DESIGN OF ACOUSTIC IMPEDANCE ANALYZER

By

ERIC F. BICKFORD

A thesis submitted to the

Graduate School-New Brunswick

Rutgers, The State University of New Jersey

In partial fulfillment of the requirements

For the degree of

Master of Science

Graduate Program in Mechanical and Aerospace Engineering

Written under the direction of

Professor Kimberly Cook-Chennault

and approved by

____

____

____

____

New Brunswick, New Jersey

May, 2015

ABSTRACT OF THESIS

DESIGN OF ACOUSTIC IMPEDANCE ANALYZER

by

ERIC F. BICKFORD

Thesis Director:

Professor Kimberly Cook-Chennault

Research into materials that are able to absorb low frequency sound waves has become an increasing field of study because of noise pollution in our daily lives. To combat noise pollution created by airplanes, researches have been looking into new light and thin materials to solve the problem. To test these materials a device known as an Acoustic Impedance Tube needs to be used. These devices can either be purchased from a retailer for thousands of dollars or built in house for a fraction of the cost. Using ASTM standards, guidelines for building a tube are presented, but there is not much information into what the tubes results should look like for specific materials. There is also not much information on how the output will change when certain design parameters are modified. The work presented in this paper is not just a guide to help others build an impedance tube but it also can be used to check their own tubes results.

Acknowledgements

I would first like to acknowledge my advisor Prof. Kimberly Cook-Chennault for her support in my research and career. I appreciate her useful comments, remarks and engagement through this entire process. Secondly I would like to thank Professor Haim Baruh and Hae Chang Gea for serving as my committee members.

I would next like to thank my friends and lab mates, Udhay Sundar, Andrew Tang, Sankha Banerjee, Wanlin Du, and Tom Hansen for their support and all of their help that they provided in my research. I am thankful for their supportive guidance, invaluably constructive criticism and friendly advice during this process. Finally I would like to thank other undergraduate and graduate students I have work with over the years.

Eric Bickford

Rutgers University

May 2015

Table of Contents

Abstract of the Thesis .... Error! Bookmark not defined.

Acknowledgements iii

List of Figures vii

List of Tables ix

Chapter 1. Introduction and Background 1

1.1. Research Motivation *1*

1.2. Definitions *5*

- *1.2.1. Acoustic Impedance* *5*
- *1.2.2. Sound Transmission Loss* *6*
- *1.2.3. Impedance Tube* *7*

1.3. Overview of the Thesis *8*

Chapter 2. Background 9

2.1. Standing Wave Ratio (SWR) Method [25, 28] *10*

2.2. Two Microphone Transfer Function Method Theory *13*

2.3. Two Microphone Transfer Function Method ASTM *14*

- *2.3.1. Tube Specifications* *15*
- *2.3.2. Microphones* *18*
- *2.3.3. Signal, Speaker, and Processing Equipment* *20*
- *2.3.4. Other Requirements* *21*
- *2.3.5. Calculations of Acoustic Properties* *22*

2.4. Two Microphone Transfer Function Method ISO [29] *26*

- *2.4.1. Tube Specifications* *26*
- *2.4.2. Signal, Speaker, and Processing Equipment* *28*
- *2.4.3. Other Requirements* *29*
- *2.4.4. One Microphone Technique* *30*

2.5. Four Microphone Method *31*

- *2.5.1. Tube Specifications* *33*

2.5.2. Other Requirements *34*
2.5.3. Theory and Calculations of Acoustic Properties *35*
2.6. Three Microphone Method *41*
2.7. Comparison of Impedance Tube System Designs *45*
Chapter 3. Description of Tube Design 50
3.1. Tube Material *51*
3.2. Tube Parameter Design *52*
3.3. Microphones *53*
3.4. Frequency Range *54*
3.5. Microphone Holders *56*
3.6. Sample Holder and Back Plate *57*
3.7. Loud Speaker *58*
3.8. Signal Generator and Signal Processing Equipment *58*
3.9. Four Microphone Design *59*
3.10. Three Microphone Design *62*
Chapter 4. Design of Experiments 65
4.1. Lower Frequency Limit of Standards *68*
4.2. Tube Material and Length *72*
4.3. Position of Samples *75*
4.4. Signal, Equalizer, and Speaker *77*
4.5. Run Time and Speaker Volume *81*
Chapter 5. Tube Comparison Results 85
Chapter 6. Conclusions and Future Work 103
6.1. Conclusions *103*
6.2. Future Work *105*
References 107

Appendices **110**

List of Figures

Figure 2.1 SWR Impedance Tube with a standing wave created by the incident and reflective Pressure waves [27]....11
Figure 2.2 Two Microphone Impedance Tube Diagram with Incident and Reflective Waves[28]....14
Figure 2.3 Four Microphone Impedance Tube Diagram[29]....33
Figure 2.4 Three Microphone Impedance Tube [38]....42
Figure 3.1 Two Microphone Impedance Tube Diagram with Dimensions[28]....50
Figure 3.2 3D Printed Microphone Holders Attached to Multiple Microphone Positions....57
Figure 3.3 Four microphone tube with open termination and microphones positioned in locations 1 and 2 of upward section....60
Figure 3.4 Anechoic termination for four microphone tube lined with fiberglass....61
Figure 3.5 Three Microphone Tube....63
Figure 3.6 Three Microphone Holder and Cap....64
Table 3.1 Tube Dimensions Based off of Available PVC Sizes....52
Table 3.2 Frequency Dependance on Microphone Spacing....54
Figure 4.1 Impedance Analyzer System Diagram....66
Figure 4.2 Multi-Position Test with Polyurethane Sample on Logarithmic Scale....69
Figure 4.3 a,b Low Frequency Multi-Position Test with Polyurethane Sample on Logarithmic Scale....70
Figure 4.4 High Frequency Multi-Position Test with Polyurethane Sample on Logarithmic Scale....71
Figure 4.5 Tube Material Effect on Absorption Coefficient for Polyurethane Foam from 200-5000Hz....73
Figure 4.6 a,b Tube Material Effect on Absorption Coefficient for Polyurethane, Low and High Frequencies....74
Figure 4.7 Sample Distance Effect on Absorption Coefficient for Polyurethane Foam....75
Figure 4.8 a,b Sample Distance Effect on Absorption Coefficient for Pine....76
Figure 4.9 Signal Type Effect on Absorption Coefficient for Polyurethane....78
Figure 4.10 Equalizers Effect on Absorption Coefficient for Polyurethane....79
Figure 4.11 Speakers Effect on Absorption Coefficient for Polyurethane....80
Figure 4.12 Run Time Effect on Absorption Coefficient for Polyurethane....82
Figure 4.13 Speaker Volumes Effect on Absorption Coefficient for Polyurethane....83
Figure 5.1 Samples Tested (from left to right, Foam Matrix, Polyurethane, Melamine, Green Foam, Cotton, Fiberglass, Denim, Pine, Acrylic)....86
Figure 5.2 Two Microphone Sample Test....87
Figure 5.3 Melamine Comparison [23]....88
Figure 5.4 Three Different Samples of Polyurethane Foam....90
Figure 5.5 Specimen Test with Four Microphone Tube....91
Figure 5.6 Transmission Loss of Four Microphone Tube....93

Figure 5.7 Absorption Coefficient using Three Microphone Tube 95
Figure 5.8 Transmission Loss of Samples in Three Microphone Tube 97
Figure 5.9 Absorption Coefficient of Denim Sample for All Three Tubes 99
Figure 5.10 Transmission Loss of Samples in Three and Four Microphone Tubes 101

List of Tables

Table 2- 1 Recommended Maximum Frequency Based on Microphone Diameter [28] 18
Table 2- 2 Measurement Configurations Based off Number of Microphones and Channels [29] 36
Table 3.1 Tube Dimensions Based off of Available PVC Sizes 52
Table 3.2 Frequency Dependance on Microphone Spacing 54
Table 4.1 Design of Experiment Test Options 67
Table 4.2 Design of Experiment List of Runs 67
Table 5.1 Tested Materials and Size 85

Chapter 1.
Introduction and Background

1.1. Research Motivation

Noise pollution caused by automobiles and airplanes has become a big topic of discussion over the last few decades [1-5]. Living near large highways and airports is an annoyance to the community and could also have a negative effect on people's health. Research has shown that long exposer to noise can cause mental health problems such as depression and stress related illnesses, such as high blood pressure, speech interference, hearing loss, sleep disruption, and loss of productivity [4]. Noise pollution has also been found to affect children's reading comprehension during early childhood development [1]. To combat noise pollution, noise attenuating and eliminating materials have been developed to absorb unwanted and harmful noise. For automobiles, an easy solution has been to build sound absorbing walls along major highways that are directly adjacent to houses. However, aircraft noise continues to be an area of growing concern.

The Federal Aviation Administration (FAA) has developed individual noise control regulations for each airport inside the United States. The regulations depend on the airport size and the location of the surrounding community. Noise field maps have been created for each airport to quantify the noise and then attempt to decrease it, by directing air traffic away from densely populated areas [6, 7]. The FAA is constantly revising and improving on these regulations by limiting aircraft engine noise, as well as soundproofing schools in the vicinity of the airport [7]. For example, the Newark Liberty International Airport has a noise limit of 112 PNdB (perceived noise level in decibels) for aircraft during take-off [2]. In addition, programs have also been implemented to insulate public schools exposed to aircraft noise. Recently the FAA ruled that Stage 2 aircraft could no longer be flown inside the contiguous United States [8]. The "Stage" number of the aircraft is used to describe the aircraft's noise level. The government has passed federal regulations values to define the maximum noise level for Stage 2, Stage 3, etc. [9].

To comply with higher Stage requirements, researchers have focused on reducing the noise generated from aircraft engines by lining the engines with highly acoustic absorbing material [10-12]. The human ear has the ability to discern acoustic tones from frequencies as low as 16 Hz to as high as 20 kHz [13]. An aircraft's engine can produce noise within this frequency range, although noise contributing to so-called noise pollution is typically considered to be between 50 Hz and 5 kHz, which coincides with the region where human hearing is most sensitive [14, 15]. Frequencies above 5 kHz are usually ignored because of high attenuation over long distances in the Earth's atmosphere. The pressure of a sound wave in a medium can be described by the Equation 1.1.

$$P = P_0 \, e^{-\alpha x} \tag{1.1}$$

Equation 1.2 describes the one dimensional pressure at a point along a sound wave with a frequency of ω.

$$p = P \, e^{-i\omega t}, \tag{1.2}$$

where α is the attenuation coefficient of the sound wave in one dimension. Since the attenuation coefficient and distance are always positive, the pressure will decrease as the distance increases. The attenuation coefficient is directly related to the frequency of the sound wave. When the medium is air, the attenuation coefficient and frequency are positively correlated. This is why frequencies above 5 kHz can be ignored when studying noise pollution caused by aircraft. At lower frequencies, the attenuation coefficient is very small, which in turn means that it will travel a long distance before being attenuated. This presents a dilemma for the development of sound attenuation materials where material thickness is limited, such as in aircraft.

