Portland State UNIVERSITY

DEPARTMENT OF
ELECTRICAL AND COMPUTER ENGINEERING

ECE331

Lab1: Acoustic Waves

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1.1 EXPERIMENT (Characteristic of sound)

Objective:

In this experiment, we will be measuring the traveling wave characteristics of sound waves including wavelength, velocity and spreading loss. Sound waves in air are three-dimensional and their amplitude is inversely proportional to the distance from the source. This is spreading loss, not attenuation. Since the energy of a spherical wave is distributed evenly over the surface of the spherical wave front, the wave amplitude must therefore decrease with the spherical waves radius in order to satisfy the law of conservation of energy.

Set Up:

Setup the experiment as shown in Figure 2:

- Connect the StarTech USB Audio interface to a USB port on the computer.

• In Windows Settings configure the computer to use the newly attached StarTech device for audio output AND microphone input.
• Connect the desktop speakers to the audio SPDIF output port on the back of the StarTech device.
• Connect the stereo Y-adapter to the audio input port on the front of the StarTech device.
• Connect the reference microphone to the gold Y-adapter port using a stereo-mono adapter.
• Connect the signal microphone to the aluminum Y-adapter port using a stereo-mono adapter.
• Open the Visual Analyzer program (A standalone version can be downloaded from http://www.sillanumsoft.org/Download/VA.exe)

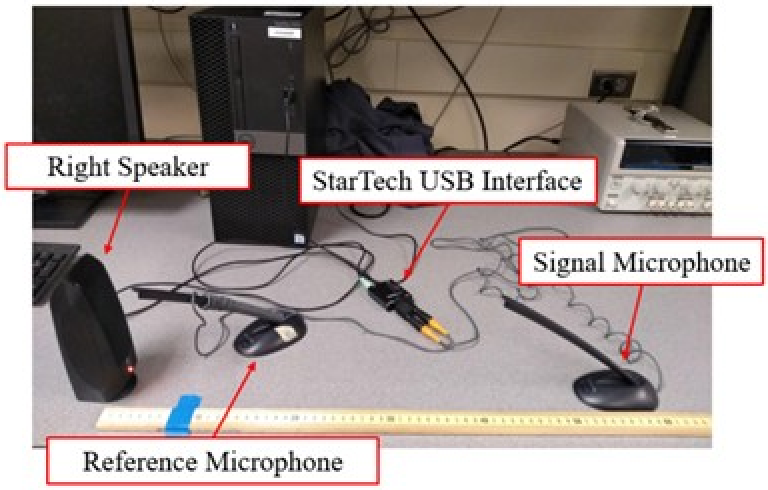


Figure 2: Experiment 1 setup.

• Configuring the Waveform Generator:
  – Open the Waveform Generator window by clicking the Wave button.
  – Disable Left (A) Channel by unchecking the Enable checkbox.
  – Set the frequency for channel B to 2000 (Hz) and click the Apply button.
  – Set the Wave Function for channel B to "Sine".
  – Close the Waveform Generator window by clicking the X button.

• Configuring the Oscilloscope: – Open the Settings window by clicking the Settings button.
  – On the Main tab, set Channel(s) to "A and B".
  – On the Scope tab, set Y-axis to "Volt".
  – Close the Settings window by clicking the OK button.
  – In the Oscilloscope window, check DC removal and Values for both channels.

• Calibrating Visual Analyzer:
  – Position both microphones beside each other and directly in front of the right speaker.
  – Turn the volume control on the side of the speaker to ≈ 50%.
  – Open the Settings window and select the Calibrate tab.
  – Turn the Waveform Generator ON by clicking the Wave On button. (Note: If you do not see a 2 kHz sinusoidal signal in the Oscilloscope display you may need to increase the input gain of the microphones in the WindowsTM Sound Control Panel.)
  – Slowly adjust the speaker volume to display reasonably strong and "clean" microphone signals

on the oscilloscope display while keeping the noise level tolerable. (Remember that others are trying to do the same experiment, so try to use the lowest volume possible.)
– Open the Settings window by clicking the Settings button.
– On the Calibrate tab, using the default settings, click the Start Measure Signal button.
– Enable the calibration by checking the Apply Calibration Settings checkbox. When the calibration is completed both channels should have a Vpp voltage of ≈ 1 V, see Figure 3.
– Close the Settings window by clicking the OK button.

