1. Vibrations of strings

A. Objectives

- Measure the frequencies of the lowest few normal transverse modes of a spring, and determine the functional relationship between these frequencies.
- Measure the velocity of wave propagation of a spring, and compare it to the velocity determined from the normal mode frequencies.
- Measure the waveform and frequency spectrum of a plucked, clamped wire under tension, and observe how they depend on how the string is plucked.
- Measure the dependence of the vibration frequency of a clamped wire with the tension in the wire and the length of its vibrating section. Determine the functional relationships between these quantities.

B. Equipment required

1. Spring
2. Timer
3. Meter stick
4. Sonometer
5. Slotted weights and weight hanger
6. Sound sensor
7. Computer data acquisition system

C. Introduction

Fourier Transform

Consider the following function of time $V(t)$:

$$\begin{aligned} V(t) = &-1.1\cos(2\pi(50)t) + 0.95\cos(2\pi(110)t) + 2.0\cos(2\pi(170)t) \\ &-1.6\cos(2\pi(260)t) + 3.4\cos(2\pi(370)t) \end{aligned} \tag{1.1}$$

This function is plotted in Figure 1.1. It is a sum of five oscillating cosine functions $\tilde{V}_n \cos(2\pi f_n t)$, with values of $\tilde{V}_n$ and f_n that are different in each of the five terms. We label each term with an index *n*. $\tilde{V}_n$ is the *amplitude* of the *n*th term, and f_n is the *frequency* of the *n*th term.

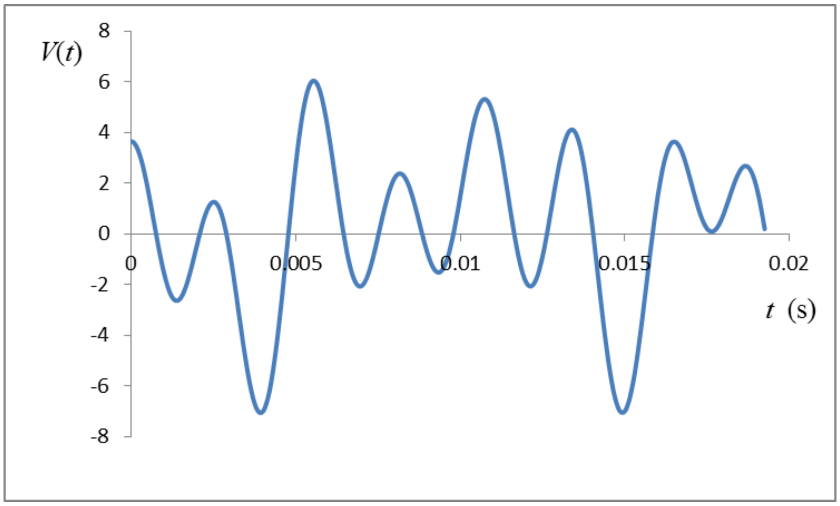

Figure 1.1. A function $V(t)$ that is the sum of five oscillating cosine functions of different frequencies.

We'll always assume that time t is measured in seconds. I'm writing formulas here with frequencies measured in Hz (cycles per second), since this is a laboratory course, and frequencies in the lab are normally measured in Hz. Frequencies can also be measured as *angular frequencies* (radians per second). Many physicists follow the convention that a frequency in Hz is represented by the symbol f, and an angular frequency is represented by the symbol ω. You can get formulas written with angular frequencies with the substitution $\omega = 2\pi f$, since there are 2π radians in a cycle.

If we know that $V(t)$ is a sum of such cosine terms, we can specify $V(t)$ just by giving the pairs of numbers $\left(f_n, \tilde{V}_n\right)$ in a table. Or, we can show $\tilde{V}_n$ *vs.* f_n in a graph, as shown in Figure 1.2. This graph contains the same information as the original graph. Either graph allows you to uniquely determine $V(t)$.

If you didn't know in advance, it might not surprise you very much that the function $V(t)$ shown in Figure 1.1 can be written as a sum of oscillating cosine or sine functions. It pretty much looks like that kind of function. But, the surprising thing is that it turns out the *any* function $V(t)$ can be written as a (discrete or continuous) sum of cosine or sine functions. If $V(t)$ is a *periodic* function, then it can be written as a *discrete* sum of cosine and sine functions, like this

$$V(t) = \sum_{n=0}^{\infty} \tilde{V}_n^{(c)} \cos\left(2\pi f_n t\right) + \sum_{n=0}^{\infty} \tilde{V}_n^{(s)} \sin\left(2\pi f_n t\right) \tag{1.2}$$

where n is an integer, and $f_n = nf_1$, with f_1 the *fundamental frequency.* Equation (1.2) is called a *Fourier series,* and the statement that any periodic function can be written this way is called *Fourier's theorem.* Sometimes it is possible to write $V(t)$ with the cosine terms only or the sine terms only. In those cases the equation is called a *Fourier cosine series* or a *Fourier sine series.*

Figure 1.2. A graph showing how the amplitudes $\tilde{V}_n$ depend on the frequencies f_n, for the function plotted in Figure 1.1.

