Portland State

UNIVERSITY

DEPARTMENT OF
ELECTRICAL AND COMPUTER ENGINEERING

ECE332

LAB 1: ACOUSTIC WAVES

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Contents

Objective

In this lab you will experimentally explore some characteristic properties of acoustic waves.

Concepts Covered

- Propagation of acoustic waves in air

- Interference of traveling waves

- Spreading loss of spherical waves

- Matched filter detection

1 Pre-Lab Assignment

- Review Sections 1.3-1.5 of Ulaby[1].
- Find some standard value for the speed of sound in air. Record that value and its source in your logbook.

2 Introduction: Theory of Acoustic Waves

2.1 Spreading Factor of Spherical Waves

From ECE331, you should be familiar with the propagation of plane 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 (1).

$$A_{sph}(r) = \frac{A_0}{r}e^{-jkr} \tag{1}$$

A spherical wave is three-dimensional and has spherical wavefronts 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 wavefront. The wave amplitude must therefore decrease with the spherical waves 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 lossy medium. In a lossy medium $k = k_R - jk_I$, would be complex and the attenuation loss would be accounted for by k_I. 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

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

where D is the largest linear dimension of the source and λ is the wavelength.

2.2 Interference in 3-D

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
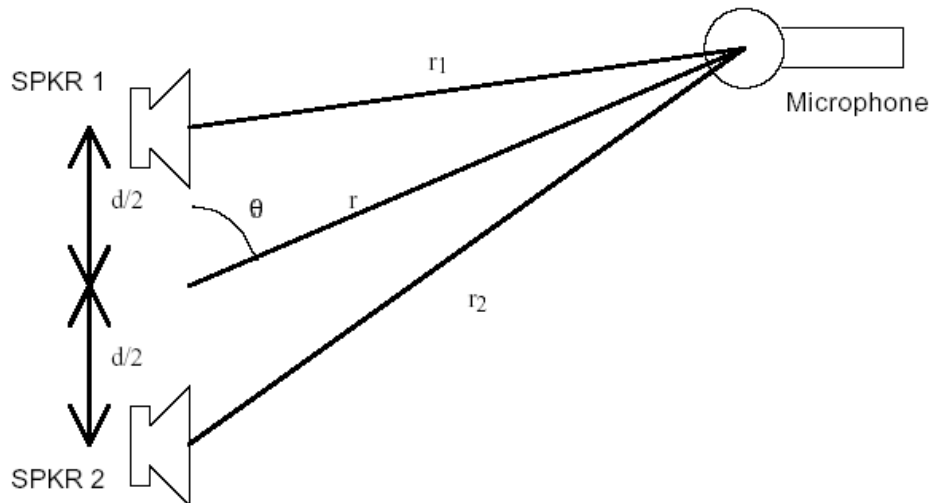
Figure 1: Geometry of an Acoustic Experiment.

pressure wave can be described as,

$$A_1(r_1) = \frac{A_{1,0}}{r_1} \exp^{-jkr_1 + \phi_1} \tag{3a}$$

$$A_2(r_2) = \frac{A_{2,0}}{r_2} \exp^{-jkr_2 + \phi_2} \tag{3b}$$

where k is the propagation constant, r_1 and r_2 are the distances between the speakers and the observation point (microphone) and A_1 and A_2 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. $A_s(r_1, r_2) = A_1(r_1) + A_2(r_2)$.

Let's assume that the speakers are driven with a common reference phase, so that $\phi_1 = \phi_2$. In this case, if $r_1 = r_2$, 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 $\lambda/2$. On the other hand, if the difference between r_1 and r_2 is an odd multiple of $\lambda/2$, 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!

3 Experiments

Desktop Speakers	x2 - Desktop Microphones
Meter Stick	Visual Analyzer 2011
Stereo Y-adapter	x2 - Stereo to Mono Adapters