Theory:

Spreading Factor of Spherical Waves:

These plane waves are solutions to the wave equation (7.15) on p. 316 of Ulaby[1] in rectangular coordinates. In contrast, spherical waves are solutions to the wave equation in spherical coordinates. A spherical wave emanating from a point source at the origin is described by Equation.

$$A_{sph}(r) = \frac{A_0}{r} e^{-jkr}$$

A spherical wave is three-dimensional and has spherical wave fronts that propagate outward from a source at the origin. Such electromagnetic or acoustic wave sources are often described mathematically as point sources. The energy of a spherical wave is distributed evenly over the surface of the spherical wave fronts. The wave amplitude must therefore decrease with the spherical wave radius in order to satisfy the law of conservation of energy even in the absence of attenuation. This spreading loss is accounted for by the 1/r dependence in Equation (1). This is different from the attenuation due to propagation through a lousy medium. In a lousy medium k = kR − jkI , would be complex and the attenuation loss would be accounted for by kI . Note that the spreading loss term causes the amplitude to go to infinity at the origin (i.e. at the point source). Physically, a true point source can not exist, so

Equation (1) must be considered approximate. It is valid as long as the point of observation is in the far-field of the source, or Interference in 3-D:

$$r \geq \frac{2D^2}{\lambda}$$

As with any waves, multi-dimensional waves can generate interference patterns. For example, Figure 1 shows two isotropic (directionless) speakers set up a distance d apart.Each speaker then generates a time-harmonic acoustic wave with a wavelength λ. For each speaker the pressure wave can be described as,

$$A_1(r_1) = \frac{A_{1,0}}{r_1} \exp^{-jkr_1+\phi_1}$$
$$A_2(r_2) = \frac{A_{2,0}}{r_2} \exp^{-jkr_2+\phi_2}$$

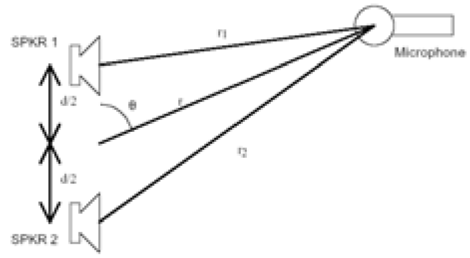


Figure 1: Geometry of an Acoustic Experiment.

where k is the propagation constant, r1 and r2 are the distances between the speakers and the observation point (microphone) and A1 and A2 are the pressure waves for speaker one and two, respectively. For either of these speakers alone, the reference phase φ doesn't matter and the microphone will measure the same signal as long as the range remains constant. However, if both speakers are turned on at the same time, the microphone will pick up the sum of their signals, i.e. As(r1, r2) = A1(r1) + A2(r2). Let's assume that the speakers are driven with a common reference phase, so that φ1 = φ2. In this case, if r1 = r2, that is, if the distance from speaker 1 to the microphone is the same as the distance from speaker 2 to the microphone, constructive interference occurs and the microphone measures twice the amplitude as it would from only one speaker. This is also true if the difference between r1 and r2 is an even multiple of λ/2. On the other hand, if the difference between r1 and r2 is an odd multiple of λ/2, destructive interference occurs and the microphone will pick up hardly any signal. Therefore, simply adding an additional speaker does not always increase the volume of sound everywhere. In some places the additional speaker will actually reduce the volume of sound!

Speed of sound /frequency= Lambda

Procedure

1. Leave the reference microphone in the same position the calibrating Visual Analyzer. 2. Position the signal microphone at a distance of 4-6 cm facing the speaker.
3. Turn the Waveform Generator ON by clicking the Wave On button.
4. Configure the oscilloscope display, as needed, to show the waveforms of both microphones.
5. Move the signal microphone away from the speaker until the signals on the oscilloscope are in-phase.
Record the distance as well as the amplitude of the wave on Ch B (signal microphone) in Table 3.
6. Repeat step 5 until you have at least five measurements at different distances, each of which satisfy the in-phase condition.
7. Repeat steps 1 - 6 with the Waveform Generator set to a 4 kHz Sine wave.

