RESONANCE and the *Speed* of Sound:

Your Name:_________________________ Lab partner:_________________________.

1. Purpose

Previously, you have examined ***localized*** oscillators, where the mass was concentrated in, say, a pendulum bob on the end of a relatively light string, or in a mass on the end of a relatively light spring. **Continuum mechanics** deals with situations where you must consider how mass is ***distributed*** over space, the *rates* at which energy and momentum can be transferred across such systems and, as in today's lab, the phenomena of *resonance*.

2. Introduction

You will study properties of sound waves using the resonance tube apparatus shown in Figure 1:

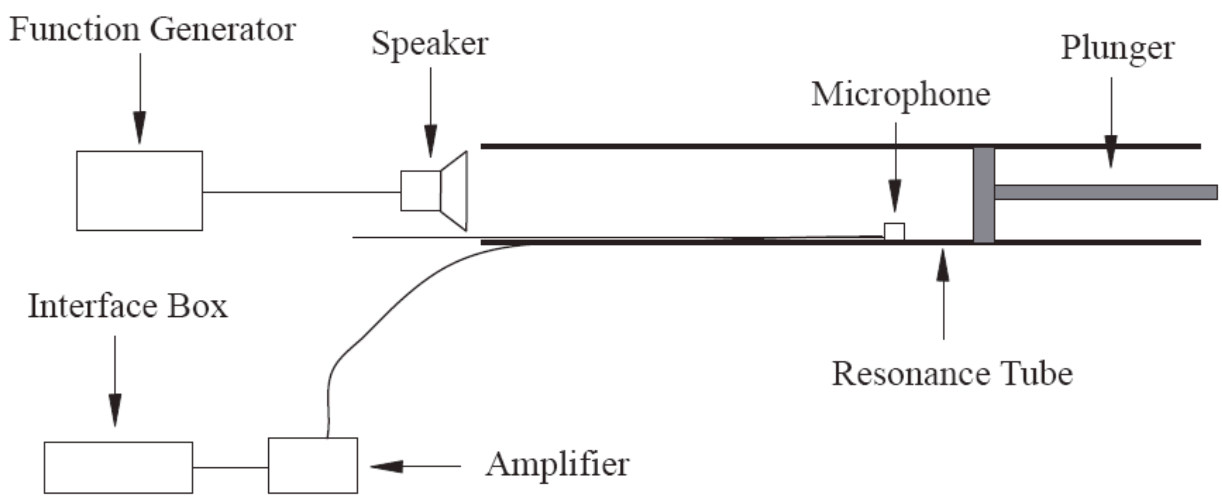


Figure 1

Sound waves are produced using a speaker that is located at one end of the tube. A *movable* microphone is used to probe the sound wave at various locations inside the resonance tube, whose effective length can be varied by moving the plunger.

A function generator is used to drive the speaker. Consult your TA or lab instructor on how to use your function generator. The microphone should be connected via a coaxial cable with a "BNC" (Berkeley Nuclear Corp.) plug. For our system, we next connect to a 3.5-mm jack adapter and then, via banana plugs, to an appropriate channel of the interface box (as a **voltage sensor**). Within all of this cabling, notice that there is a small black module containing an on/off switch: to preserve battery function, the module should be switched to the off position when not in use, but must be switched to the on position for operation.

When a speaker vibrates near a tube, there are certain frequencies at which the tube will *resonantly amplify* the sound from the speaker. This happens because the sound wave propagates down the tube and is reflected back and forth from each end in the tube. As the wave travels away from the speaker, it will encounter the wave that has been reflected from the end of the tube. Where this happens, we observe the *sum* of the two waves, a result known as *superposition*.

If the wavelength of the sound is related to the effective length of the tube in such a way that forward directed and backward reflected waves end up *in phase* with each other, the result is a self-reinforcing standing wave pattern. This is known as **a resonance mode** for the tube, and the frequencies at which such resonances occur are called the tube's resonant frequencies. Due to the self-reinforcing nature of the phenomenon, at these frequencies the sound will appear "louder."

Conversely, at *other* frequencies, where the forward directed and backward reflected waves end up (to various degrees) being *out of phase*, they will (at least partially) cancel each other out, which we describe as "destructive interference." If the two waves are exactly out of phase, they will cancel completely.

The frequencies that produce standing waves for a given effective tube length are referred to as *harmonics*. The lowest frequency that produces a standing wave is known as the *fundamental* frequency (or *first harmonic*).