Table 1: Experiment Equipment List

3.1 Characteristics of Sound Waves

Introduction

In this experiment, you will measure 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 wavefront, the wave amplitude must therefore decrease with the spherical waves radius in order to satisfy the law of conservation of energy.
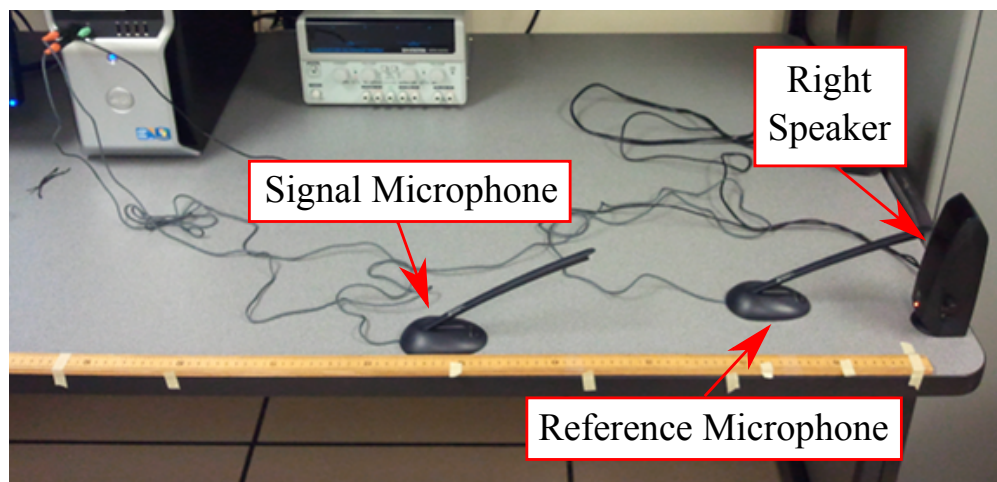
Setup

Figure 2: Experiment 1 setup.

Setup the experiment as shown in 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 frequency for channel B to one of the frequencies of 1900, 2000, 2100, 2200 (Hz) and click the *Apply* button (each group is better to have different frequency than other groups,why?).
  - 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. Try to have the microphones input in same height as output of Speaker.
  - Turn the volume control on the side of the speaker to $\approx 50\%$.
  - 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 Windows™ 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 V_{pp} voltage of ≈ 1 V, see Figure 3.

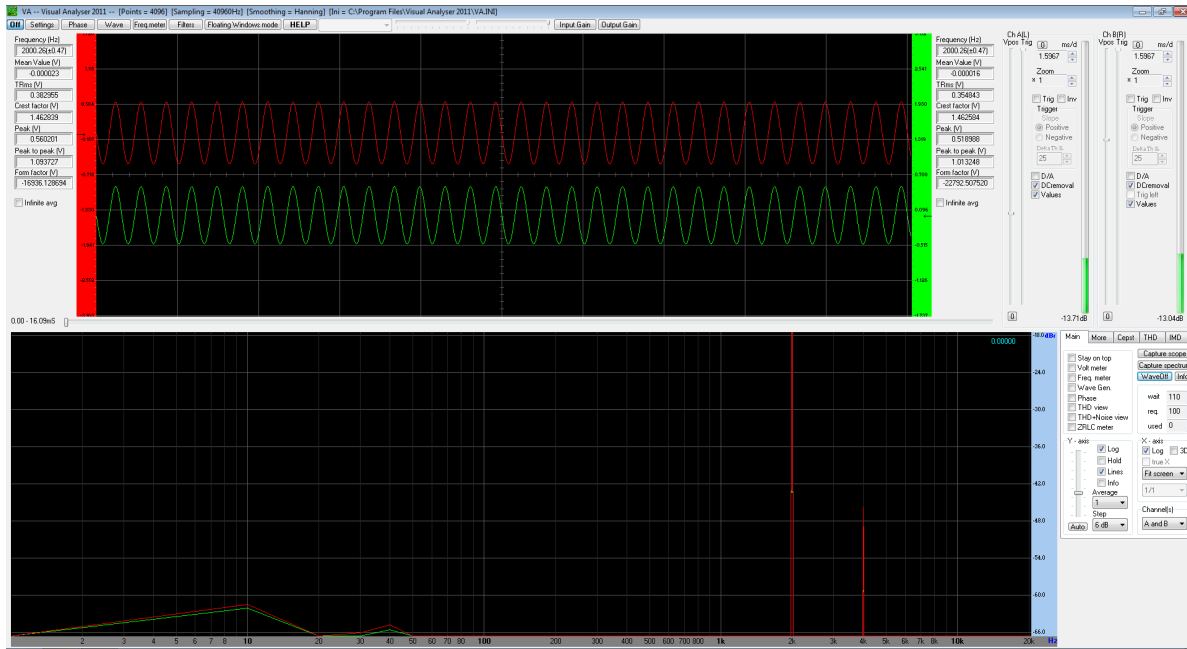- Close the **Settings** window by *clicking* the **OK** button.

Figure 3: Visual Analyzer after calibration of levels.

Procedure

1. Leave the reference microphone in the same position the that was used for 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.

Collected Data

- Measured values:

  - For 2 kHz and 4 kHz sinusoidal signals, five contiguous signal microphone distances in which the reference and signal are in-phase.
  - Amplitude of the measured signal (in volts) at each in-phase position.

- Calculate and record in your logbook the theoretic wavelength and speed of sound for 2 kHz and 4 kHz frequencies.

- Calculate the average λ and v from the data points collected in Table 3.