The example of Figure 1.1 and Figure 1.2 is a Fourier cosine series with $f_1 = 10$ Hz.

If $V(t)$ is not periodic, then it can be written as a *continuous* sum of cosine and sine functions. A continuous sum is just an integral, and the integral expression for $V(t)$ is

$$V(t) = \int_{-\infty}^{\infty} \tilde{V}(f)\left(\cos\left(2\pi ft\right) + i\sin\left(2\pi ft\right)\right)df \qquad (1.3)$$

where $i = \sqrt{-1}$. Equation (1.3) is called a *Fourier integral.* $\tilde{V}(f)$ is called the *Fourier transform* of $V(t)$. In your lecture course you will learn how to calculate $\tilde{V}(f)$. The only important thing to know for this lab is that it can be calculated, and once you've done that you can write the initial function $V(t)$ as the integral expression (1.3). Again, this basically the continuous limit of equation (1.2). Also, the formula will be a little different with frequencies ω in radians per sec.

<u>Effect of finite sample length on Fourier transform</u>

In laboratory work, one often records a sample of length *T* from a signal of much longer duration. One common situation is the recording of data by a digital storage oscilloscope, in which the sample length is often just a little longer than the duration of one sweep of the scope. The Fourier transform of the signal depends on the length of this sample. For example, consider a sample of a cosine function of duration *T*:

$$V(t) = \begin{cases} \cos\left(2\pi f_0 t\right), \text{ if } 0 < t < T \\ \quad 0, \text{ otherwise} \end{cases} \qquad (1.4)$$

This function is shown in Figure 1.3 for the case $f_0 = 200$ Hz and $T = 15$ ms. The magnitude of its Fourier transform $\tilde{V}(f)$ is shown in Figure 1.4. (To plot the Fourier transform $\tilde{V}(f)$ itself, we need to plot out both its real and imaginary parts. But the magnitude of $\tilde{V}(f)$ is a single real function, and often it's sufficient to just plot that out.)

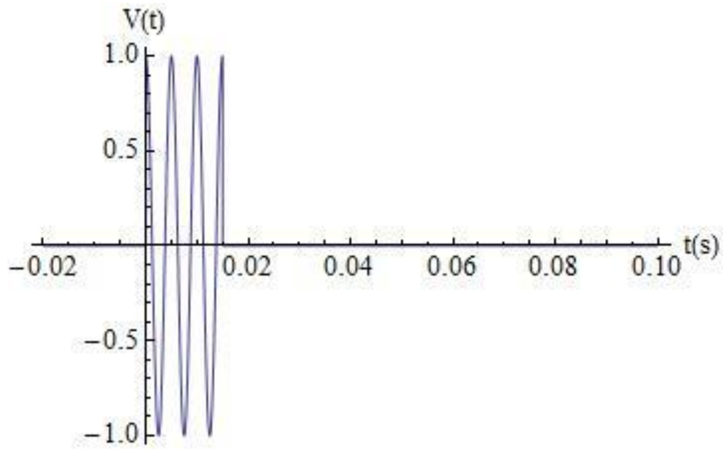

Figure 1.3. A sample of a cosine function of length T = 15 ms.

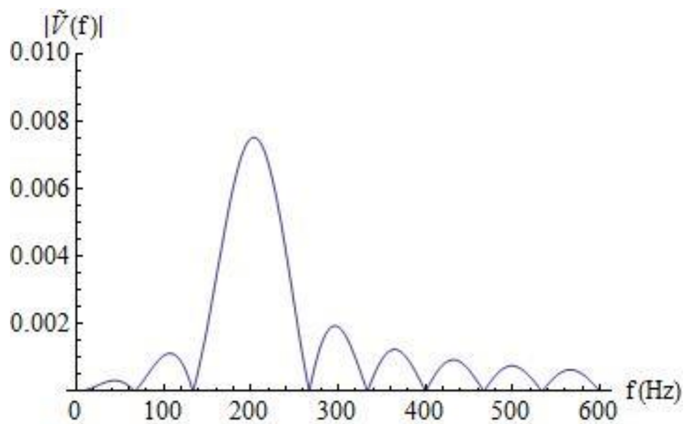

Figure 1.4. The magnitude of the Fourier transform $\tilde{V}(f)$ of the function $V(t)$ shown in Figure 1.3.

