiples of the fundamental? Compute the ratios of the overtones to the fundamental.
2. The sound of the pipe may depend upon where you hold the pipe or where you hit it. Record while you hit the pipe in different locations or hold the pipe at different locations. Does the frequency spectrum change?
3. Measure the overtones for the largest and smallest length ½" pipes. Compute the ratios of the overtones divided by the fundamental. Are these ratios the same for the two different diameter pipes?
4. Record while you play a scale on the copper pipe lengths. Which overtones depend on pipe length? You may need to look at overtones up to 5000 Hz or so to find some that don't vary with pipe length.

D. Making a Slotted Gong

Safety warning: If you are using power tools wear goggles. Check that anybody nearby and watching is also wearing protective eyewear.

1. Cut a 10" or so length of 3/4" copper pipe using a pipe cutter. To use a pipe cutter: Tighten it until it lightly grips the pipe. Swing the cutter around the pipe a couple of times. Tighten the cutter a little bit more. Repeat the last two steps until the pipe breaks.
2. Drill a hole through the pipe on one end. Make the hole wide enough (7/64") that an 8 penny finishing nail will fit through it. The nail will hold the pipe at a fixed orientation in the jigs we use when cutting the slots. The hole should go through the middle of the pipe. Cut a piece of wire to hold the gong. Hit the pipe with the mallet while holding the wire. Record the sound. Save the sound file so you don't lose the spectrum!

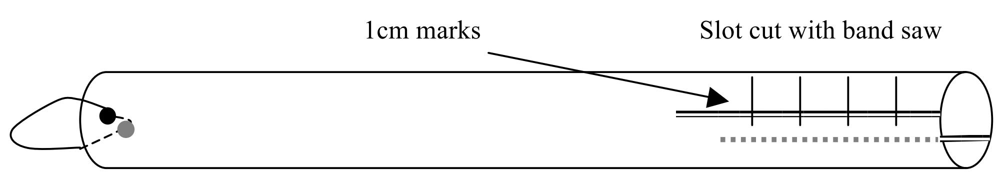


Figure 5. Slotted gong made from a copper pipe. When struck both bending modes and tuning fork modes are excited.

Safety warning: It is possible to *loose* a finger if you let your fingers near the blade on the band saw. We are using jigs to hold the pipe while cutting the slot in the copper pipe so that our fingers **never** get near the blade. Please remember to keep your hands away from the blade at all times. If you see a colleague using the band-saw unsafely don't just watch hoping that they won't hurt themselves – prevent the injury before it happens. Shut the band saw down and be obnoxious until your colleague uses the band-saw safely.

3. Mark 1cm lengths on one side of the pipe so you can remember which hole goes up in the jig and so you can cut to particular slot lengths. I have made jigs to guide the band saw blade so that a perpendicular cut can be reliably made in the pipe. The jigs consist of wood with slots already cut in them and holders for the pipes. They also have a role of holes. When using the jig, put a nail through the hole in the end of

the pipe to hold it fixed so that each time you extend the slot the band saw will cut along the same path. The slot in the wood will guide the blade of the band saw so that it will remain at the same angle**.**

4. **C**ut a ~1cm slot in the end of the pipe using the jigs we have made for this purpose. Record the sound again. Save the sound file. Repeat a few times. Has the sound of the gong changed? How has the spectrum changed?
5. R**e**peat the last step until the slot is about ¼ to 1/3 of the length of the pipe or/and you like the sound of the gong.

DISCUSSION

1. For the string and wind instruments frequency is proportional to length. Did you find that fundamental frequency was proportional to pipe length? Why might you see a non-linear relationship between fundamental frequency and length?
2. How well were you able to measure α ? How far off could your measurement be?
3. Discuss the relation between the fundamental pitch and pipe length, the fundamental pitch and pipe diameter, thickness and the fundamental pitch and pipe material. We might expect stiffer pipes to have higher frequency fundamentals.
4. Discuss the differences you see in the harmonics for the different pipes. Did all overtones depend on pipe length? What types of motions might depend on pipe length?
5. Why does the gong make lower sound when you put a slit in the copper pipe? Did you see a pattern in the changes in the spectrum as you extended the slot?

Appendices

Appendix A: Scientific Notation and its Prefixes

SI PREFIXES		
Factor	**Prefix**	**Symbol**
10^{24}	Yotta	Y
10^{21}	zetta	Z
10^{18}	exa	E
10^{15}	peta	P
10^{12}	tera	T
10^{9}	giga	G
10^{6}	mega	M
10^{3}	kilo	k
10^{2}	hecto	h
10^{1}	deka	da
10^{-1}	deci	d
10^{-2}	centi	c
10^{-3}	milli	m
10^{-6}	micro	μ
10^{-9}	nano	n
10^{-12}	pico	p
10^{-15}	femto	f
10^{-18}	atto	a
10^{-21}	zepto	z
10^{-24}	yocto	y

Examples of use:

$89 \text{ ns} = 89 \times 10^{-9} \text{ s} = 8.9 \times 10^{-8} \text{ s}$

$230 \text{ km} = 230 \times 10^{3} \text{ m} = 2.3 \times 10^{5} \text{ m}$

Appendix B: Notes on the Piano

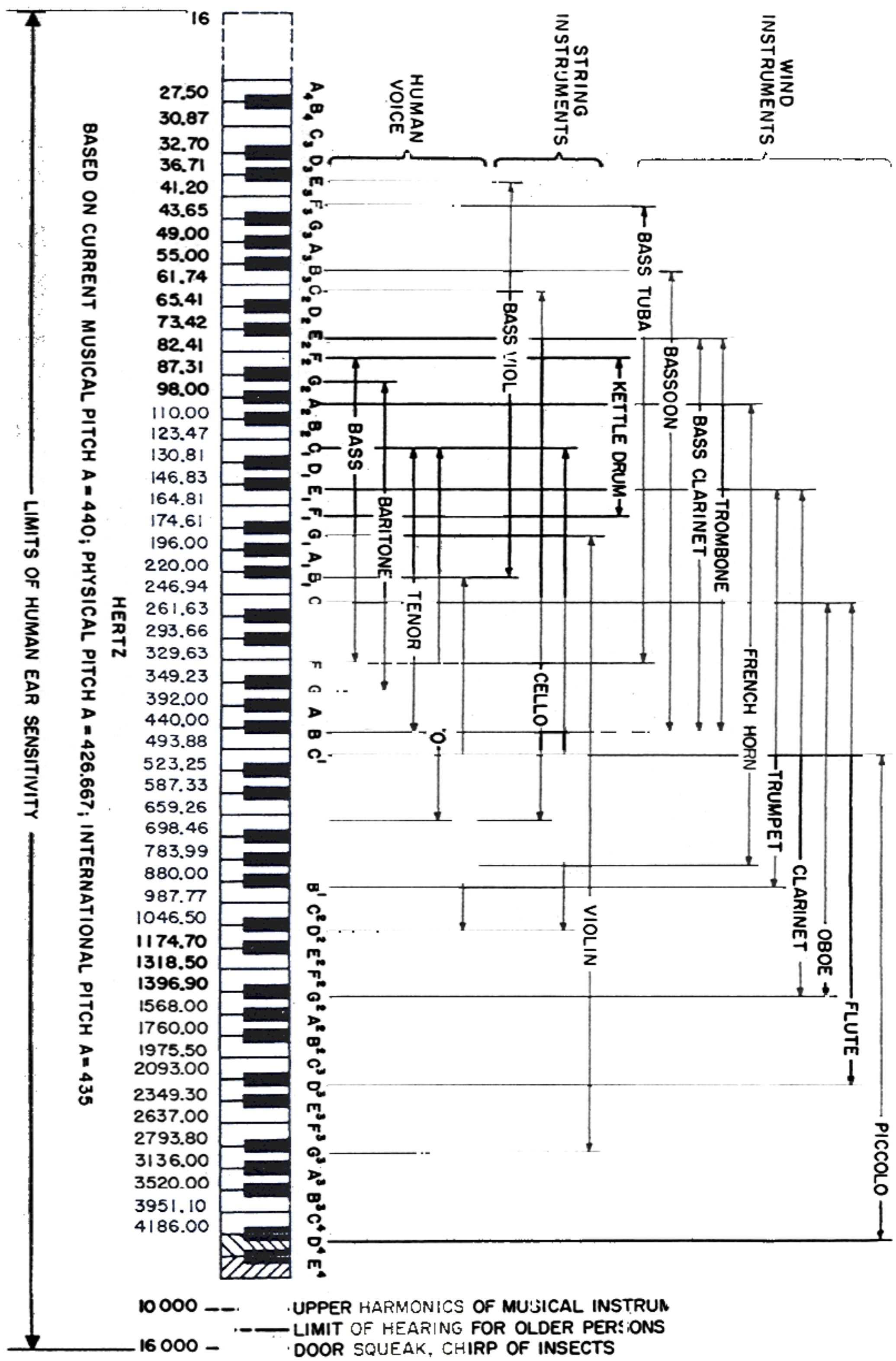


Fig. 18—Acoustic spectrum in hertz.

Appendix C: Equal Tempered Music Scale

TABLE 13.5
Frequencies of Notes in the Equal Tempered Scale P252 J. Rigden

C_0	16.352	C_3	130.81	C_6	1046.5
	17.324		138.59		1108.7
D_0	18.354	D_3	146.83	D_6	1174.7
	19.445		155.56		1244.5
E_0	20.602	E_3	164.81	E_6	1318.5
F_0	21.827	F_3	174.61	F_6	1396.9
	23.125		185.00		1480.0
G_0	24.500	G_3	196.00	G_6	1568.0
	25.957		207.65		1661.2
A_0	27.500	A_3	220.00	A_6	1760.0
	29.135		233.08		1864.7
B_0	30.868	B_3	246.94	B_6	1975.5
C_1	32.703	C_4	261.63	C_7	2093.0
	34.648		277.18		2217.5
D_1	36.708	D_4	293.66	D_7	2349.3
	38.891		311.13		2489.0
E_1	41.203	E_4	329.63	E_7	2637.0
F_1	43.654	F_4	349.23	F_7	2793.8
	46.249		369.99		2960.0
G_1	48.999	G_4	392.00	G_7	3136.0
	51.913		415.30		3322.4
A_1	55.000	A_4	440.00	A_7	3520.0
	58.270		466.16		3729.3
B_1	61.735	B_4	493.88	B_7	3951.1
C_2	65.406	C_5	523.25	C_8	4186.0
	69.296		554.37		4434.9
	73.416	D_5	587.33	D_8	4698.6
	77.782		622.25		4978.0
E_2	82.407	E_5	659.26	E_8	5274.0
F_2	87.307	F_5	698.46	F_8	5587.7
	92.499		739.99		5919.9
	94.999	G_5	783.99	G_8	6271.9
	103.83		830.61		6644.9
	110.00	A_5	880.00	A_8	7040.0
	116.54		932.33		7458.6
B_2	123.47	B_5	987.77	B_5	7902.1

Appendix D: The speed of sound in air

$V \sim 331.4 + 0.6\ Tc$ where V is the speed of sound in m/s and Tc is the air temperature in Celcius.
This is a good approximation at sea level.
At 25 C the speed of sound is approximately 346 m/s.

♭