In this lab, you will first search for standing waves in a tube that is open at one end (near the speaker), and then will search for standing waves in a tube that is closed at both ends. Based upon your exploration of the PhET simulation "Waves on a String," what do you think will be the consequences of these changes in "**boundary conditions**"?

When you have set your function generator to a frequency where a standing wave is formed (which you can do by maximizing the detectable signal within the tube), then the movable microphone can be used to ***map out*** locations within the tube where the amplitude is a maximum ("anti-nodes") and locations where the amplitude is a minimum ("nodes").

Before taking data, first try to **predict** what you expect to see:

For the tube that is closed at one end, where do you **predict** the nodes and anti-nodes will be?

For the tube that is closed at both ends, where do you **predict** the nodes and anti-nodes will be?

On the figure below, draw your predicted distribution of nodes and anti-nodes, for *each* of the first few harmonics, for a tube that is closed at one end.

❖ **Predict** a relationship between the length of the tube and the wavelength of the standing wave

❖ On the figure below, draw your predicted distribution of nodes and anti-nodes, for *each* of the first few harmonics, for a tube that is closed at both ends.

❖ **Predict** a relationship between the length of the tube and the wavelength of the standing wave. (Is the change in "boundary conditions" important?)

3.1 Qualitative Examination of Standing Waves

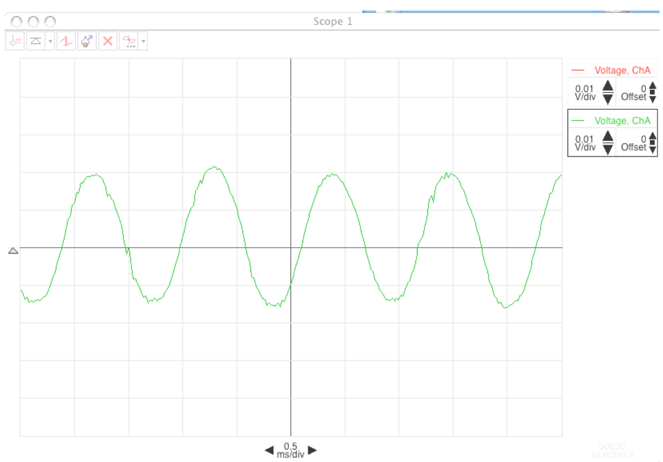


Figure 2

- To *view* signals as they change in time, you'll use the oscilloscope feature in *DataStudio*: drag the measured voltage from the microphone onto a "***Scope***" display. The scope window will have the sorts of axes shown in Figure. 2 above. The signal *measured* by the microphone should be all that you display on the "Scope" (*i.e.,* there is no need to display the *output* of the function generator).

  Before you START a measurement:

- Set the **sample rate** on *DataStudio*, to something *of order* 40,000 Hz
- Set the **gain** to 100
- Set the **voltage scale** to roughly 0.02 V/div

Once you have data captured, feel free to adjust the scales on your oscilloscope. (Feel free to call upon your TA or lab instructor for assistance.)

- Create a tube that is closed at only one end by leaving a small gap, ~1 cm, between the speaker and the resonance tube.

- You can vary the effective length of the tube by moving the plunger in and out. For now, set the plunger at the 80-cm mark on the tape inside the resonance tube.

- Start by driving the speaker at a frequency of *around* 1000-Hz: if there are other groups around, it might be helpful to try a frequency that is audibly different from what you are hearing from other groups. Whatever frequency you select should be recorded below. As you **vary the length** of the tube, use a microphone mounted at the end of the tube to measure the amplitude of the sound wave.

- Record (as many as possible of) the lengths of the tube when you appear to have resonance and, separately, those where destructive interference seems most pronounced.

Drive frequency: _________ Hz

Trial	Effective Tube Length ( )	Was this a Resonance or "Anti-Resonance"?
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		

How do your observations compare with your predictions, regarding the relationship between the length of the tube and the occurrence of resonances and “anti-resonances,” for a tube open at one end?

Next, repeat the experiment using a tube closed at both ends:

Trial	Effective Tube Length ( )	Was this a Resonance or “Anti-Resonance”?
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		

How do your observations compare with your predictions, regarding the relationship between the length of the tube and the occurrence of resonances and “anti-resonances,” for a tube closed at both ends?