You might think that the Fourier transform should have a very sharp spike at frequency f_0, since the function is basically a cosine wave at the frequency. It does have a peak there, but the peak is kind of fat. The reason is that the function $V(t)$ is not a pure sine wave, but a wave that is clipped by the finite sample length so that it is only a few cycles long. Such a waveform has a spread in its Fourier transform that is roughly equal to $1/T$ (about 67 Hz, in this case). If the cosine wave lasts much longer than T, this spread in the Fourier transform is just an artifact of the finite sample length, not a characteristic of the signal itself.

If you want to avoid this artificial broadening of the peaks in a Fourier spectrum, the way to do that is to just record a longer sample length. For instance, Figure 1.5 shows what happens if I change T to 80 ms. (Mathematica code used to produce this figure is given at the end of this section of the manual.)

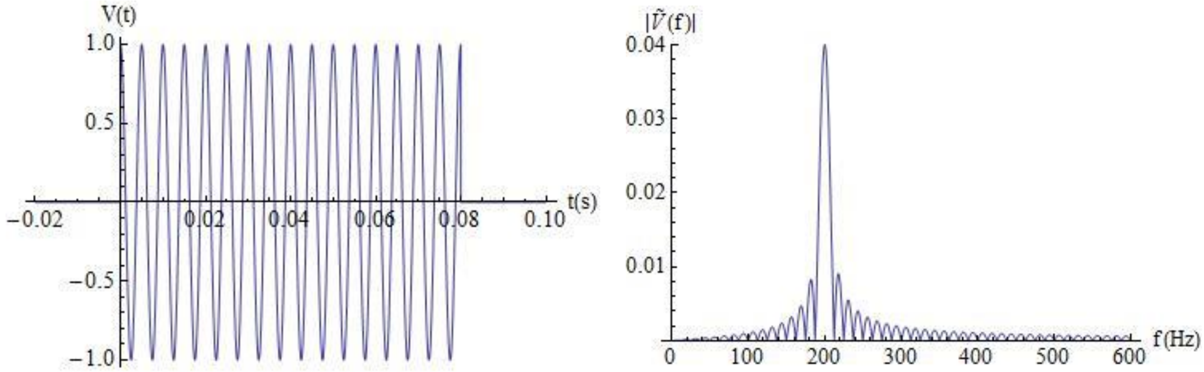

Figure 1.5. $V(t)$ and $|\tilde{V}(f)|$, with all parameters the same as in Figure 1.3 and Figure 1.4 except that T has been increased to 80 ms.

You can see now that the peak in the Fourier spectrum has become much sharper. In the lab, if you are measuring FFT (Fourier) spectra that have sharp frequency peaks, you will want to use samples of the signal that contain many oscillations, so you get sharp peaks like this.

Discrete representation of a function, sampling rates

Computers can only deal with sets of numbers, not true analog signals. For instance, in your lab you will use an interface that contains an analog to digital (A/D) converter. You'll plug an analog voltage $V(t)$ into the A/D converter. This converter will measure the input voltage at a sequences of equally spaced times $t_m = m\tau,\ \ m = 1,2,3,...$, as shown in Figure 1.6. The sequence of numerical results of these measurements is transferred by the interface to the computer. That is, the computer is recording a discrete approximation to the actual analog signal.

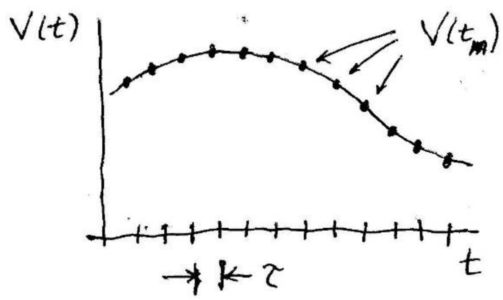

Figure 1.6. Discrete representation of a function $V(t)$ by a sequence of samples $V(t_m)$.

The *sampling rate* is the number of measurements per second, equal to $1/\tau$. The sampled digital signal will be a reasonable approximation to the true analog signal provided that the sampling rate is fast enough, and that the measurements are reasonably accurate.

Discrete Fourier Transform and Fast Fourier Transform (FFT)

A computer can be programmed to carry out a discrete approximation to the Fourier transform of equation (1.3), using as input the discrete approximation to the function as shown in Figure 1.6.