Materials that are good at absorbing low frequencies usually need to be very thick or denser than a high frequency absorbing material. The materials with large attenuation constants are usually denser and less dense materials have to be made much thicker to absorb the same amount of sound. Use of thick or extremely dense materials for noise dampening in aircraft causes problems. Thick sound absorbing panels can take up considerably large amounts of room, which is not available in the confined spaces of an aircraft. Dense materials can be heavy, and use of them in an aircraft would cause the aircraft to weigh more, thereby increasing the amount of fuel use.

Currently aircraft contain a combination of honeycomb structured materials along with foams to reduce sound from entering and exiting the cabin. The honeycomb structures are components of the cabin floor and fuselage. Honeycomb structures are used because of their high strength to weight ratio and high sound transmission loss. A honeycomb structure of the same weight and double the thickness of a single plate can have 12 times greater stiffness, and 6 times greater strength and have a sound transmission loss of around 20dB at 1000Hz [16-18]. This includes a high absorbing material like mineral wool inside the honeycomb structure. Honeycomb structured panels are typically acoustically efficient between 1000-2000 Hz, but can be designed for lower frequencies by increasing the stiffness of the panels [17, 19]. Panels are usually developed to be less than an inch thick to fit inside the fuselage of an aircraft, but the thickness depends on the specific aircraft [17, 20]. In addition to use of honeycomb structures, sound absorbing foams are also used in aircraft. For example, the new Boeing Dreamliner 787 uses Basotect foam made of a melamine resin, has a density of 9 g/L and has adequate sound deadening properties [21-24].

Noise pollution in the commercial aerospace industry has sparked a significant amount of research into developing materials that have a small form factor and weight. NASA, in particular, is interested in materials that minimize transmission of noise into the cabin of the aircraft in addition to materials that reduce the transmission of engine noise to the surrounding environment.

1.2. Definitions

1.2.1. Acoustic Impedance

The acoustic impedance can be used to calculate how much sound is absorbed and reflected by a material. Acoustic impedance is the ratio between the average pressure of sound, per unit area, to the volume velocity. The acoustic impedance is described by Equation 1.3 for one dimension. The phase between the pressure and velocity determines the magnitude of the impedance. For example, a large pressure with a corresponding large velocity will have a lower impedance than the same pressure with a lower velocity.

$$Z = \frac{p}{U} \tag{1.3}$$

Sound waves can be classified into several categories: plane waves, cylindrical, and spherical. The acoustic impedance of a plane wave is constant for a specific medium. This is usually referred to as the characteristic acoustic impedance. Plane waves are sound waves which have a constant pressure in each plane that is perpendicular to the sound wave's velocity. In general, the pressure and velocity will be slightly out of phase, which causes the acoustic impedance to be a function of frequency. At the boundary between a medium and a material, the acoustic impedance describes properties of the material, e.g. the normal specific acoustic impedance. The impedance can be separated into the real and imaginary components. The resistive, or real, portion of the impedance describes the work that is done on the medium since pressure and flow are in phase with each other. The reactive, or imaginary, portion of the impedance transfers no energy since the pressure and flow are out of phase. This idea is analogous to the electrical impedance of circuits with Alternating Current (AC). Similar to electrical impedance, acoustic impedance is highly dependent on the frequency of the sound wave.

1.2.2. Sound Transmission Loss

The sound transmission loss (STL) is the decrease in sound energy that a sound wave experiences when traveling through a material. This value is important in determining how well sound is isolated when traveling through the material. The transmission loss is described by Equation 1.4, where p_u is the incident pressure in the portion of the tube that is in front the material, and p_d is the incident pressure in the portion of the tube after the material.

$$STL = 20\log\frac{p_u}{p_d} \tag{1.4}$$

It can be seen that the less sound pressure that makes it through the material or smaller p_d, the higher the transmission loss. STL is different from the absorption coefficient because a material that is good at preventing sound from traveling through it does not necessarily absorb the sound. Examples of materials with high absorption coefficients are foam, wool, fiber, etc. Materials such as these are usually porous. Materials that have high STL values typically act as barriers of sound and are generally made from vinyl, impermeable films, and other non-porous materials. High acoustic dampening materials that can be used for the aforementioned applications generally require high absorption coefficient and STL values.

1.2.3. Impedance Tube

Impedance tubes are used to measure the ability of a material to absorb or eliminate sound. Specifically, impedance tubes are used to measure the acoustic impedance and sound transmission loss of a material. Acoustic impedance tubes employ a sound source and specially placed microphones to measure the acoustic properties of a material that is placed inside of the tube. Like other laboratory equipment, it can be purchased from a commercial company, or it can be built in-house. Since there is a very small market for impedance tubes, the price to purchase one is greater than ten to twenty thousand dollars[25, 26]. A custom built acoustic impedance tube can be a fraction of the cost of the commercial grade tube and provide similar results. There are several types of acoustic impedance tubes currently on the market, and the properties of the tubes vary depending on the acoustic properties required. There are several standards used for the design and assessment of impedance measure tubes; ASTM C384, ASTM E1050, ASTM E2611, ISO 10534-1, and ISO 10534-2 [27-31]. These standards provide guidelines for the design of the tube including ranges for relative dimensions, microphone spacing, type of speaker used, tube material, and type of impedance tube, which influence the measured data. Though these standards are used throughout the literature in design of impedance tubes for measurements of acoustic properties, little work is available on the influence of variation in these parameters on the measured data. This works seeks to understand the influence of parameter variability, (within the acceptable ranges of the standards) on measured values from several impedance tubes.

1.3. Overview of the Thesis

The thesis is organized in the following manner. Chapter 2 deals with the theory behind an impedance tube and the standards that can be used to build one. Four different types of tubes will be presented and compared. In Chapter 3 the design and construction of the impedance tubes are presented. Chapter 4 shows the results of a design of experiments for the two microphone tube along with an economic assessment. In Chapter 6 several different types of materials are tested and the results presented. Finally, in Chapter 6, the conclusion of the thesis is presented.

Chapter 2.
Background

This chapter provides an overview of several types of impedance tubes developed for determining the acoustic properties of materials. The standing wave ratio was one of the first impedance tubes developed, and is described by the American Society of Testing and Materials (ASTM) and International Organization for Standardization (ISO) standards (ASTM C384, ISO 10534–1) [27, 30]. The two microphone impedance tube was later developed to solve some of the limitations of the former tube. The design of the two microphone impedance tube is also described by ASTM and ISO standards (ASTM E1050, ISO 10534–2) [28, 31]. The two microphone impedance tube was later developed to solve some of the limitations of the former tube evaluation technique, had such as only measuring one frequency per test. Measuring the full frequency spectrum would require a multitude of tests. The design of these type of tubes is also described by both the ASTM and ISO standards [28, 31]. The two microphone impedance tube is appropriate for studying acoustic impedance and absorption coefficients. The four microphone impedance tube method was developed to determine the transmission loss of materials. This tube leverages the two microphone design and ASTM standards (ASTM E1050 for this tube are available[29]. A three microphone tube impedance measurement technique has also been developed to replace the four microphone tube when determining transmission loss [32].

2.1. Standing Wave Ratio (SWR) Method [27, 30]

One of the earliest forms of acoustic impedance tube is used to measure properties for the calculation of acoustic properties of materials by measuring the standing wave ratio (SWR). A typical SWR tube is depicted in Figure 2-1. The SWR tube has a speaker on one end of the tube with a test specimen on the other end. The speaker produces a single frequency sinusoidal tone, which forms plane waves as it travels down the tube. This is the incident pressure wave that is designated in Figure 2.1 as P_i. In this figure, the arrow points in the direction that the wave travels. The sound wave travels through the specimen causing some of the sound to be absorbed and some to be reflected. This is the reflective pressure wave that is designated in Figure 2.1 as P_r, where the arrow points in the direction of the speaker. A standing wave is formed by the combination of the traveling incident and reflected waves. This wave is denoted in Figure 2.1 as the line between the two maximum dotted lines. The pressure inside the tube is measured using a microphone probe by moving the microphone probe further inside the tube in an attempt to discover the minima and maxima of the standing wave. Once the maximum and minimum pressure are found, their corresponding values are recorded so that the standing wave ratio can be calculated.

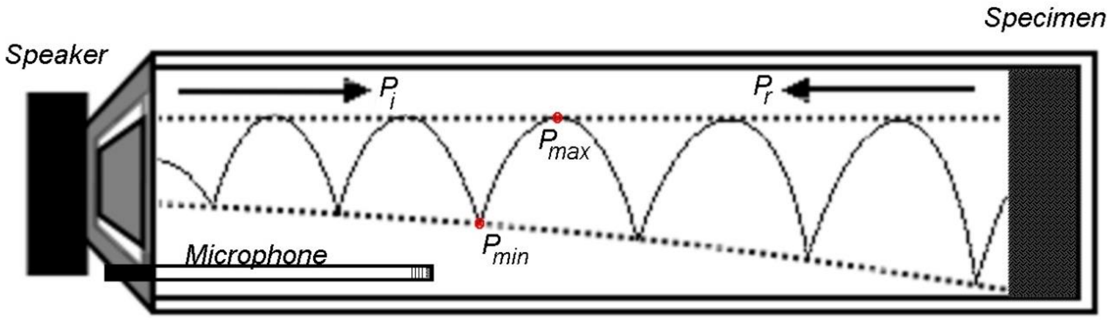


Figure 2.1 SWR Impedance Tube with a standing wave created by the incident and reflective Pressure waves [27]

The maximum points of the standing wave sinusoidal curve (depicted in Figure 2-1) are collinear, with a slope of zero. Conversely, the minimum points, though collinear, lie on a linear line with a slope greater than zero. This decrease in pressure at minimum points can be attributed to the sound attenuation from the walls of the tube. Under ideal conditions, where the tube is constructed from a perfectly reflective material, both minimum and maximum points are parallel with a zero slope. The voltage output of the microphone corresponding to the minimum and maximum pressure is required to calculate the SWR. This ratio is defined by the equation below,

$$SWR = \frac{V_{max}}{V_{min}} = \frac{P_{max}}{P_{min}} = \frac{P_i + P_r}{P_i - P_r}. \tag{2.1}$$

In Equation 2.1, P_i and P_r are the pressures of the incident and reflective waves, respectively. Although the incident and reflective pressure waves cannot be directly measured by the microphone, the sum and difference between them can be. This is done by measuring the maximum and minimum points on the standing wave. Since measured voltage of the microphone is proportional to the pressure by only constants, the direct voltage output can be used. Then, using this ratio, the acoustic reflection coefficient can be calculated. The underlying process described in the equation

$$R = \frac{[SWR - 1]}{[SWR + 1]}. \tag{2.2}$$

Using the reflection coefficient, the absorption coefficient α can be calculated as

$$\alpha = 1 - |R|^2. \tag{2.3}$$

The absorption coefficient is then used to calculate the specific acoustic impedance, Z as

$$Z = (1 + R)(1 - R)\rho c_a, \tag{2.4}$$

where ρ is the density of air and c_a is the speed of sound in air. Using similar techniques, other acoustic properties can be calculated such as the acoustic resistance, reactance, and capacitance. ASTM and ISO have developed standards building this type of impedance tube. There are recommendations for diameter size, length of tube, materials and other specifications on equipment and procedures.