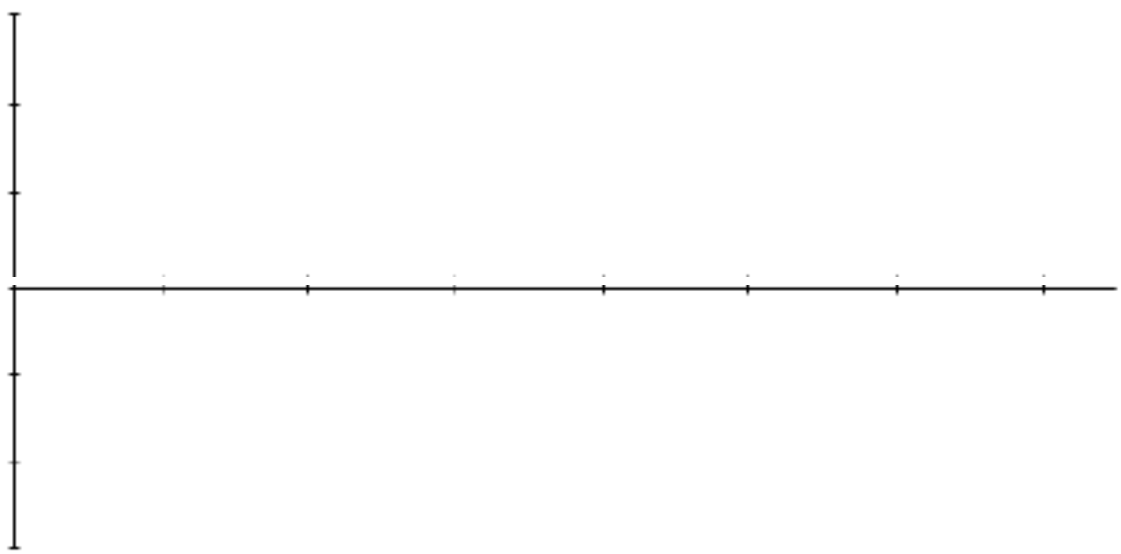
How would you *contrast* the case of a tube open at one end with the case of a tube closed at both ends?

3.2 Examining the structure of standing waves

Let's now examine the "internal structure" of the standing waves.

A microphone is a ***pressure*** transducer: a location where the microphone produces a maximum signal is a "pressure anti-node." Similarly, a minimum signal corresponds to a "pressure node."

- Adjust the tube length to one of the **resonance** configurations that you found in Part 1. (**Record** these conditions under which you will take your next data.)
- Start out with the microphone at the end of the tube near the speaker.
- With the microphone ***turned on***, record the amplitude of the wave as a function of position as you move the microphone through the tube.
- **Plot** the measured amplitude as a function of position. Be sure to take enough data points that you have a good feel for the structure of the wave.



- Does the observed structure of the standing wave agree with your expectations?
- What is the wavelength of the standing wave?

At this point, you know the *frequency* of the standing wave, because *you set it* (using the signal generator). Additionally, you just measured the wavelength of the standing wave. From these, describe ***how*** you would attempt to calculate the speed of sound, and then ***calculate*** the speed.

Your textbook notes that the speed of sound in air depends on the temperature, T (in *Celsius*):

$$v_{sound} = 331 + 0.6T \tag{1}$$

How does your experimentally derived value compare with the value expected from Equation 1?

3.3 Directly measuring the speed of sound

At this point, let's measure the speed of sound more directly. You can do this by creating a ***pulse*** of sound and, with the microphone positioned towards the end of the tube nearest the source, measuring how long it takes the pulse to travel, from the microphone position, down the tube *and back*.

From your previous work, you have noticed that sound travels reasonable distances over a fairly short amount of time. As such, we would like to make the distance that it travels as large as possible. Thus the plunger should be as far from the speaker as possible.

A sound pulse can be created by driving the speaker with a **square wave**. When you select a frequency of 10 Hz, you should hear a *clicking* sound.

As a single sound pulse *first* passes the microphone, you will see a signal similar to what is seen in Figure 3. This “ringing” is caused by the sudden voltage increase of the square wave applied to the speaker. (Perhaps this is an example “damped oscillatory motion,” hmm...)