This is called a *discrete Fourier transform.* The result will have the form of a discrete function $\tilde{V}(f_m)$, where $f_m = mf_1$, $m = 0,1,2,3,...$, and f_1 is a fixed frequency that is related to the sampling rate. Note that this is different than the discrete case of the Fourier series given in equation (1.2). For the Fourier series, the discrete frequencies are determined by the periodicity of the function. For the discrete Fourier transform, the discrete frequencies are an artifact of the digital sampling, and will change with the sampling rate. Normally the interval f_1 between these frequencies should be small enough that $\tilde{V}(f_m)$ is "practically continuous."

There are different mathematical algorithms than can be used to compute a discrete Fourier transform. A *fast Fourier transform (FFT)* is a specific kind of numerical algorithm that, not surprisingly, can calculate a discrete Fourier transform very quickly. An FFT function is built into most data analysis software packages. In this lab, the data acquisition system is equipped with a real-time FFT function, which displays the FFT of the signal as the data is coming in.

There are many further mathematical details to this topic, which we'll leave for the lecture course. For the lab, you just need to have some basic understanding of what a Fourier transform is. The key points to understand are:

- *Any* function $V(t)$ can be written as either a discrete sum or a continuous integral over oscillating cosine and sine functions $\tilde{V}(f)\cos(2\pi ft)$ and $\tilde{V}(f)\sin(2\pi ft)$. In the continuous case, the function $\tilde{V}(f)$ is called the Fourier transform of $V(t)$.
- $\tilde{V}(f)$ represents the "how much" oscillation at frequency f is "contained" in the signal $V(t)$. If the function $V(t)$ is just a sum of single frequency oscillations as in equation (1.1), then the Fourier transform will contain one peak for each frequency component, as in Figure 1.2. In the more general case $\tilde{V}(f)$ will be a continuous function. The greater the extent of fast changes in $V(t)$, the greater the amplitude of $\tilde{V}(f)$ at high frequencies.

Normal modes

A *normal* mode is a vibration of an extended object at only one frequency. A vibration can always be characterized by some kind of displacement function $\psi(\vec{x},t)$, where ψ is the displacement, $\vec{x}$ is the position within the object, and *t* is the time. For example, the vibration of a stretched string can be characterized by a displacement $\psi(x,t)$, where ψ is the distance the string is displaced from its relaxed state, and *x* is the position along the length of the string, as shown in Figure 1.7a.

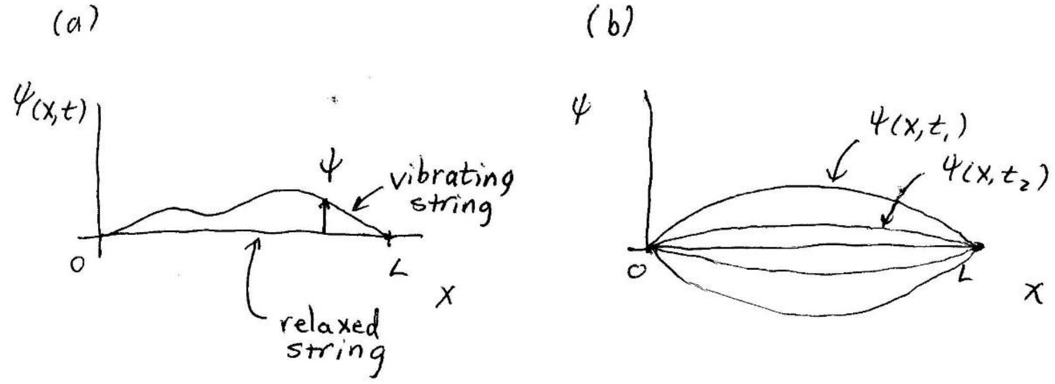

Figure 1.7. (a) Displacement of a string at some fixed time *t*. The displacement ψ measures how far away the string has moved from its relaxed state. (b) Illustration of a string vibrating in its lowest frequency normal mode, at several different times $t_1, t_2, \ldots$

When the string vibrates in a normal mode, its displacement can be written as

$$\psi(x,t) = Ay_n(x)\cos\left(2\pi f_n t + \phi\right) \tag{1.5}$$

where *A* and ϕ are constants. That is, the displacement factorizes into a function of *x* times a function of *t*, and the function of *t* is just a sinusoidal oscillation at one frequency f_n. The different parts of the string will oscillate by different amounts – depending on $y_n(x)$. But if you focus your attention on any one piece of the string, that piece will oscillate as $\cos\left(2\pi f_n t + \phi\right)$. This behavior is illustrated for the lowest frequency normal mode of a string in Figure 1.7b.

Objects don't usually vibrate in just one normal mode – real vibrations are typically more complicated. But, the normal modes are still important to understand. The reason, as you'll see in this course, is that *any* vibration of an object can be written as a linear combination of its normal mode vibrations.