The SWR method provides accurate data for the individual frequencies that are tested. Since each frequency needs to be tested independently, obtaining results for a wide frequency range is very tedious. Testing multiple samples could take an entire day to complete. The two microphone transfer function method was developed to solve this problem. The impedance or absorption coefficient could be calculated for an entire frequency spectrum in only a few minutes[33].

2.2. Two Microphone Transfer Function Method Theory

Like the SWR method, the two microphone method consists of a tube with a speaker at one end and a test sample at the other. Instead of using a microphone probe, two microphones are located along the tube at known distances between each other and the surface of the specimen. Instead of a single sine wave, a noise signal is used. The noise signal has a uniform spectral density which in principle contains an equal amplitude of each frequency. A simple diagram for the impedance tube is shown in Figure 2.2.

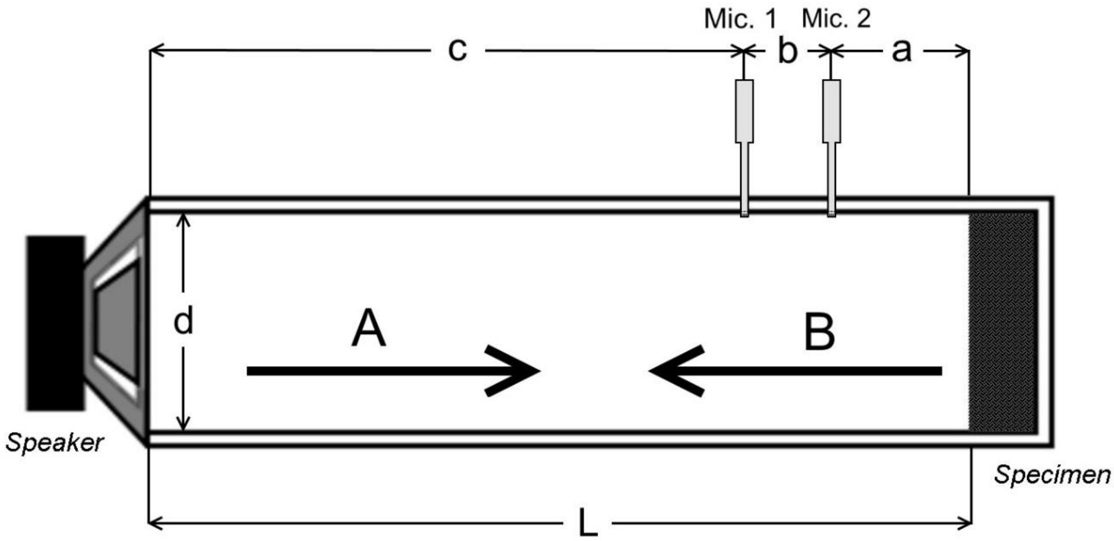


Figure 2.2 Two Microphone Impedance Tube Diagram with Incident and Reflective Waves[28]

Looking at Figure 2-2, there is an incident and reflective wave shown as A and B reflectively. Since the signal is now noise it is not possible to measure a single SWR. Instead, a transfer function between the two microphones can be used to calculate this value. The transfer function is defined as the complex pressure ratio between the pressures at the two microphones. The absorption coefficient and acoustic impedance can then be calculated using the Equations 2.3 and 2.4 from the SWR method. The equations for calculating the transfer function will be shown later in this section.

2.3. Two Microphone Transfer Function Method ASTM

Once the theory of the two microphone transfer function was developed, a tube need to be built to be used in practice. The ASTM has a collection of informative standards regarding the design of a two microphone impedance tube. These standards set specific requirements on how the impedance tube should be built. If a tube is built using these standards then the results between different tubes should provide the same results even if they were built by different individuals. ASTM tested tubes, manufactured by ten different laboratories, with the same test specimen to confirm that each tube yielded the same results. These standards are presented in the next sections.

2.3.1. Tube Specifications

The tube can be constructed out of a variety of different materials. The most common materials that are used are metal, plastic, cement, and wood. The tube can have either a circular or rectangular cross section. For this thesis circular tubes were chosen because a round cross section would make it easier to fabricate samples in our lab. Also, since Polyvinyl chloride (PVC) was the first material of interest, it was easier to be purchased locally with a circular cross section. For these reasons, rectangular cross sections will not be considered further in this report. Equations for rectangular cross sections are provided in the ASTM standards [27, 28]. The inside of the tubes surface should be smooth, nonporous, and clean of dust or any other particles. This will limit the amount of sound attenuation. The tube should be thick enough to limit the transmission through the walls.

The length of tube depends on the distance between microphones, the distance between the speaker and the closest microphone, and the distance between the specimen and the closest microphone. The tube needs to be long enough so that the sound from the speaker can develop into plane waves. The length between the speaker and the first microphone is driven by

$$c > 3d. \tag{2.5}$$

To also ensure that the plane waves are maintained, the diameter of the tube is directly related to the upper frequency limit and the speed of sound. The upper frequency limit is defined as the maximum frequency that accurate results can be obtained from running a test in the impedance tube. This means that the tube diameter should be chosen so that the maximum frequency of interest falls below this value as

$$d < \frac{0.586c_a}{f_u}. \tag{2.6}$$

The spacing between the microphones depends on the upper frequency of interest as

$$b \ll \frac{c_a}{2f_u}. \tag{2.7}$$

This spacing is required because the reflection coefficient cannot be determined when the microphone spacing reaches an integer multiple of the half wavelength [34]. The ASTM standards claim that 80% of the right side of Equation 2.7 is sufficiently low enough for the upper frequency of interest. This would give a value for the spacing of $b = \frac{0.4c}{f_u}$. The microphone spacing also has a direct relation to the minimum frequency that can be measured. The lower frequency limit is shown in

Equation 2.8. Since the microphone spacing effects both the upper and lower frequencies, special care must be taking to ensure that the desired frequency range is obtained. For wide frequency ranges that includes both high and low frequencies, multiple microphone positions must be considered. A larger spacing will allow for lower frequencies to be studied and a smaller spacing for high frequencies. It is also important to know that a larger spacing will also increase the accuracy of the measurements at lower frequencies.

$$f_l = \frac{0.01 c_a}{b} \tag{2.8}$$

The spacing between the last microphone and the test specimen depends on the type of specimen being tested. Every specimen can be categorized into either a flat specimen, nonhomogeneous specimen, or asymmetrical specimen by the surface of the sample. The required distance for each specimen is described by Equations 2.9a, 2.9b, 2.9c, respectively.

$$a > \frac{d}{2}, \qquad a > d, \qquad a > 2d \qquad \text{(2.9a, 2.9b, 2.9c)}$$

The distance between the specimen and the first microphone will be used in calculations for the absorption coefficient and impedance of the specimen being tested. This distance is apparent for flat specimens due to the fact that it is just the frontal surface however, it may not be as simple for nonhomogeneous and asymmetrical specimens. In these cases you should use several different distances to the surface when calculating the acoustic properties of the specimen. One distance should include the point that is closest to the microphone. For calculating the absorption coefficient this distance has no effect on the calculation, though it does affect the calculation for the acoustic impedance.

2.3.2. Microphones

To measure the pressure in the tube two identical microphones are required. The microphones diameter should be small compared to the spacing between each other. The ASTM standards recommend that the microphone diameter be less than 20% of the maximum wavelength of interest as

$$d_m < 0.2\lambda_u. \tag{2.10}$$

The larger the microphone diameter, the higher the sensitivity at lower frequencies. If the tube is being built for lower frequencies only, then a larger diameter microphone would be beneficial. A table of upper frequency limits for common microphone sizes is shown in Table 2-1.

Table 2- 1 Recommended Maximum Frequency Based on Microphone Diameter [28]

Nominal Diameter (in)	Diaphragm Diameter (mm)	Maximum Frequency (Hz)
1	22.70	3000
½	12.20	5600
¼	5.95	11500

Microphones have a vent to the outside so that they can equalize with the atmospheric pressure. They can be purchased with vents on the side of the microphone and with vents in the rear. The standards recommended that rear venting microphones be used since they show low frequency accuracy improvements. The microphone's body should be sealed inside the microphone holder. This can be done by using a gasket or O-ring. If the microphone is sealed on the protection cap then the threads to the cap should be sealed with silicone grease. When the microphone is mounted the edge can be recessed into the tube or protrude a little into the tube. Consistency in mounting the microphones should be maintained to limit the error that different microphone positions would cause. If multiple microphone positions are being used then the empty microphone holders should be sealed during tests.

The acoustic center of each microphone may not match the physical center of the microphone. This will cause the distances between the microphones and the specimen to be off by an amount that depends on the microphone. To maintain the same distances between tests, the microphones should be marked and lined up to the same position whenever the microphones are moved or switched positions. This will maintain consistency in terms of the error caused by the acoustic center between testing of different materials.

2.3.3. Signal, Speaker, and Processing Equipment

Signal generation and processing equipment are needed to process the output of the impedance tube. This includes a signal generator, signal amplifier, sound source, and digital frequency analysis system. A signal conditioner and equalizer are optional additions to the list of equipment. The test signal is recommended to be a random noise with uniform spectral density. It is also possible to use pseudo-random noise or a swept sine wave since they also have a uniform spectral density. The signal needs to be synchronized by repeating the signal throughout the experiment. The signal will be amplified and the signal at each frequency band should be at least 10 dB above the background noise. To check to make sure this is satisfied, a highly absorptive material should be placed inside the tube and a test should be taken with the speaker both on and off. The results from any frequencies that do not exceed 10 dB should not be taken into account.

A temperature, pressure, and humidity sensor are needed for calculations of the acoustic impedance. The following are recommended values from the standards. The temperature sensor should be able to measure in temperature of the air inside of the tube to a tolerance of ±1℃. The atmospheric pressure should be measured with a tolerance of ±0.5 kPa. The humidity of the air around the tube should be measured with a tolerance of 5%.