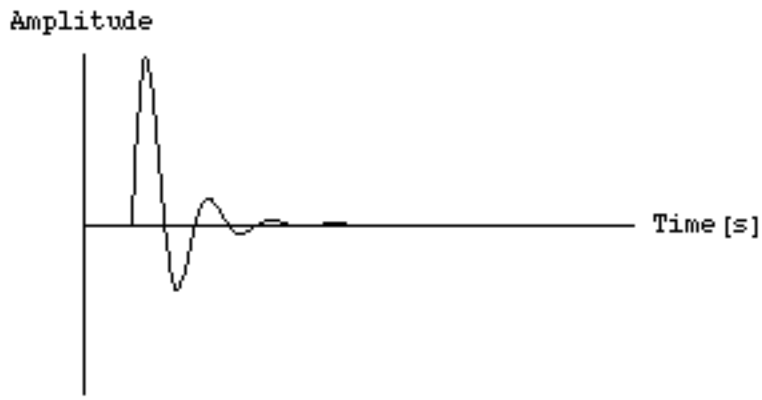
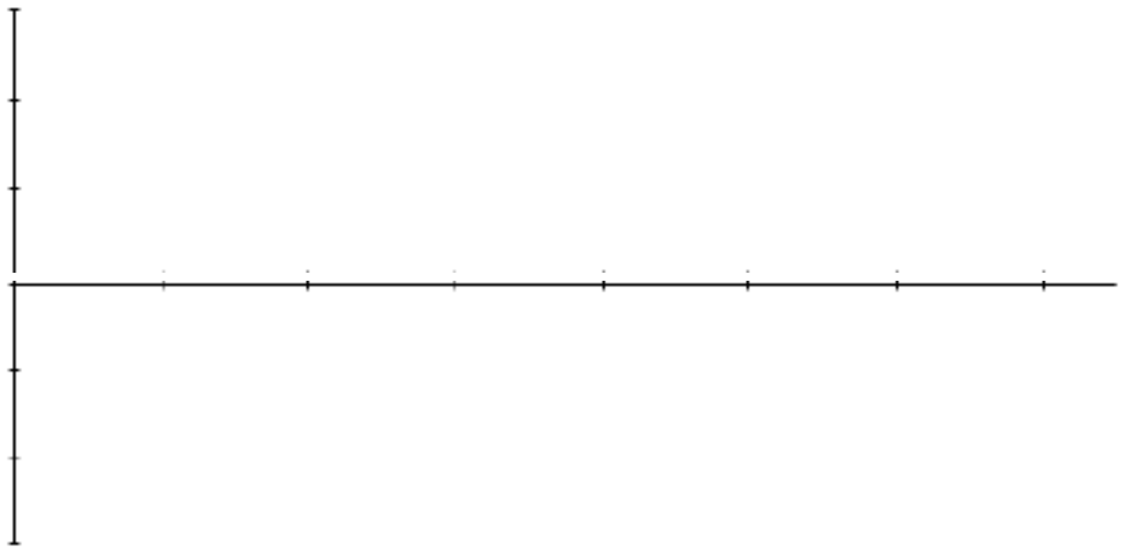


Figure 3

When this pulse reaches the end of the tube it will be *reflected*, creating an echo. What do you **predict** that the reflected pulse look like?

The echo will travel back to the speaker and be detected by the microphone. It will then be reflected from the speaker and travel back down the tube. This process will continue, resulting in observing a *series* of pulses, where the intensity will decrease with the number of multiple reflections. Based on what you have just worked through, and understanding that 10Hz is very slow, on the time scale you’ll want to display, what do you **predict** that you will see on your graph? (This exercise should help you to appropriately **set the time scale** on your graphical display.)



Because all of this happens very quickly, you will need to take a lot of data over a small amount of time. Luckily, you can tell the computer how much data to take, so this will not be a problem. At the same time, we don't want to be overwhelmed with a huge amount of data. Fortunately, you can also tell the computer when to start taking data. To do this, you first need to **delete** any data that you may currently have saved.

- To change how the rate at which data is taken, you need to change the "*Sample Rate*" such that you will be able see the reflected pulses. Increase this value to *no more than* 40000 Hz.

- To tell the computer when to **start** taking data, select "Sampling Options" under the "Experiment" menu. You should see the window shown in Figure 4 below. Select Channel "*Output Voltage"* and set the value to 0.02V *"Falls Below."* This causes the computer to start taking data once the speaker stops creating a pulse.

- To tell the computer when to **stop** taking data, select "Automatic Stop" under the "Experiment Setup" menu. You should see the window shown in Figure 5. Set the time to an appropriate value, based upon your prediction.

- To perform your measurement, display your data as a function of time using the **graph** feature (rather than the oscilloscope feature).