Sonometer

A sonometer is a guitar-like instrument with one or more wires held under tension over one fixed bridge and one movable bridge, as illustrated in Figure 1.8. The wire is held under tension by slotted weights hanging from one end of the wire. The section of the string between the two bridges can be set into vibration by plucking it. As in a stringed instrument, the vibrations of the string transmit vertical vibration to the fixed bridge, which in turn causes the upper surface of the sound box to vibrate. As it does, air is drawn or expelled through holes in the sound box. This vibrating air mass transmits sound waves into the surrounding area.

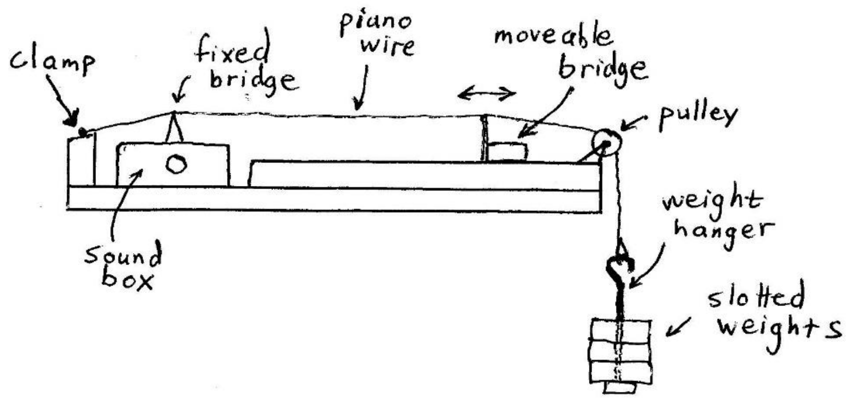

Figure 1.8. A sonometer.

The length of the vibrating section of string can be varied with adjustment of the moveable bridge. The tension in the string can be quantitatively adjusted with the addition of removal of weights. Thus, the sonometer allows the experimenter to study the vibration of the wire as a function of its length and tension.

D. Experimental Procedure

Vibrations of a Spring

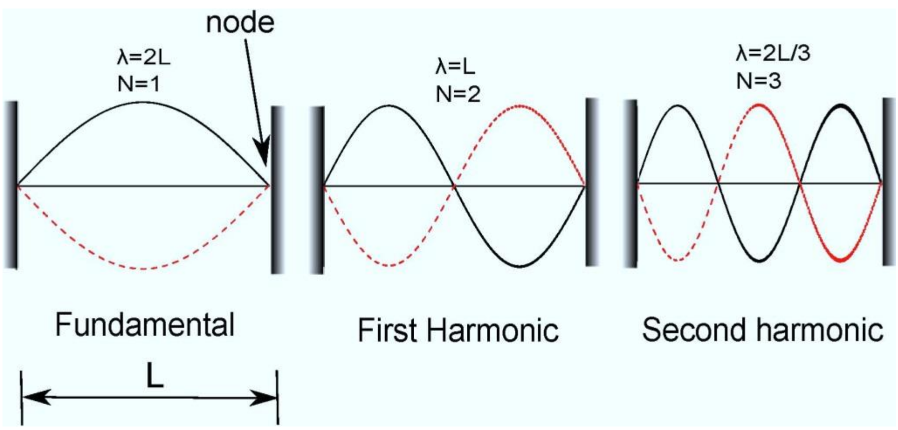

Figure 1.9. Illustration of the first three transverse vibration modes of a stretched spring.

A spring has both longitudinal and transverse vibration modes. The displacement in a longitudinal mode consists of compressions and extensions of the spring, without any sideways motion. The displacement in a transverse mode consists of sideways motion of the spring, as illustrated in Figure 1.9.

In the first part of this lab, you will study the transverse normal vibration modes of a spring. Stretch the spring out between yourself and your lab partner. You'll study the modes of the "clamped" spring, in which both ends of the spring are held fixed. So, one of you should hold his or her end of the spring in a fixed position. You'll have to move the spring a little to get the vibration mode going, so the other person will have to move the end of the spring slightly to get the motion going. After that, the second person can hold their end fixed, in order to observe the free vibrations of the spring with both ends held fixed.

<u>Measure the length of your stretched-out spring, and be sure to keep that length the same throughout this part of the lab.</u>

<u>Produce the first three vibration modes of the spring</u> as illustrated above. Using your timer, <u>determine the frequency of each mode</u>. A *node* is a point where the vibration amplitude is zero. Note that the ends of the springs are nodes in this experiment, and that the first harmonic has one additional node, the second two additional nodes, *etc.* Plot the frequency of the modes you have measured vs. the "scale length", where the "scale length" is equal to the distance between adjacent nodes.