The data received from the microphones will be used to calculate the transfer function between the two signals. This will be done using an Fast Fourier Transform (FFT) analyzer. A time weighting function or window is required since the recorded data is of a finite length. The window is used to turn the finite signal into a continuous one by linking the end of the signal back to the beginning. This is used to smooth the end of the data and remove spectral leakage caused by the Fourier Transform. It is recommended that a Hanning Window is used in the analysis with random noise signal [28]. If pseudo-random noise or a swept sine wave is used, then uniform or boxcar window should be used.

2.3.4. Other Requirements

There are other requirements that should be taken into consideration when building an impedance tube. These are listed below.

- Pressure can build up in the tube when new specimens are added to the tube. This excess pressure may damage the microphones. To prevent this a 1-2 mm hole should be drilled and threaded so that a bolt can be used to equalize the pressure.
- A back plate behind the specimen to reflect the sound during a test. It is recommended that the back plate be made out of a dense metal material with a minimum thickness of 20 mm.

- The end of the tube with the specimen will need to be closed off. This cap should have the same inner dimension as the tube. The connection between the tube and the cap should have an air tight seal. It is recommended that a sealant like petroleum jelly or silicone grease be used.
- At least two specimens of the same material should be constructed and tested. The results from each specimen should be averaged together when analyzing the acoustic properties.
- The specimen needs to fit perfectly in the tube and rest cleanly against the back plate. For softer materials, the specimen should snugly but should not be forced since that may cause a bulge in the center. It is possible that some specimen will not have a flat back. A small amount of a putty material should be used to fix the specimen. Care should be taken to limit the amount used since it can affect the results of the experiment.

2.3.5. Calculations of Acoustic Properties

The speed of sound inside the tube needs to be known when calculating the absorption coefficient. Since the temperature inside the tube is measured, it can be used to calculate the speed of sound. This is shown in Equation 2.11 where the temperature is measured in $^\circ\mathrm{C}$ and the speed of sound is measured in $\mathrm{m/s}$.

$$c_a = 20.047\sqrt{273.15 + T} \tag{2.11}$$

In order to validate the speed of sound, additional experiments can be performed to use the microphones themselves to calculate the speed of sound. Since the previous equation has already been previously validated, it will be used. The density of the air is also needed for the calculation of the acoustical impedance. This equation is shown below and is in units of gm/cm^3.

$$\rho = 1.290\left(\frac{P}{101.325}\right)\left(\frac{273.15}{273.15 + T}\right) \quad \frac{\mathrm{gm}}{\mathrm{cm}^3} \tag{2.12}$$

Once the speed of sound and air density is calculated, the next step is to use the pressure measured from the two microphones to calculate the transfer function between both signals. An FFT analyzer or computer with software for calculating transfer functions should be used. Since the transfer function is defined as the complex ratio of the two signals, any difference in the amplitude or phase between the two signals will affect the accuracy of the measurement. Even though both microphones should be identical, there will be differences in the amplitude and phase due to manufacturing defects or tolerances. This means that the transfer function calculated between the two signals will need to be corrected to account for the differences between the two microphones.

The transfer function can be corrected by the procedure presented next. A highly absorptive material should be placed inside the impedance tube. Since most materials are poor at absorbing sound at low frequencies, the accuracy at these frequencies will be limited. With the highly absorptive material inside the tube, a test should be performed and the transfer function between the two signals calculated. Then the two microphones positions should be switched. A second test should be performed with this positioning. The microphones are switched so that their position has no influence on the calculation since only the mismatch in microphones is what needs correcting. With these two transfer functions, Equation 2.13 can be used to calculate the calibration factor H_c.

$$H_c = \sqrt{H_{12} \times H_{21}} \tag{2.13}$$

H_{12} and H_{12} are the transfer functions for the standard and switched microphone positions, respectively. This equation is valid for when the microphone normally closest to the specimen is always used as the reference signal when calculating the transfer function. This means that during the second test when the microphones are switched, the microphone now furthest from the specimen is used as the reference microphone. If when switched you would rather use the other microphone as the reference signal because it is now closest to the sample, then a different equation must be used. It is similar to Equation 2.9 but the multiplication operator is changed to division. The microphones should then be switched back before an actual test is performed. This calibration factor can be used to correct the transfer function calculated during a regular test. This is shown in Equation 2.14, where H_t is the transfer function measured during an actual test of a specimen.

$$H = \frac{H_t}{H_c} \tag{2.14}$$

The reflection coefficient can now be calculated. This is shown in Equation 2,15.

$$R = \frac{H - e^{-jks}}{e^{jks} - H}\, e^{j\,2k(l+s)} \tag{2.15}$$

In the equation above the variable ‘k’ is the complex wave number. This number is can be affected by the sound attenuation. Both incident and reflected waves that travel through the tube are subject to sound attenuation due to the viscous and thermal losses. The effects of sound attenuation can sometimes be ignored if the specimen is very close to the first microphone. In this case $k = \frac{2\pi f}{c}$.[35] If the specimen is further than three diameters from the first microphone then it must be taken into consideration. The wave number for this case is $k = \frac{2\pi f}{c} - j\,0.02203\frac{\sqrt{f}}{cd}$. Using the reflection coefficient, the absorption coefficient can be calculated using Equation 2.3 from the standing wave ratio. The impedance can be calculated by using Equation 2.4

Other acoustic properties can be calculated like the acoustic resistance, reactance, and capacitance. For the equations to these values look at the ASTM standards.

2.4. Two Microphone Transfer Function Method ISO [31]

Similar to the ASTM, the International Organization for Standardization (ISO) also has a collection of information for standards in building an impedance tube using the two microphone transfer function method[31]. In general the two standards are very similar to each other, though there are differences in the values for the constants that are used in many of the equations. There are also some requirements that are not mentioned in the ASTM standard. The ISO standards also have a second technique that only uses one microphone instead of two. This standard is presented in the next section with emphasis on differences from the ASTM standard.

2.4.1. Tube Specifications

The tube should be built so that it has smooth non-porous walls. For our purposes only circular cross section tubes will be considered. The tube can be made of metal, concrete, and wood. Unlike the ASTM standards, there is no mention of plastic as a material. If concrete is chosen then an adhesive needs to be used to seal the pours throughout the wall to ensure air cannot pass though. Wood walls should have a sheet of steel or lead surrounding the outside of the tube. The tubes general structure is exactly the same as the two microphone impedance tube based off of the ASTM standards. Figure 2-2 will be used to reference locations of microphone distances. The locations of the microphones should be known within ±0.2mm. Just like in the ASTM standards the length between the speaker and the first microphone should be three tube diameters or stated by Equation 1. To also ensure that the plane waves are maintained, the diameter of the tube is directly related to the upper frequency limit and the speed of sound. This is shown in Equation 2.16.

$$d < \frac{0.58c_a}{f_u} \tag{2.16}$$

Tests should not be measured above this upper frequency. The ASTM standard described by Equation 2.6 is similar to Equation 2.16 except for the constant being 0.586 instead of 0.58. The spacing between the microphones depends on the upper frequency of interest. This is described by Equation 2.17.

$$b < \frac{0.45c_a}{f_u} \tag{2.17}$$

This spacing is required because the reflection coefficient cannot be determined when the microphone spacing reaches an integer multiple of the half wavelength [34]. The ASTM standard described by Equation 2.7 is similar to Equation 2.17 except for the constant being 0.40 instead of 0.45. The microphone spacing also has a direct relation to the minimum frequency that can be measured. This is shown in Equation 2.18.

$$f_l = \frac{0.05c_a}{b} \tag{2.18}$$

Since a single microphone spacing will not cover all frequency ranges, multiple microphone positions must be considered if a wide frequency range is studied. A larger spacing will increase the accuracy of the measurements. The ASTM standard described by Equation 2.8 is similar to Equation 2.18 except for the constant being 0.01 instead of 0.05. The difference between the two standards for the lower frequency is significant since low frequencies studied using the ISO standard would require a microphone spacing that is five times the distance of the ASTM required spacing. This would require the tube to be much longer if very low frequencies are to be studied.

Similar to the ASTM standard the spacing between the last microphone and the test specimen depends on the type of specimen being tested. The required distance for each specimen is described by Equations 2.9a, 2.9b, 2.9c for each specific type of specimen surface.

The microphone requirements are similar to those described by the ASTM standards and presented in section 2.3.2 of this paper. The standard recommends that the microphone diameter should be much less than the wavelength corresponding to the upper wavelength of interest. It is also recommended that the microphone diameter be less than 20% of the spacing between the two microphones. This recommendation is not mentioned in the ASTM standards.

2.4.2. **Signal, Speaker, and Processing Equipment**

Just like the ASTM standards signal generation and processing equipment are also needed for the impedance tube. The requirements included in section 2.3.3 for the

ASTM standards are the exact same requirements described in the ISO standards. There is no mention in the ISO standards of using an equalizer in the test setup. The dynamic range of the FFT analyzer should be greater than 65dB. The load speaker membrane should be at least two-thirds the cross sectional area of the tube. The speaker can be directly connected straight on or with the use of an elbow. The connection between the speaker and tube requires a gasket to remove mechanical vibration. The loudspeaker should be enclosed in a sound insulated box to avoid outside sound generated from the speaker reaching the microphone.

Another key difference in the ISO standard is the requirement to line the inside of the tube near the speaker with a sound absorbing material 200mm long. This is used to stop the resonance of the air in the tube.

2.4.3. Other Requirements

There are other requirements that should be taken into consideration when building an impedance tube. All requirements stated in 2.3.4 are also required in the ISO standards unless stated below.

- The use of a hole for equalizing the pressure of the tube is not stated in the ISO standards like it is in the ASTM standards.
- The cap sealing the tube with the specimen should be sealed without the use of an elastic gasket. This is not mentioned in the ASTM standards. It

is recommended that a sealant like Vaseline be used.

The calculations described in section 2.3.5 for the ASTM standards are described the same in the ISO standards.

2.4.4. **One Microphone Technique**

The ISO standards also describe a technique to use one microphone instead of two. For this method a test is performed with the same microphone in each of the two microphone locations. The benefit to using one microphone is that there is no calibration required since the same microphone is being used through the entire test. This technique is recommended when tuned resonators are being studied or if a more precise measurement is required. This technique can be performed using two different techniques. One is with fixed microphone locations and the other is with variable microphone locations. The fixed location will be studied because it can be used with the two microphone impedance tube. The other would require a different tube entirely. The sound at the two locations needs to be recorded sequentially with a sound source that is unchanging. It is recommended to use a deterministic signal as the sound source. The periodic pseudo-random noise is recommended. This will require a maximum length sequence using the Walsh-Hadamard transform to produce an impulse response. [28] Then the Fourier transform can be used to measure the frequency response. The transfer functions between the two microphone locations is described by Equation 2.19.

$$H_{12} = H_{x2}/H_{x1} \tag{2.19}$$

where ‘x’ is the generated signal. A transfer function is measured between the signal and the first microphone. Then the transfer function is measured between the signal and the second microphone. Using Equation 2.19 the equation between the microphones can be calculated.