Data Collection:

2 kHz	**In-Phase Distance (cm)**	**Amplitude (Vpp)**	**λ (m)**	**v (m/s)**
1	**0**	**1**		
2	**17**	**0.2**	**.17**	**340**
3	**34**	**0.15**	**.17**	**340**

4	**51**	**0.11**	**.17**	**340**
5	**71**	**0.2**	**.2**	**400**
Average values for λ and v :			**.1775**	**355**

4 kHz	In-Phase Distance (cm)	Amplitude (V)	λ (m)	v (m/s)
1	**0**	**1**		
2	**20**	**0.2**	**.2**	**800**
3	**35**	**0.18**	**.15**	**600**
4	**52**	**0.15**	**.17**	**680**
5	**71**	**0.14**	**.19**	**760**
Average values for λ and v :			**.1775**	**710**

Plot/Image:

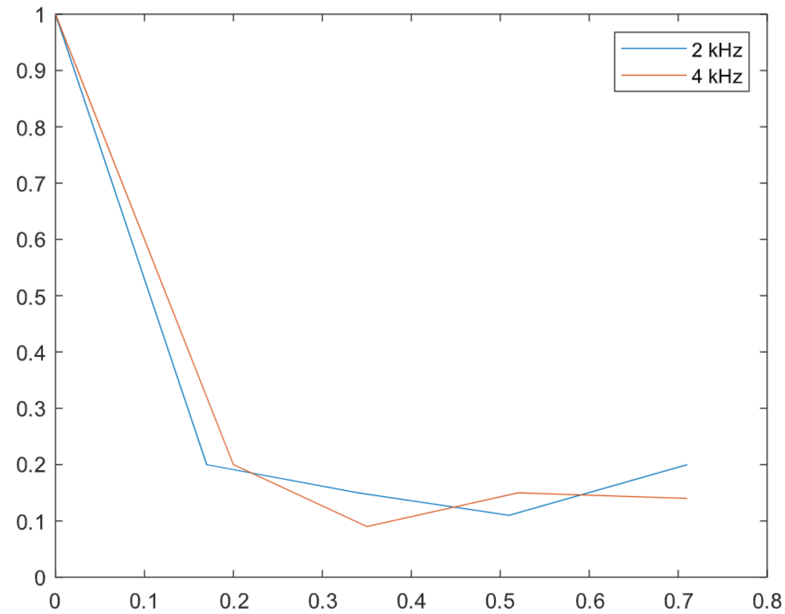


Figure: A plot of distance vs. amplitude from table 1

1.2 EXPERIMENT (Characteristic of sound)

Objective:

In this experiment we will measure the speed of sound in air by determining the propagation time of sound waves over a known distance. In order to make these measurements you have to be able to clearly identify the start of the received signal. To accommodate this we will use a matched filter detection technique. The matched filter is an important topic in signal processing. It brings together the concepts of filtering, random sequences and convolution of Linear Time Invariant (LTI) systems.

Set Up:

Setup for the experiment is similar to Experiment 3.1, see Figure 2:

- Connect the desktop speakers to the audio output port on the computer.
- Connect the stereo Y-adapter to the audio input port on the computer.
- Connect the reference microphone to the gold Y-adapter port using a stereo-mono adapter.
- Connect the signal microphone to the aluminum Y-adapter port using a stereo-mono adapter.
- Open the Visual Analyzer program.
- Configuring the Waveform Generator:

– Open the Waveform Generator window by clicking the Wave button.
– Disable Left (A) Channel by unchecking the Enable checkbox.
– Set the Wave Function for channel B to “Sweep (Sine)”.
– On the Sweep tab, enter a starting frequency in the From textbox and an ending frequency in the To textbox. Enter “1” in the Time to Sweep textbox. (Note: Don’t forget to record your sweep frequencies in your logbook.)
– Close the Waveform Generator window by clicking the X button.

- Configuring the Oscilloscope:

– Open the Settings window by clicking the Settings button.
– On the Main tab, set Channel(s) to “A and B”.
– On the Scope tab, set Y-axis to “Percent”.

– On the Calibrate tab, disable the calibration by unchecking the Apply Calibration Settings checkbox.
– Close the Settings window by clicking the OK button.
– In the Oscilloscope window, check DC removal and Values for both channels.

Theory:

A classic example of matched filtering is pulsed radar, where a train of short pulses is sent out by the transmitter and the receiver has to detect echoes of similar shape reflected by objects at a distance. The transmitted waveform is known in advance, and is not the 7 ECE332 Lab 1: Acoustic Waves important feature; what matters is the detection of the echoes, which are often weak and contain random noise. The matched filter optimally discriminates against certain types of noise in the received waveform. The impulse response h(t) of the matched filter is a reflected, or time reversed, version of the transmitted signal s(t), i.e. h(t) = s(−t). The terminology "matched filter" refers to the fact that the impulse response of the radar receiver is "matched" to the transmitted signal. To estimate the time delay from the matched filter we evaluated the convolution, as shown in (4), where r(t) is the received signal.