- Use Matlab to plot your measured in-phase distance vs amplitude for each frequency. On the same plot include the amplitude from Equation (1), i.e. $|A_s(r)| = \frac{A_0}{r}$.

3.2   Speed of Sound

Introduction

In this experiment you 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.

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 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.

$$y(t) = r(t) * h(t) \tag{4}$$

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.

$$Y(\omega) = R(\omega)H^*(\omega) \tag{5}$$

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

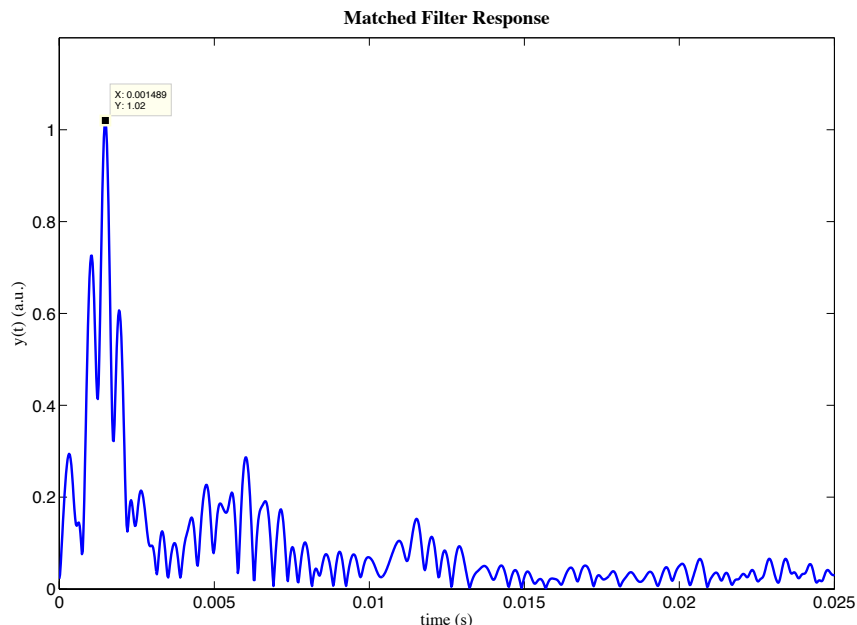Table 2: Properties of the Fourier Transform

Figure 4: First echo of the matched filter response.

Setup

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.

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- 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 "Volt".
  - 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.

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.

Collected Data

1. Measured values:

   - Three time domain data files for each separation distance.

2. Using the matched filter detection technique, write a Matlab function that reads your time domain data files and computes the time that the transmitted signal reached the signal microphone. (Hint: Matlab has built-in functions for computing the Fast Fourier Transform (FFT), Inverse Fast Fourier Transform (IFFT) and spectrogram)

3. For each time domain data file, record the Δt time that your matched filter function computed in Table 4.

4. For each separation distance, compute and record the average Δt value in Table 4.

5. Using the average Δt values for each separation distance, compute and record in Table 4 the measured speed of sound in air.

2 kHz	In-phase distance (m)	Amplitude (V)	λ (m)	v (m/s)
1				
2				
3				
4				
5				
Average values for λ and v :				
4 kHz				
1				
2				
3				
4				
5				
Average values for λ and v :				

Table 3: Experiment 3.1 - Measured in-phase positions and amplitudes for 2 kHz and 4 kHz sinusoidal signals.

Distance (cm)	Δt_1 (ms)	Δt_2 (ms)	Δt_3 (ms)	Avg. Δt (ms)	v (m/s)
50					
75					
100					
				Overall Average	

Table 4: Experiment 3.2 - Matched filter detection times.

Lab Report Questions

Experiment 3.1

Items to Turn in

- Table 3 (10 pts)

- Matlab plots (15 pts)

Questions

1. What are the main sources of uncertainty in your measurements? (5 pts)

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

3. Based on your plots, explain how well the calculated amplitudes from Equation (1) fit the experimental data? Speculate about possible reasons for any discrepancy. (5 pts)

Experiment 3.1

Items to Turn in

- Table 4 (10 pts)

- Matlab plots of your matched filter response for each separation distance (15 pts)

- Value and source of reference for the excepted speed of sound in air (5 pts)

Questions

1. In this experiment, we have claimed to measure the speed of sound in air. To do this, we have used a 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 propagates in air.) (5 pts)

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 over look this delay? (5 pts)

3. Empirically 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, it they exist. (5 pts)

References

[1] F. Ulaby, E. Michielssen, and U. Ravaioli, *Fundamentals of Applied Electromagnetics*, 6th ed.   Prentice Hall, 2010.

[2] S. Haykin and B. V. Veen, *Signals and Systems*, 2nd ed.   Wiley, 2003.