2.5. Four Microphone Method

The two microphone impedance tube has been widely used for measuring the normal incident absorption coefficient and impedance of materials. Another important acoustic property that cannot be measured by the traditional two microphone impedance tube is the sound transmission loss of the material. Sound transmission loss is defined as an integer that describes how well a material attenuates sound. The two microphone impedance tube cannot measure this since there is no microphone behind the specimen to measure the acoustic wave that has been transmitted through the material. The back plate behind the specimen reflects this acoustic wave back through the specimen. Instead, the four microphone method was developed to calculate this value by removing the back plate and adding two more microphones behind the specimen. The original four microphone tubes were developed using an existing two microphone impedance tube with an adapter that added two additional microphones after the specimen[36]. ASTM has created a standard developed after this design. The requirements to create the tube are identical to those of the ASTM two microphone impedance tube. This standard is presented in the next section with emphasis on the additional information about the four microphone tube.

2.5.1. **Tube Specifications**

The tube will be constructed as if the two microphone impedance tube was being created with an additional two microphone tube after the specimen. A simple diagram for the four microphone impedance tube is shown in Figure 2-3. Equations 2.1-2.5 for the two microphone impedance tube apply for this tube as well. Equation 1 can be used for determining the distances between the speaker and first microphone. Equation 2 can be used to determine the diameter of the tube. Equations 2.6 -2.8 are used to determine the spacing between the microphones along with the upper and lower frequencies being studied in the tube. Equation 2.9 describes the minimum distance a test specimen can be from the closes microphone depending on the surface of the material.

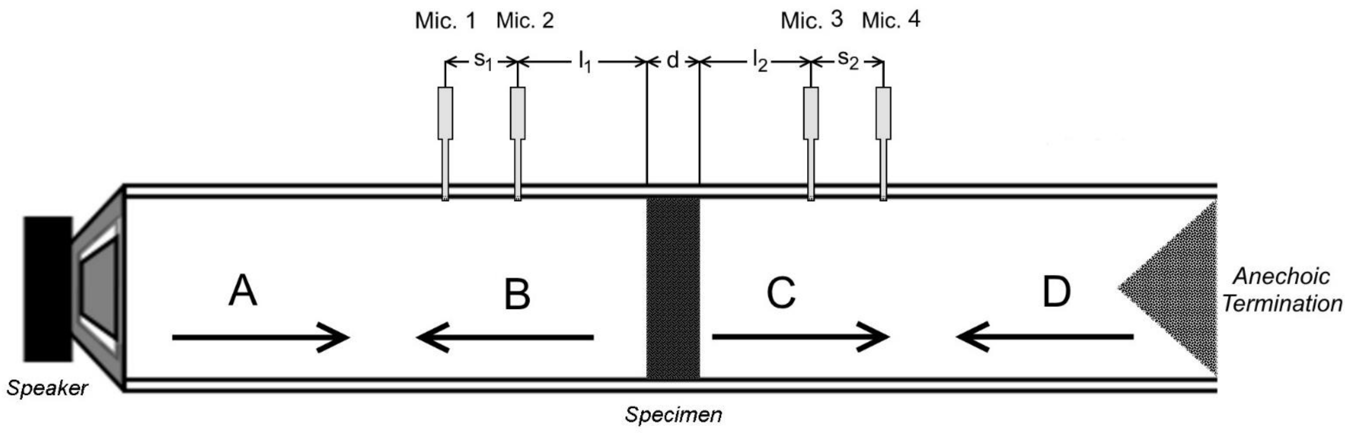


Figure 2.3 Four Microphone Impedance Tube Diagram[29]

The microphones, sound source, speaker, and processing equipment requirements are also the same as the two microphone ASTM standard and can be seen in sections

2.3.2 and 2.3.3. There are two different methods for using the tube. The one-load method involves only taking one measurement which requires an anechoic termination at the opposite end of the speaker. This could be an open tube but it is recommended that a sound absorbing material in the shape of a wedge or pyramid that is 30cm long be placed inside the tube. The two-load requires that two measurements be taken with different end terminations. This would be one that is anechoic and another that is reflective. This could be the wedge insert and a reflective cap, respectively.

The microphone requirements are exactly the same as those described by the ASTM standards for the two microphone tube and are presented in section 2.3.2 of this paper. The requirements for the signal, speaker, and processing equipment are also identical to the two microphone tube and can be read in section 2.3.3.

2.5.2. Other Requirements

There are other requirements that should be taken into consideration when building an impedance tube. Unless otherwise stated all requirements stated in 2.3.4 are also required except for those pertaining to a back plate. This is obvious since there is no back plate in this method. These are listed below.

- The mounting of the specimen has a large impact on the transmission loss calculation. The specimen can be mounted or stand freely but the method for mounting should be consistent with different samples.

- The specimen should fit perfectly in the tube as to not have cracks in between the tube wall and itself since this will greatly affect the transmission loss. The standard recommends that cracks can be filled with petroleum jelly or modeling clay.
- At least three specimen of the same material should be constructed and tested. The results from the specimens should be averaged together. Additional samples should be tested if the surface of the material is not uniform.

2.5.3. Theory and Calculations of Acoustic Properties

The calculations for the speed of sound and air density are shown in 2.3.5. Once the speed of sound and air density is calculated, the next step is to use the microphones in their fixed positions to calculate the transfer function between both signals. These transfer functions can be calculated by using only one microphone to as many as five microphones. The number of microphones used affects the amount of time that is required to take a measurement of a sample. If only one microphone is used, then that single microphone will be used to measure the pressure at all four of the microphone locations. The transfer functions should be calculated relative to the source signal. Table 2-2 shows the different configurations that can be used and the procedure used when taking data. For example, if only one microphone is used, no calibration needs to be performed since the same microphone measures the pressure at each location. The table describes what transfer functions need to be calculated along with if the microphone needs to be calibrated. Location 0 is location other than the four microphone locations required for testing and is near the sound source. It is similar to using the signal as the reference for calculating transfer functions. The procedure for each case is also shown.

Table 2- 2 Measurement Configurations Based off Number of Microphones and Channels [29]

Number of Channels	Number of Mics	Transfer Functions Reference	Transfer Functions Measured	Correction	Procedure
2	1	Source Signal	$H_{1s}, H_{2s}, H_{3s}, H_{4s}$	None	Mic moves from locations 1-4
2	2	Mic 1 at location 0	$H_{10}, H_{20}, H_{30}, H_{40}$	None	Mic 2 moves to locations 1-4
2	2	Mic 1 at location 1	$H_{11}, H_{21}, H_{31}, H_{41}$	None	Mic 2 moves to locations 2-4
4	4	Mic 1 at location 1	$H_{11}, H_{21}, H_{31}, H_{41}$	$H_{21}{}^{c}, H_{31}{}^{c}, H_{41}{}^{c}$	Mic in Fix locations
5	4	Source Signal	$H_{1s}, H_{2s}, H_{3s}, H_{4s}$	$H_{1s}{}^{c}, H_{2s}{}^{c}, H_{3s}{}^{c}, H_{4s}{}^{c}$	Mic in Fix locations
5	5	Mic 5 at location 0	$H_{10}, H_{20}, H_{30}, H_{40}$	$H_{10}{}^{c}, H_{20}{}^{c}, H_{30}{}^{c}, H_{40}$	Mic in Fix locations

An FFT analyzer or computer with software for calculating transfer functions should be used. Since the transfer function is defined as the complex ratio of the two signals, any difference in the amplitude or phase between the two signals will affect the accuracy of the measurement. This means that the transfer function calculated between the two signals will need to be corrected to account for the differences between the two microphones. This is only required if more than two microphones are used when testing.

The transfer function can be corrected by the procedure presented next. A highly absorptive material should be placed inside the impedance tube. The sound absorbing material that is in the shape of a wedge or pyramid can be used. A test should be performed and the transfer function between the two signals calculated. Then the two microphones positions should be switched. A second test should be performed with this position. With these two transfer functions, Equation 2.13 can be used to calculate the calibration factor. Then Equation 2.14 can be used to correct the microphone differences during a test.

To calculate the transmission loss of a material there are two different methods for using the tube. There is the one-load method and the two-load method which were described earlier. Both methods involve calculating the transfer matrix. To calculate the transmission loss for the two load method two measurements must be taken with reflective cap at the end and one with an anechoic end. The relationship between the acoustic pressure, velocity, and transfer matrix are presented in Equation 2.20 and 2.21.

$$\begin{bmatrix} p_a \\ u_a \end{bmatrix}_{x=0} = \begin{bmatrix} T_{11} & T_{12} \\ T_{21} & T_{22} \end{bmatrix} \begin{bmatrix} p_a \\ u_a \end{bmatrix}_{x=d} \tag{2.20}$$

$$\begin{bmatrix} p_b \\ u_b \end{bmatrix}_{x=0} = \begin{bmatrix} T_{11} & T_{12} \\ T_{21} & T_{22} \end{bmatrix} \begin{bmatrix} p_b \\ u_b \end{bmatrix}_{x=d} \tag{2.21}$$

The pressure and velocity can be calculated from the measurements of the four microphones. This can be done by calculating the forward and transverse traveling waves. Where 'A' and 'B' that were shown in Figure 2.3 are the forward and transverse wave in the first portion of the tube respectively. 'C' and 'D' are the forward and transverse wave in the second portion of the tube after the specimen respectively. These waves can be calculated by using the transfer functions between each position relative to the reverence and are presented in Equations 2.22a,b,c,d.

$$A = j\,\frac{H_{1,r}e^{-jkl_1} - H_{2,r}e^{-jk(l_1+s_1)}}{2\sin ks_1} \tag{2.22a}$$

$$B = j\,\frac{H_{2,r}e^{jk(l_1+s_1)} - H_{1,r}e^{jkl_1}}{2\sin ks_1} \tag{2.22b}$$

$$C = j\,\frac{H_{3,r}e^{jk(l_2+s_2)} - H_{4,r}e^{jkl_2}}{2\sin ks_2} \tag{2.22c}$$

$$D = j\frac{H_{4,r}e^{-jkl_2} - H_{3,r}e^{-jk(l_2+s_2)}}{2\sin ks_2} \tag{2.22d}$$

Using Equations 2.22a,b,c,d.the acoustic pressure and wave velocity can be calculated. This is presented in Equations 2.23a,b,c,d.