$$y(t) = r(t) * h(t)$$

Since the impulse response and frequency response of a LTI filter are related by a Fourier transform pair, it follows that the frequency response of the matched filter is (5), where we have made use of the Reversal and Convolution properties, as shown in Table 2.

$$Y(\omega) = R(\omega)H^*(\omega)$$

Property	**Time Domain**	**Frequency Domain**
Reversal	$s(-t)$	$Y(-\omega)$ or $Y^*(\omega)$
Convolution in t	$s_1(t) * s_2(t)$	$Y_1(\omega)Y_2(\omega)$

Table 2: Properties of the Fourier Transform

Procedure:

1. Position the reference microphone as close to the speaker as possible while maintaining a "clean" signal on the oscilloscope display.
2. Position the signal microphone at a distance of 50 cm facing the speaker.
3. Turn the Waveform Generator ON by clicking the Wave On button.
4. Capture the time domain signals by clicking the Capture Scope button.
5. Save the captured time domain signals as text files.
6. Capture and save two more time domain waveforms.
7. Repeat steps 1 - 6 for separation distances of 75 and 100 cm.

Data Collection:

Distance(cm)	T1 ms	T2 ms	T3 ms	Avg t ms	V m/s
50	.001367	.001318	.001318	.001334	361.27
75	.002124	.0021	.0021	.002108	355.78
100	.002808	.002808	.002856	.002824	354.10
Overall average					357.05

Plot/Image:

Image 1 (50)

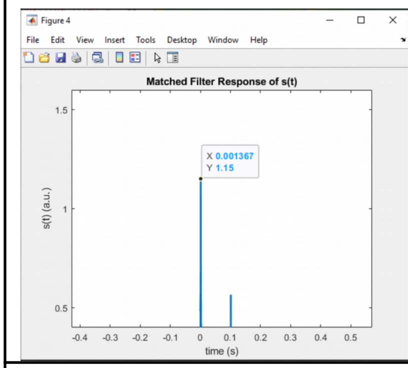


Image 2 (50)

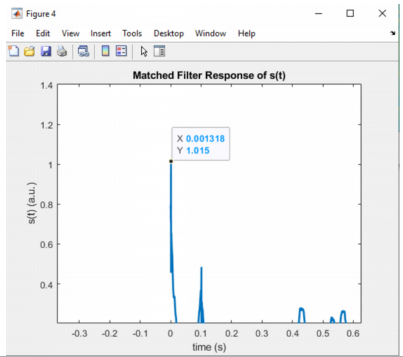


Image 3 (50)

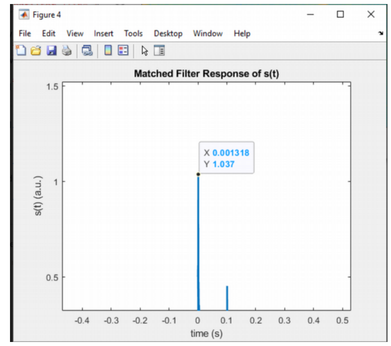


Image 4 (75)

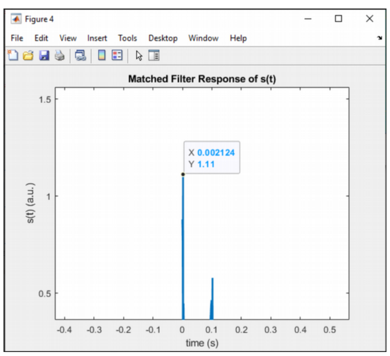


Image 5 (75)

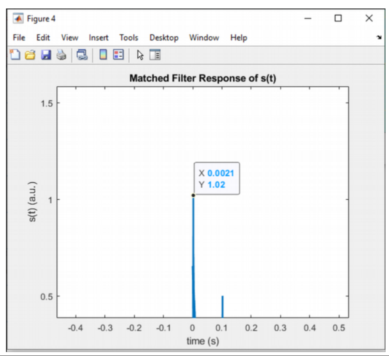


Image 6 (75)

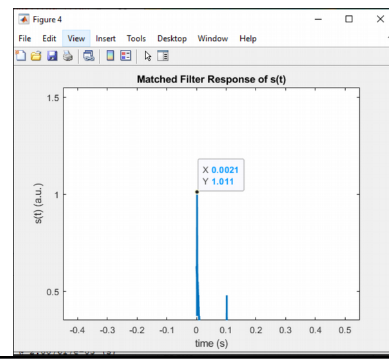


Image 7 (100)