<u>See if you can excite the fourth mode. If you can, measure and plot the frequency of that one, too.</u>

Does there seem to be a relationship between the frequency and the scale length? If so, what is it? If you see a relationship, give an explanation for it.

Plot the mode frequency *vs.* the inverse of the scale length. As always, be sure to put correct units on your plot.

<u>Determine the product of the each mode's frequency and its scale length</u>. Note this quantity has units of velocity.

<u>Measure the velocity of a pulse propagating down the spring</u>. Try this both with a transverse pulse and a longitudinal pulse, and see if the velocities of the two vibrations are the same. Make sure to use the same spring length as you used in your previous measurements. Compare the velocity you measure with the velocity you calculated from the vibration frequencies in the previous step.

<u>Sonometer set-up</u>

Set the sonometer on the bench with the pulley hanging off of the side of the bench. Keep the hanging wire within a few inches of the table edge, since otherwise the weights can tip the sonometer up off of the table. You may find the apparatus easier to work with if you place the computer keyboard and mouse in front of the sonometer.

Make sure the sonometer wire runs over both bridges and over the pulley as shown in Figure 1.8. Place some mass on the weight hanger. It is possible that the sonometer wire will break. If this happens, the wire presents a hazard to your face or eyes, and the weights present a hazard to your feet. For this reason, **when you add mass, keep your face clear of the wire and your feet clear of the weights.**

Most of the sonometers are strung with #3 piano wire. This is an 0.013" diameter wire made of a high strength steel with an ultimate yield stress of 2.2 to 2.5 GPa. The ultimate yield stress is the material stress (force per cross-sectional area) which will cause the wire to break. If you work it out, these wires will break if their tension exceeds about 188 N, or if they are supporting a mass of more than about 19 kg against the force of gravity. However in practice the wires will break at a somewhat lower tension due to nicks, sharp bends, or other weak spots in the wire. For this reason, **do not exceed a mass of 10 kg on your hanger**. In practice, you should be able to complete the experiments with about 6 kg of mass or less. Some of the sonometers are strung with thicker wire up to 0.018" diameter, but about 6 kg of mass or less should work with those, too.

Notice the change in pitch with increasing mass. Continue to increase the mass until you have a pleasant tone.

<u>Set up to take sonometer data on the computer</u>

In this class, we will be using a computer data acquisition system consisting of a Science Workshop 850 Interface (SW850) and a Mac Computer running OS X. We will use Pasco Capstone software to control the SW850, and to take and display data. Rather than spend lots of time learning Capstone, please go through the following sequence of steps, which will walk you through the subset of Capstone features that are needed to take data for this experiment. You do not need to include anything about these steps in your report; this part of the lab is just to familiarize you with how to operate the data acquisition system.

To get started, plug the Pasco sound sensor into Analog Channel A of the Science Workshop 850 Interface. Turn on the computer and SW850 using the switches in the back left corner, and log in to the computer, if these steps have not been done already. You will log into the computer using your UT eid and password.

Click on the pyramid-shaped icon. This will start Capstone. Click on "hardware setup". Then click on channel A on the image of the SW850 that pops up. Select "Sound Sensor" from the drop down list. Click on "Hardware Setup" again to end display of the hardware setup window.

The large middle area of the screen is the display area. To collect and display data with Capstone, you must open up one or more of the Display options shown in the Display column on the right hand side of the screen. The ones of interest for this lab are "Graph", "Scope", and "FFT".

Click on "Scope" and drag to the display area. You should see an empty graph. Then, click on "<select measurement>" next to the vertical axis. Select "Sound Intensity (V)" from the drop down menu. This causes the program to display the output of the sound sensor on the vertical axis, and time on the horizontal axis.

I will continue to use the term "Sound Intensity" to match Capstone's user interface. However, this is a physics error. Intensity is proportional to displacement squared, and is always a positive number. For a simple oscillation, an intensity should be proportional to $\cos^2(2\pi ft)$. The sound sensor output is proportional to the acoustic (sound) displacement, *not* the acoustic (sound) "intensity". This is obvious from the signal itself, because it looks roughly like a sine wave and goes both positive and negative.

Data acquisition controls are placed in the row at the bottom of the screen. One is labeled "Continuous Mode". Click on this icon, and you'll see there are three choices, "Continuous Mode," "Keep Mode", and "Fast Monitor Mode". Select "Fast Monitor Mode." At this point, the data acquisition functions as a digital storage oscilloscope. As data comes in, it is displayed real time on the screen. The time scale of the display is fixed. So, if the time scale is 10 ms total from left to right, the scope will display 10 ms of data, then clear the display and show the next 10 ms of data, etc. This makes it easy to view the details of your signal as it comes in.