$$p_0 = A + B \tag{2.23a}$$

$$p_d = Ce^{-jkd} + De^{jkd} \tag{2.23b}$$

$$u_0 = \frac{(A - B)}{\rho c_a} \tag{2.23c}$$

$$u_d = \frac{\left(Ce^{-jkd} - De^{jkd}\right)}{\rho c_a} \tag{2.23d}$$

Once the measurements are taken with a reflective cap at the end and with an anechoic end, the acoustic pressure and wave velocity can be calculated for both positions. Then by using Equations 2.23a and b, the transfer matrix can be solved in terms of the acoustic pressure and wave velocity. This is shown in Equation 2.24 where a and b subscripts denote the reflective and anechoic end cap respectively.

$$T = \begin{bmatrix} \frac{p_{0a}u_{db} - p_{0b}u_{da}}{p_{da}u_{db} - p_{db}u_{da}} & \frac{p_{0b}p_{da} - p_{0a}p_{db}}{p_{da}u_{db} - p_{db}u_{da}} \\ \frac{u_{0a}u_{db} - u_{0b}u_{da}}{p_{da}u_{db} - p_{db}u_{da}} & \frac{p_{da}u_{0b} - p_{db}u_{0a}}{p_{da}u_{db} - p_{db}u_{da}} \end{bmatrix} \tag{2.24}$$

There is a special case that arises when the specimen is geometry symmetric. In this case the transfer matrix will have the properties that T_{11}=T_{22} and $T_{11}22 - T_{12}T_{21}$=1. This allows for only one test to be performed instead of two. This is considered the one-load method. The transfer matrix can be calculated using Equation 2.25.

$$T = \begin{bmatrix} \frac{p_d u_d + p_0 u_0}{p_0 u_d + p_d u_0} & \frac{{p_0}^2 - {p_d}^2}{p_0 u_d + p_d u_0} \\ \frac{{u_0}^2 - {u_d}^2}{p_0 u_d + p_d u_0} & \frac{p_d u_d + p_0 u_0}{p_0 u_d + p_d u_0} \end{bmatrix} \tag{2.25}$$

Using the elements from transfer matrix from the one-load or two-load method, the transmission coefficient can be calculated. This is shown in Equation 2.26. In the one-load case, the anechoic cap should be used in the calculations for the transmission coefficient and subsequently the transmission loss.

$$t = \frac{2e^{jkd}}{T_{11} + \left(\frac{T_{12}}{\rho c}\right) + \rho c_a T_{21} + T_{22}} \tag{2.26}$$

From the transmission coefficient it is possible to calculate the normal incidence transmission loss. This is shown in Equation 2.27.

$$TL_n = 20log_{10}\left|\frac{1}{t}\right| \tag{2.27}$$

Just like the two microphone impedance tube, the reflection coefficient can also be calculated. This is shown in Equation 2.28. In the one-load case, the reflective cap should be used in the calculations for the reflection coefficient and subsequently the absorption coefficient and normal specific acoustic impedance.

$$R = \frac{T_{11} - \rho c_a T_{21}}{T_{11} + \rho c_a T_{21}} \tag{2.28}$$

The absorption coefficient and normal acoustic impedance can be calculated from Equations 2.3 and 2.4 from section 2.3.5. It is also possible to calculate the propagation wavenumber and characteristic impedance inside the material being tested. This is shown by Equations 2.29 and 2.30.

$$k' = \frac{1}{d}\cos^{-1} T_{11} \tag{2.29}$$

$$z = \sqrt{T_{12}/T_{21}} \tag{2.30}$$

2.6. **Three Microphone Method [32, 37, 38]**

To calculate the normal incident sound transmission loss another method was developed which allows for the use of one less microphone. This method is similar to the ASTM standard E2611-09 except that the third microphone is placed behind the sample inside the metal back plate. This method is more common among international researches and there has been no standard developed yet for the development of these tubes. Figure 2-2 shows what the configuration for the three microphones is. The third microphone is turned 90 degrees in relation to the first two microphones.

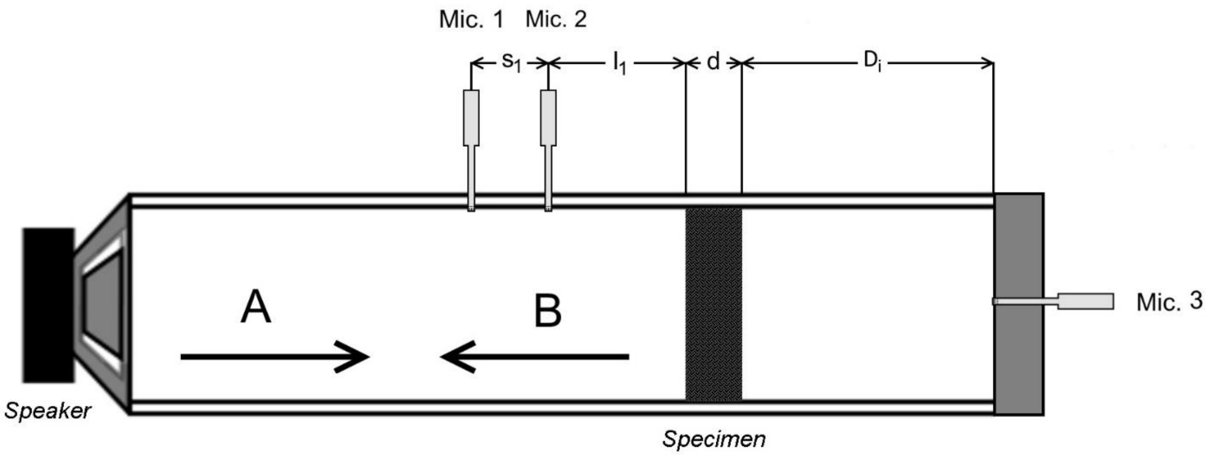


Figure 2.4 Three Microphone Impedance Tube [38]

The first configuration was developed by T. Iwase [39] but also performed by Salissou [40] and places the microphone and back plate directly behind the sample being tested. This is similar to the two microphone method with the addition of the third microphone inside the back plate. This method is limited to specimens with flat surfaces that are both homogenous and symmetrical. To solve this problem a second method can be used. A second method was developed by Salissou[37, 38, 41, 42] which uses the third microphone to measure the sound pressure at two locations with an air gap between the back plate and the specimen. This is known as the two load method. If the specimen being tested is symmetrical and homogenous then the back plate only needs to be tested at one location. This is known as the one load method. If there is no air gap and the back plate is directly behind the sample, then this is identical to the configuration performed by T. Iwase. In this thesis Salissou's configuration with the air gap behind the specimen will be studied. Since the three microphone method is based off of the ASTM standard E2611-09, the tube specifications are exactly the same except for the end of the tube. First we will look at the two load case which can be used with a non-symmetric specimen. To calculate the acoustic properties of the specimen the transfer matrix needs to be computed. Equation 2.24 from the four microphone method is repeated in Equation 2.31 but the a and b subscripts denote the two different positions that the third microphone will be placed at.

$$T = \begin{bmatrix} \frac{p_{0a}u_{db} - p_{0b}u_{da}}{p_{da}u_{db} - p_{db}u_{da}} & \frac{p_{0b}p_{da} - p_{0a}p_{db}}{p_{da}u_{db} - p_{db}u_{da}} \\ \frac{u_{0a}u_{db} - u_{0b}u_{da}}{p_{da}u_{db} - p_{db}u_{da}} & \frac{p_{da}u_{0b} - p_{db}u_{0a}}{p_{da}u_{db} - p_{db}u_{da}} \end{bmatrix} \tag{2.31}$$

To solve the transfer matrix, the acoustic pressure and the particle velocity can be calculated upstream and downstream. These can be calculated using equations 2.31a,b,c,d.

$$p_0 = j\frac{H_{1,r}e^{-jkl_1} - H_{2,r}e^{-jk(l_1+s_1)}}{2\sin ks_1} \tag{2.31a}$$

$$p_d = Ce^{-jkd} + De^{jkd} \tag{2.31b}$$

$$u_0 = \frac{(A-B)}{\rho c_a} \tag{2.31c}$$

$$u_d = \frac{\left(Ce^{-jkd} - De^{jkd}\right)}{\rho c_a} \tag{2.31d}$$

Looking at Equations 2.30 and 2.31a,b,c,d if they were combined, it can be seen that there is a case where the transfer matrix cannot be solved. This is when $\cos kD_a \sin kD_b = \cos kD_b \sin kD_a$. The spacing between the two back plate positions is required to satisfy Equation 2.32.

$$D_a - D_b < \left|\frac{\pi}{k}\right| \approx \frac{172}{f_u} \tag{2.32}$$

When the specimen is symmetric, just like in the four microphone setup, the transfer matrix can be simplified. In this case the transfer matrix will have the properties that T_{11}=T_{22} and $T_{11}T_{22} - T_{12}T_{21}$=1. This is shown by Equation 2.25 from section 2.5.4. For the one load method, only one test has to be performed. The back plate only needs to be placed at one location.

The transmission loss and reflection coefficient can be calculated from Equations 2.27 and 2.28 from section 2.5.4. The absorption coefficient and normal acoustic impedance can be calculated from Equations 2.3 and 2.4 from section 2.3.5.

2.7. Comparison of Impedance Tube System Designs

There are several methods for measuring the acoustic properties of materials. This thesis is only looking at methods based off of the standing wave theory and the utilization of a tube. There are other methods like the reverberation room method that have also been used. This method however requires a large amount of material to be tested. This is why our study only looks at the use of an impedance tube. In this section each method will be examined and compared to the other designs.

The standing wave ratio method does have some benefits to it. Since the standing wave method is performed by measuring the actual standing wave by hand, there is no need for an impedance analyzer which is required for all of the other methods. This would allow for a less expensive tube to be built because of it. This method only requires one microphone which can further decrease the cost of the equipment and ease of use. The main disadvantage to this method is how long it takes to perform an experiment. Since each test can only be performed at a single frequency, to measure a full frequency band would be tedious. This method is very time consuming when you consider that multiple samples of the same material must be tested. This is where the two microphone method would be beneficial.

The two microphone method takes only a few minutes to perform a test over the entire frequency range of interest. Since this is done by the transfer function method, an impedance analyzer is required. This would increase the cost of building this type of tube. Since the operating frequency range is dependent on the microphone spacing, multiple microphone locations are needed if a full spectrum required. This would require a few more tests to be performed but it would still be quicker than the standing wave ratio method. Since two microphones are used, a calibration procedure must be performed to account for microphone amplitude mismatch. The values measured from both methods are in good agreement with each other[33, 43]. In general it was found by Suhanek that the absorption coefficient measured from the two microphone method was slightly lower than the values measured from the standing wave ratio. It was concluded that the measurements accuracy could be improved if the tube as more rigid and smaller microphones were used. There experiment was performed with a ½" diameter microphones.