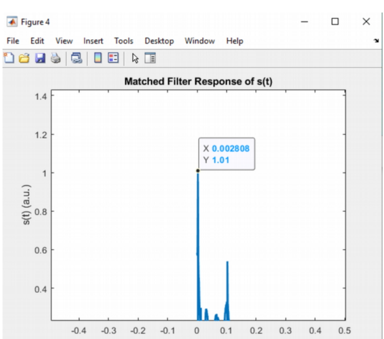


Image 8 (100)

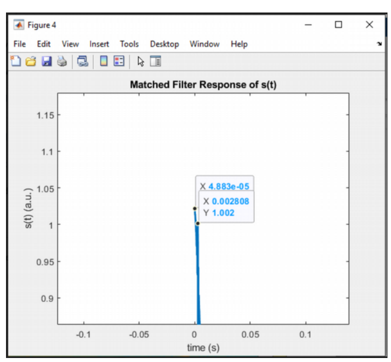
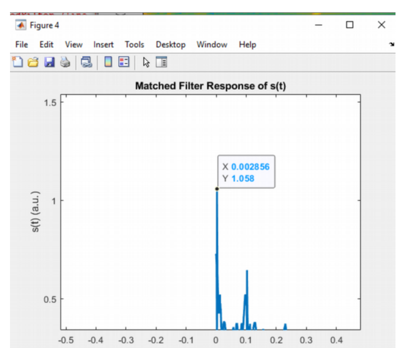


Image 9 (100)



Experiment 3.1

Questions

1. What are the main sources of uncertainty in your measurements? For each, indicate *whether the source is systematic or random. (5 pts).*

The uncertainty in our measurements was mostly from human error. It is hard to look at the value for the exact number. We measure the source in a systematic way, but still only have the appromine value for measuring with our hands and eyes.

2. Explain the significance of the in-phase condition and how it is used to find wavelength and velocity. (5 pts)

When we increase the distance between Mic 1 and Mic 2. The wavelength of the source wave compared to the resulting wavelength moves out of phase. When the wavelength is in phase, it means that this is the length of the lambda.

3. Based on your plots, discuss how well does the calculated amplitudes from Equation (1) fit the experimental data? Discuss possible reasons for any discrepancies, if they exist. (5 pts)

The graph and calculations looked like it was pretty close. We can not be perfect because of human error. The graph is distance over amplitude.

Experiment 3.2

Questions

1. In this experiment, we have claimed to measure the speed of sound in air. To do this, we have used an audible frequency sweep. Why can we claim that the speed of sound will be the same over the bandwidth of the sweep? Is this assertion justified based on your results from experiment 2? Hint: Consider how sound (longitudinal pressure waves) propagate in a medium. (5 pts)

This is because as the bandwidth of the waves changes through the medium (in this case air or free space) the time changes as well. One period is inversely proportional to frequency. So this explains that the change in bandwidth(frequency) will change the period. Therefore, changing the wavelength accordingly. The table/results show this.

2. When computing the delay time between the transmitted and received signal, you did not take into account the added delay due the coax cable. Explain why you can overlook this delay? Hint: Consider the possibility that the coax cables may be of differing lengths. (5 pts)

We can overlook the delay in the coax cable because the acoustic sound traveling at the speed of sound, is translated to electrical waves that are traveling at the speed of light. The speed of light is significantly faster than the speed of sound. So, the delay is almost zero.

3. Analytically compare the accepted value (that you looked up for the pre-lab assignment) to the value you computed in this experiment and to the values you calculated in Experiment 3.1. Comment on your results and explain what may cause discrepancies, if they exist. (5 pts)

There were several errors in this experiment. For example, there were several sound waves bouncing off other objects and moving in different directions. There was human error in conducting the experiment when recording and transmitting the sound waves. As well as software errors in transmitting the signal and receiving the signal. The initial values did match for experiment 3.1. However, the values did match for experiment 3.2.

Conclusion/Discussions

The experimental portion of the lab was easy. Everything was straightforward and clear. We also have two TA's to help us complete the lab. We finished the lab early and decided to complete the calculations and matlab portions on our own. One of the TA's said that he would send us on what we need to do for the second portion of the lab. He did email us the matching filter and we were able to complete the matching filter graphs.

We had some issues trying to figure out what the correct equations to use to find the velocity and Lambda. After, Spending some time working on the equations, we finally just googled the equation to get the same result we were expecting.

Overall, this lab was a good learning experience. We spent some time relearning some stuff from our physics days, in order to complete the Lab. It turns out that the values that we got were really close to the speed of sound in free space.