Click on monitor, and pluck the string a few times. You should see the signal from the sound sensor on the display. You should also see that the system functions much like an oscilloscope. Experiment with the placement of the sound sensor until you find a position that gives a strong signal. You may find it helpful to use a wood block to set the sensor on.

Pressing "Stop" will halt display of the incoming data, and pressing "Monitor" will restart it.

The SW 850 contains an analog to digital converter, as discussed in the introduction. The sampling rate of the SW850 can be set as high as 10 MHz in single channel applications. In practice you'll need MUCH less sampling rate than this, since you are viewing audio signals which have very little frequency content above a few kHz. Capstone usually sets the sampling rate to an appropriate value automatically, so you probably won't need to adjust this. The sampling rate is shown (some number of kHz) in the box next to "Sound Sensor" in the bottom row. This rate can be adjusted with the up and down arrows, if necessary.

You can compress or expand the time scale by clicking and dragging anywhere near the horizontal axis. Try it. You will notice that the sampling rate automatically adjusts to a shorter interval between samples for a shorter time scale. Try viewing the signal for several different time scales, such that you can see a single oscillation cycle highly magnified, or see many oscillations across the screen.

There are a number of tools along the top edge of the scope window. One is a control for the scope trigger. The default is for an untriggered repeating sweep, but you can change that to a triggered sweep on positive or negative slope with an adjustable trigger level. You can also do things like set the scope for a single sweep or establish a voltage offset. The settings for both horizontal and vertical sensitivity are controlled by clicking and dragging near the appropriate axis.

You will also need the FFT Display in this experiment. Leave the open Scope display in the display area. Click on "FFT" and drag into the display area. You should see an open Scope display and an FFT display side by side. Set the vertical axis of the FFT display to measure

"Sound Intensity". With "monitor" engaged, pluck on the string. You should see the scope trace of the signal and the Fourier transform of that signal in the FFT window. (Capstone is actually plotting the magnitude of the Fourier transform.)

Most likely, the frequency range that is displayed for the Fourier transform will not be the appropriate range for this measurement. This range is set by default to ½ of the sampling rate. This has to do with the lack of information content in the digitized waveform at frequencies higher than ½ of the sampling rate. (Google "Nyquist Theorem" if you want to know more about this.) You can change the frequency range for the Fourier transform directly by clicking and dragging on the frequency axis. This happens without any effect on the Scope time scale. Alternatively you can change the FFT frequency scale indirectly by clicking and dragging on the Scope time scale – this changes both at the same time. With a combination of these two adjustments, you can get to pretty much any FFT frequency range and scope time scale that you want.

In practice you are going to want an FFT frequency scale that is something like five to ten times the fundamental mode vibration frequency of the string. Try setting the scale for 2 KHz range to start with and see what the signal and FFT look like. You should start to see one or more peaks in the FFT. The true Fourier transform of your string vibration should have very sharp peaks, because your string is vibrating at very well defined frequencies. If the peaks in your FFT display are not sharp, this is because your data trace does not contain very many cycles of oscillations (so it doesn't look all that much like an ideal sine wave), as discussed above. Your goal should be to get a scope display that has quite a few cycles of oscillation, while still having an appropriate frequency range on the FFT display. See if you can get it do to that.

Once you see many cycles of oscillation of the Scope display, and a frequency spectrum (FFT display) with clear, sharp peaks, the system is set up to record data. For some measurements you'll want both displays open. If you are only measuring frequencies, you can close the scope display and use only the FFT display if you'd like. (This may make it easier to have the appropriate frequency and time range for FFT analysis.) To close a display, you click on it so that the entire display is selected, and then click on a red X in the upper part of the screen.

If you want to measure frequencies on the FFT display, there is a measurement tool that you can use at the top of the FFT window. When you click on it, it opens up a little cursor that you can drag around the screen. When the cursor gets close to a peak, it automatically hops onto the peak, and you can read out the coordinates of the peak.

The data that you are viewing in Fast Monitor mode is not permanently stored in the computer. However, you can store the last sweep displayed after the stop button is pressed. (This could be a triggered single sweep, if you want.) To do that, with "Stop" engaged, click on the multi-colored triangle with the red diagonal arrow above the scope display. You can tell that your data was stored by clicking on "Data Summary." This will show a listing of stored runs. You can close the summary window by clicking on "Data Summary" again.