The difference between the ASTM and ISO standards for a two microphone impedance tube are very small. The main difference between them is the equation for calculating the microphone spacing required for the frequency range of interest.

Microphone locations can be chosen so that both standards are satisfied and a full frequency range is still achieved. The ISO standards also have no mention of the tube being allowed to be constructed from a plastic material. The ISO standard also has an added recommendation that the microphone diameter be less than 20% of the spacing between the two microphones. This is not mentioned in the ASTM standards. The ISO standards adds a one microphone method which removes the need for calibrating the microphones since only one microphone is used. There are a few other small additions to the tube requirements but in general the two standards are the same.

The four microphone method tube is similar to the two microphone tube except for the added microphones after the sample. The four microphone method's main use is for measuring sound transmission loss but it can also be used measure the absorption coefficient that the two microphone method can. If the sound transmission loss is the property being studied then the two microphone method could not be used. Since both methods can be used to measure the absorption coefficient of a sample, there results can be compared to one another. The results depend on the type of material that is being tested. The results obtained from fibrous materials would be in good agreement with both types of tubes [44]. However the results obtained from a poroelestic foam material would not be as good. This is because of the boundary condition between the two tubes. In the two microphone tube, the sample is held by the back plate behind it. Since the four microphone tube is only being held by the inside of the tube, this still allows the sample to move in the tube [44]. This should be taken into consideration when using a four microphone tube to measure the absorption coefficient.

The last method to consider is three microphone method. Like the four microphone method, it is used to measure the transmission loss of a sample. This method has the benefit to requiring one less microphone. This requires fewer transfer functions to be calculated. Since the third microphone is positioned on the end cap, this makes it easier to transition from a two microphone setup, to a three microphone set up. This method is more commonly used outside of the United States and there has been no standard created yet. Salissou compared the four microphone setup to his own design and found that they were in good agreement with each other [32]. He also shows that the three microphone configuration also complies with the ASTM standard E2611-09.

An impedance tube analyzer if purchased can be a very expensive piece of equipment. Brüel & Kjær is a popular company to purchase an impedance analyzer from. They have models of both the two and four microphone impedance tubes based off of the ASTM standards E1050-12 and E2611-09. This can cost tens of thousands of dollars. Purchasing the DAQ equipment and using the standards to build an impedance tube, a large amount of money can be saved.

When first starting to build an impedance tube there are many design parameters to take into consideration. These parameters can be include but are not limited to the tube material, type of speaker, signal generated, etc… When choosing the signal, although the standard allows for multiple types, it is possible that the results can be different. Suhanek found that when using sine sweep as their signal, there were dips in the absorption coefficient for certain materials [45]. This was less prominent when pink noise was used as the signal. Suhanek also claims that the accuracy of the measurement would be increased if the tube was more rigid and smaller microphones of ¼” diameter are used. Care should also be taken when deciding on what microphones will be used since the accuracy depends on the quality of the microphones. According to O’Malley, a significant amount of distortion in his results was found when testing his newly built impedance tube[46]. He attributed this to both distortion in the microphones and the driver, with the microphones contributing more due to overloading. Reducing the voltage to the driver helped minimize the distortion. It is recommended that high quality microphones be used in the experiment. It is mentioned that linearity is the most important quality when choosing a microphone, not sensitivity [46]. O’Malley also studied a variety of different drivers. He looked for the driver with the least amount ringing and best frequency response characteristics over the frequency band of interest [46]. He found that the Selenium DH200E showed the least amount of ringing but the frequency response was poor under 500Hz [46].

Chapter 3.
Description of Tube Design

After doing a comparison on the different types of impedance tube designs, several tube configurations were chosen to be examined, based on flexibility and efficiency. The SWR method will not be considered as part of the analysis in this thesis because of the length of time it takes to perform measurements over a full frequency range, as the SWR method does not provide a full spectrum for the absorption coefficient. It was decided that since the two, three, and four microphone impedance tubes are very similar in design that all three tubes should be built and analyzed for this thesis. A comparison of the tubes will be performed. The diagram for the two microphone impedance tube is shown in Figure 3.1 with the dimensions.

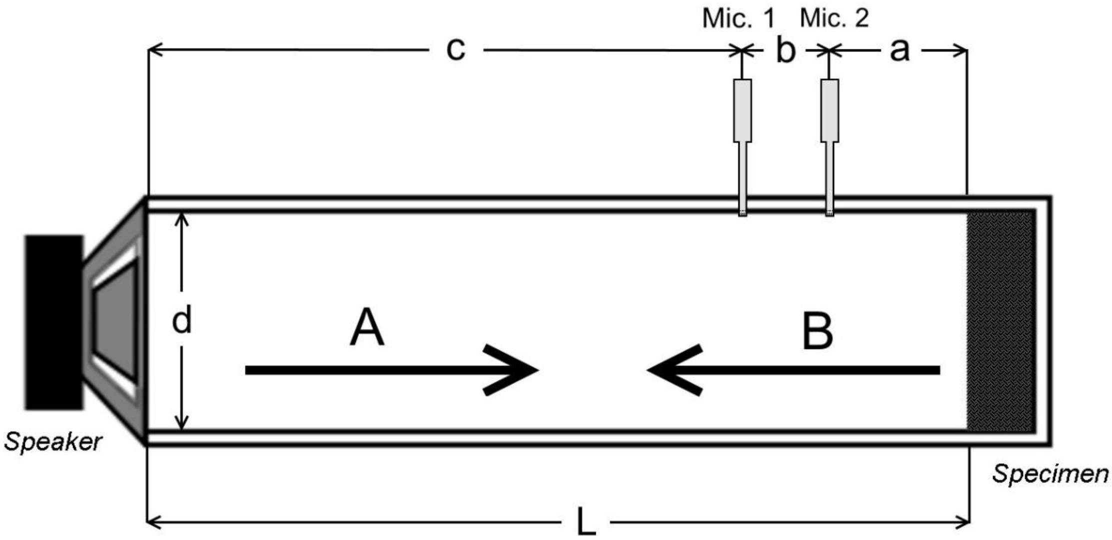


Figure 3.1 Two Microphone Impedance Tube Diagram with Dimensions[28]

During the process of building the two microphone impedance tubes, questions arose as to what would happen if some of the tube design parameters were slightly different even though they still complied with the standards. For this reason, several different tubes were built and different test parameters were changed during operation of the tube. Unless otherwise noted, ASTM standards were used in the design of the acoustic impedance tube designs.

3.1. Tube Material

The tube material was one the first variables to be varied in the analysis of acoustic impedance tube performance as a function of design parameters. This was because commercially only a limited number of diameters are manufactured depending on the material. Since our goal at first was to build an impedance analyzer for the smallest amount of money possible, it was decided that standard PVC tubing would be the perfect material. During testing, however questions arose as to whether PVC was dense enough to allow the tube to lose noise through the tube walls. Since this could be design problem, the PVC was wrapped in stainless steel foil as mentioned in the ASTM standards and the results compared to those of a tube made with an even denser material brass [28]. This study will be discussed in detail in Chapter 4.

3.2. **Tube Parameter Design**

After performing a full review of the ASTM standards, there are a significant number of parameters that contribute to design of the tube, such as tube length, diameter, microphone separation, microphone size, etc… Table 3.1 provides a list of the design parameters as a function of the tube diameter. These diameters were determined based on the sizes of commercially available PVC sizes. The variables, *a*, *b*, and *c* are the same variables described in Figure 2.2. The distance between the sample and first microphone "a" is the minimum distance required if the test specimen has a flat surface. When the test specimen has a front surface that is either nonhomogeneous or asymmetrical, then the distance will be larger. The maximum microphone spacing "*b*" is determined according to the upper frequency limit.

Table 3.1 Tube Dimensions Based off of Available PVC Sizes

d(in)	f_{upper}(Hz)	Λ_{upper}(in)	d_{mic} (in)	a(in) min	b(in) max	c(in) min	L(in)
0.5	15827	0.853	0.171	0.25	0.341	1.5	2.091
0.75	10551	1.280	0.256	0.375	0.512	2.25	3.137
1	7913	1.706	0.341	0.5	0.683	3	4.183
1.25	6331	2.133	0.427	0.625	0.853	3.75	5.228
1.5	5276	2.560	0.512	0.75	1.024	4.5	6.274
1.75	4522	2.986	0.597	0.875	1.195	5.25	7.320
2	3957	3.413	0.683	1	1.365	6	8.365
2.25	3517	3.840	0.768	1.125	1.536	6.75	9.411
2.5	3165	4.266	0.853	1.25	1.706	7.5	10.456
3	2638	5.119	1.024	1.5	2.048	9	12.548
4	1978	6.826	1.365	2	2.730	12	16.730
5	1583	8.532	1.706	2.5	3.413	15	20.913

Studying Table 3.1, it can be seen that as the diameter increases, the upper frequency limit that can be tested decreases. For the experiments in our laboratory, the frequencies of interest are between frequencies 50Hz-5000Hz. This means that a tube diameter equal to 1.5 inches or smaller can be chosen. A tube diameter of 1.25 inches was chosen so that there could be more flexibility in the upper frequency limit if needed. This size is relatively simple to fabricate and test. Care was taken in the selection of the microphones since the microphones' diameters were required to be less than or equal to that stated in the chart. The spacing between the microphones should also be carefully looked at since to achieve 6331Hz for a 1.25in diameter tube, the microphones need to be 0.853in apart. Some microphones can have a large base which could make it impossible to fit two microphones that close together.

3.3. Microphones

The recommended microphone diameter increases as the tube diameter increases. Since the tube diameter of 1.25in was chosen, the microphones diameter needs to be equal to or less than 0.427in. There are three types of condenser-type microphones: free-field, pressure-field, and random incident-field. The two types that would work for our situation are free-field and pressure-field, and while pressure field is more ideal, free-field is much cheaper. The companies that provide microphones used in most of these commercially available impedance tubes were much too expensive. The only alternative is to look into test-microphones used for room acoustics. For out setup the Audix TM-1 microphone was chosen because of its size, price comparability, usable frequency range, and great customer support. Audix also provides a purchasable sound source to mate with the microphone as a pre-amplifier. The TM-1 has a diameter of 6mm (0.236in) which is smaller than the recommended size for a tube diameter of 1.25in.

3.4. **Frequency Range**

The maximum operating frequency is determined by the diameter of the tube. This can be measured when the microphones are located at the position shown in Table 3.1. This microphone positioning has a very high limit for the lower frequency that can be studied. This means that you wouldn't be able to accurately measure the lower frequencies. To measure this and the full frequency spectrum multiple microphone locations are required. Table 3.2 shows the upper and lower frequency limits for different microphone spacing's. Since the ASTM and ISO standards have different equations for these calculations, both are shown in the table for comparison.