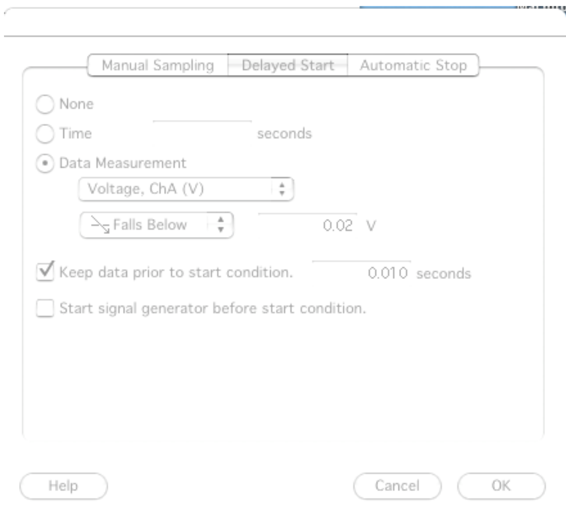


Figure 4

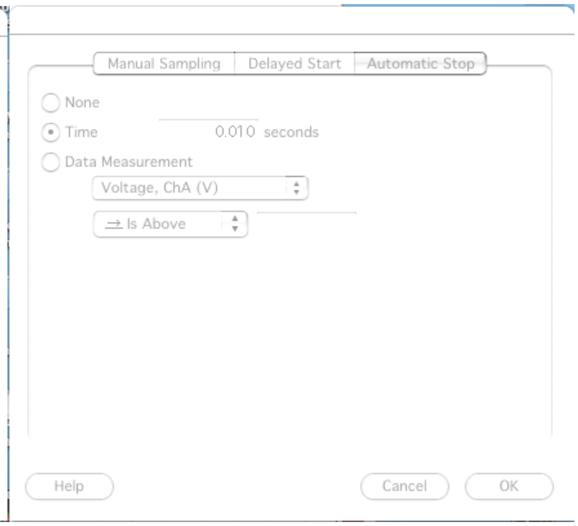
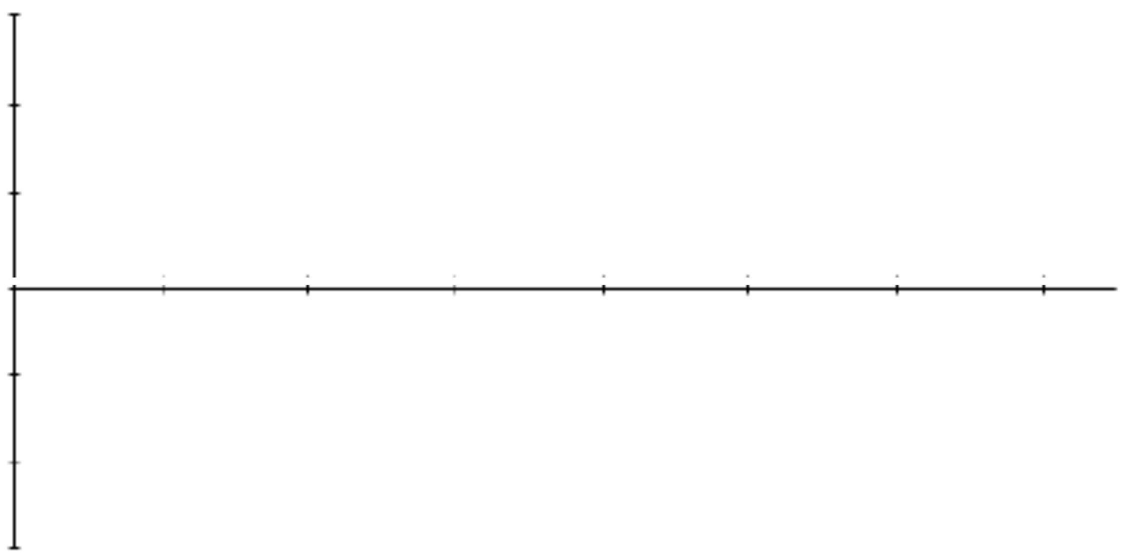


Figure 5

➢ Let's take data! In the space below, **sketch** what your microphone measures:



❖ *How far* does the sound wave travel in between each of these *observed* pulses?

In the table below, record the distance traveled and the time taken to travel these distances.

Time ( )	Distance Traveled ( )

❖ Do these results agree with your predictions? Discuss.

Using your favorite plotting software, **plot** the distance traveled by the wave as a function of time. Be sure to include your plot and subsequent discussion in your lab notebook.

To preserve battery function, the module should be switched to the off position when not in use.

4. Questions

i. Is the speed of sound ***constant***? Justify your answer **using your data**.

ii. What is your directly measured result for speed of sound? Explain how **you** found this value.

iii. How does this value compare with your previous result? With the theoretical expectation?

5. Initiative

Possible ideas:

It is *simple* to add an independent measurement of the speed of sound with an IOLab device.

Alternatively, check out Pivot Interactives:

- Exploring Wave Propagation Speed in a Ruben's Flame Tube

6. Conclusions