You can also record the data from the FFT display, but the actual recorded data is the same as from the Scope window. In other words, you *cannot* store the Fourier transform of the signal as a function of frequency. Recording the signal from either window just stores the time-dependent signal. If you want the Fourier transform of your data after closing the FFT window, you'll have to reconstruct it from the saved time-dependent data. For this lab, it will be OK to just print out a few spectra to paste into your report if you want to do that.

Once the data is listed in the Data Summary window, you can view it later in a Scope Display, Graph Display, or FFT Display. You can also view it in spreadsheet form by clicking on "Table", dragging to the Display area, and selecting the appropriate columns and data set.

There are two main ways to save data to your USB drive. One is to save the entire experiment, through either the diskette icon or the file menu. This will save everything in a Capstone-compatible format, so if you open the file back up in Capstone you'll get back to where you are when you saved the file. The other way is to use File>Export Data in the top level menu. This saves a Tab or Comma delimited file that you can open in other programs such as Excel. Each run is saved as two adjacent columns containing the time and recorded signal. Whatever data that you want to keep should be saved in this way to your USB drive. Try saving a data set in tab-delimited form, and then viewing that data with Excel.

Capstone has many other useful features and controls. There are do/undo controls in the upper part of the screen. If you click on Capstone help you can find your way to a user guide.

This is a very abbreviated set of instructions, and Capstone has many more features and controls that you may want or need to use. Like most modern software, it is supposed to be intuitive, and hopefully that will be the case.

<u>Sonometer measurements – waveforms, harmonic content, and oscillation decay</u>

<u>With the moveable bridge pulled near the pulley, pluck the string and record its waveform and FFT spectrum.</u> You should see a spectrum consisting of a number of sharp peaks, that correspond to the fundamental mode and a number of harmonics. (If the fundamental mode has a relatively low amplitude in the spectrum, this may be due to the fact that the sound sensor response falls off at low frequencies.)

<u>Pluck the string in different ways. Try the soft part of your finger *vs.* your fingernail. Try plucking the string at different positions.</u>

Which quantities change and which stay the same with different ways of plucking the strings? (Frequencies? Waveforms? Strength of the harmonics?) How does the harmonic content correlate with the waveform you see? With the sound you hear? Can you find a way to produce a vibration that consists almost entirely of the fundamental mode? Can you produce a vibration that consists almost entirely of one of the harmonics? <u>Document some of these results with recordings of the waveforms and FFT spectra.</u>

Notice the decay of the waveform and harmonics as the string vibration loses its energy. Do the harmonics decay at the same rate as the fundamental? If not, can you give a reason for why the rates of decay are different?

Sonometer measurements – fundamental vibration frequency *vs*. vibrating wire length and tension

Using your apparatus, measure the frequency of vibration of the fundamental mode of the wire as a function of the length of the vibrating wire section, with the wire tension held fixed.

Using your apparatus, measure the frequency of vibration of the fundamental mode of the wire as a function of the wire tension, with the length of the vibrating wire section held fixed. (The masses are mostly in fixed 1 kg and 2 kg sizes, and some are marked. If you have any doubt about a mass, there is a scale near the door that you can use. And, don't forget the mass of the hanger.)

Plot your results, and determine the functional relationships between the vibration frequency and length, and between the vibration frequency and tension. Show fit curves to your data for these two relationships.

Summary

In your report, present a brief summary of your results.

Appendix

Mathematica code used to produce Figure 1.5:

```
f0 := 200
T := 0.080
V[t_] := UnitBox[(t/T)-(1/2)] Cos[2 Pi f0 t]
Plot[V[t], {t,-0.02,0.10}, AxesLabel -> {"t(s)","V(t)"}, LabelStyle ->
Directive[FontSize -> 14]]
Vtilde[f_] = FourierTransform[V[t],t,f,FourierParameters -> {0,-2 Pi}]
Plot[Abs[Vtilde[f]], {f,0,3 f0}, PlotRange -> {0,0.04}, AxesLabel ->
{"f(Hz)", "|V(f)|"}, LabelStyle -> Directive[FontSize -> 14]]
```

You are not required to understand this code. I am just giving it for those students who may be interested in it. The instruction `FourierParameters->{0,-2 Pi}]` tells Mathematica to follow the convention used in these notes, in which frequencies are measured in Hz. It is necessary to include the absolute value function `Abs` inside the `Plot` instruction, since `Plot` only works with real functions, and the Fourier Transform `Vtilde` is complex. I don't know how to do the overscript tilde on the FFT vertical axis label with instructions in text form like this. If you cut and paste this code into an open Mathematica window, you can add the tilde by highlighting the V character and then applying the key sequence control-& followed by (shift) ~.