Table 3.2 Frequency Dependance on Microphone Spacing

b (in)	f_{upper} ASTM (Hz)	f_{upper} ISO (Hz)	f_{lower} ASTM (Hz)	f_{lower} ISO (Hz)
0.853	6332.444	7124.000	158.311	791.556
1.000	5401.575	6076.772	135.039	675.197
2.500	2160.630	2430.709	54.016	270.079
3.000	1800.525	2025.591	45.013	225.066
4.500	1200.350	1350.394	30.009	150.044

5.000	1080.315	1215.354	27.008	135.039
6.000	900.262	1012.795	22.507	112.533
7.000	771.654	868.110	19.291	96.457
7.500	720.210	810.236	18.005	90.026
8.000	675.197	759.596	16.880	84.400
10.000	540.157	607.677	13.504	67.520
12.000	450.131	506.398	11.253	56.266
13.000	415.506	467.444	10.388	51.938
14.000	385.827	434.055	9.646	48.228
15.000	360.105	405.118	9.003	45.013
16.000	337.598	379.798	8.440	42.200
17.000	317.740	357.457	7.943	39.717
18.000	300.087	337.598	7.502	37.511
20.000	270.079	303.839	6.752	33.760
25.000	216.063	243.071	5.402	27.008
30.000	180.052	202.559	4.501	22.507
33.000	163.684	184.145	4.092	20.461
34.000	158.870	178.729	3.972	19.859
35.000	154.331	173.622	3.858	19.291

When first looking at Table 3.2 the differences between the ASTM and ISO calculations for the upper and lower frequency limits can be seen. For the upper frequency limit the ISO standard claims that slightly higher frequencies can be studied for a specific microphone distances. The lower frequency limit however shows a much bigger difference between the two standards. With the ASTM standards, a microphone spacing of 7in should be sufficient in measuring the absorption coefficient at 20 Hz. The ISO standards however require a much larger distance equal to about 34 inches of separation between the microphones. This is a pretty significant difference which will be summarized later in Chapter 4. To study this, seven microphone locations were selected to provide a wide frequency range. The seven microphone locations are highlighted in Table 3.2.

3.5. Microphone Holders

The microphone holders were designed in Solidworks to exactly fit the Audix TM-1 microphones. The microphones rest in the holder so that the diagram is slightly recessed. The microphone holders contain a gasket inside of the tube to provide an air tight seal between itself and the microphone. The microphone holders were 3D printed using Stratasys Objet Polyjet at the Rutgers University Mechanical Engineering department so that the gasket could be printed inside of the microphone holder. This also ensures that all seven microphone holders are identical so that the microphones are in the same location relative to the tube wall. The microphone holders are depicted in Figure 3.2.

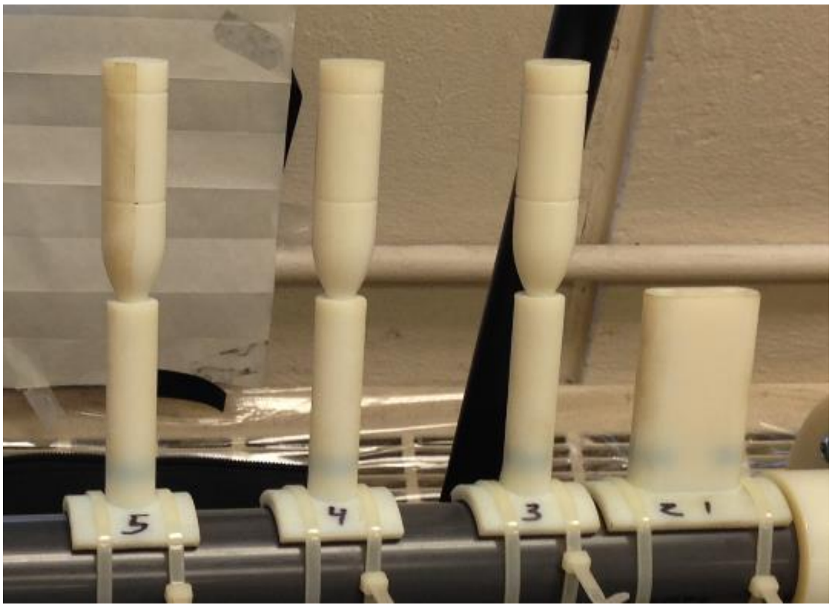


Figure 3.2 3D Printed Microphone Holders Attached to Multiple Microphone Positions

3.6. **Sample Holder and Back Plate**

The sample holder was designed in mind so that samples with varying thicknesses can be tested inside of the tube. To ensure that there are no gaps between the sample and the holder, modeling putty is used to fill these small cracks. This also provides a very tight seal. Two different back plates were constructed. The first back plate is made of solid aluminum that fits a sample of only 1 in. This is used to compare results between different tube setups. The second back plate is also made from aluminum but also contains a threaded rod so that the back plate can be adjusted for varying size test samples. The sample holder is attached to a flange which is connected to the end of the impedance tube. The ISO standards mentions that a gasket should not be used between the back plate and flange to provide an airtight seal. This is what is usually used when connecting typical PVC pipe flanges together. This is a problem because the gasket itself will absorb some of the sound during a test and can give results that are slightly higher for absorption coefficient. Although this is not mentioned in the ASTM standards this requirement was satisfied by inserting the gasket inside of the flange so that it never comes in contact with the specimen or inside of the tube. Vaseline can also be used to ensure an air tight seal.

3.7. Loud Speaker

Two different loud speakers were chosen for the operation of the impedance tube. The first loud speaker is the speaker Selenium DH200E that O'Malley, had suggested [46]. This speaker works adequately for frequency ranges about 500Hz but below that the results contain a lot of noise. To improve on this the Selenium D250-X was purchased. This loud speaker has provided a lot less noise at these lower frequency ranges. The end of the impedance tube was drilled and tapped to 1-3/8"-18 TPI so that both loud speakers could be screwed onto the end of the tube. A neoprene gasket on the end of the tube and Teflon tape on the threads were put in place to reduce any mechanical vibration to the microphones which could cause errors.

3.8. Signal Generator and Signal Processing Equipment

The signal generator and data acquisition are performed using National

Instruments Labview Signal Express. The software allows for different types of signals to be generated along with high and low pass filters if desired. The NI cDAQ-9174 is used along with the NI 9263 for signal generation to the loudspeaker and the NI 9234 for data acquisition of the microphone outputs. Once the microphone response is recorded, Matlab is used to calculate the transfer functions between the microphones. Then Matlab is also used to calculate the desired acoustic properties. The Matlab code used is included in appendix.

3.9. Four Microphone Design

The ASTM standards for the four microphone design are very similar to those of the two microphone design. The standards recommend the same requirements that have been shown above. The only difference is that the cap is replaced by another tube with microphone positions. It is as if a second two microphone tube is built and placed on the back side of the specimen. The microphones locations were chosen to match that of the two microphones. The only difference is that the distance from the first microphone holder to the end of the tube measures 3.6in instead of 2.6in. This allows for a 1in specimen to be mounted in the downward portion of the tube. If testing of larger samples is required then a new specimen holder would need to be built and placed in between the two sections of the tube. Figure 3.3 shows the two microphone with the downward end attached to create the four microphone tube.

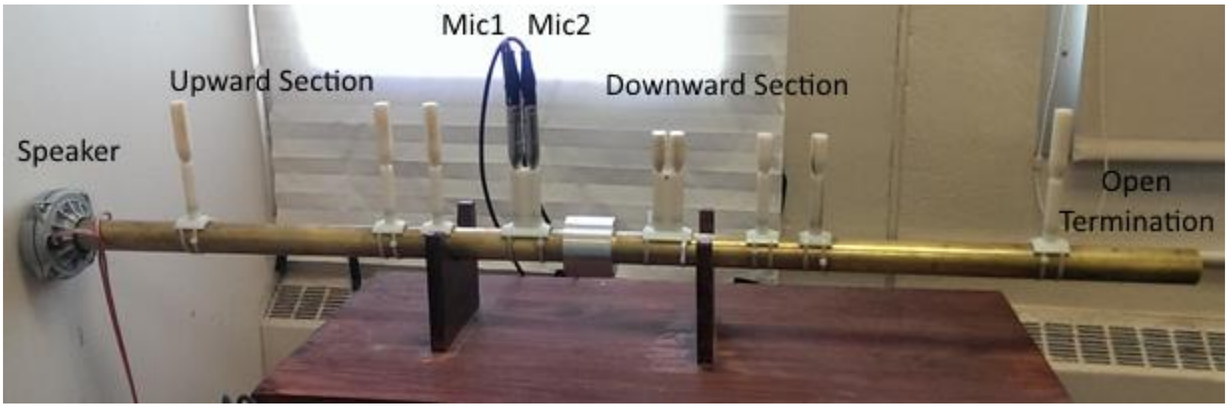


Figure 3.3 Four microphone tube with open termination and microphones positioned in locations 1 and 2 of upward section

The two sections of the impedance tube are connected by a coupling made out of aluminum. A coupling made from ABS was used previously but the weight of the brass tube would cause the joint to bend. This would create a gap between the two sections which lead to erroneous results. The termination of the tube depends on whether one load or two load method is used. For the one load method an anechoic termination was created by using a 28” piece of PVC containing fiberglass insulation inside of it shown in Figure 3.4. The two load method uses the anechoic termination along with the open termination. A blocked termination was tested but this caused noise in the results which can be attributed to the high reflections in the downward end of the tube.

Figure 3.4 Anechoic termination for four microphone tube lined with fiberglass

Since the lab only has two microphones, the first microphone was kept in the position closest to the speaker and four separate tests are run with the second microphone moving to the four positions sequentially. Since the second microphone is each of the four locations, there is no calibration step. The transfer function measured is between the second microphone and the first microphone’s data during the same test. Four microphones would allow for the setup to run quicker but would require four calibration factors between each of the microphones.

3.10. Three Microphone Design

Since there are no standards for the three microphone tube, there is little guidance on how to build this tube. Similar to the four microphone tube, the three microphone tube is an additional attachment to the two microphone tube. Figure 3.5 shows the three microphone attachment. The design in this thesis is unique because it uses only two microphones instead of three that most other designs use. Like the four microphone set up, one microphone is left at the position close to the speaker while the second microphone is positioned at each of the three locations sequentially. Since the microphone is required to move during the test the third horizontal location cannot be of the movable piston design. It would be very difficult to remove the microphone without altering the distance between the back of the sample and the microphone.

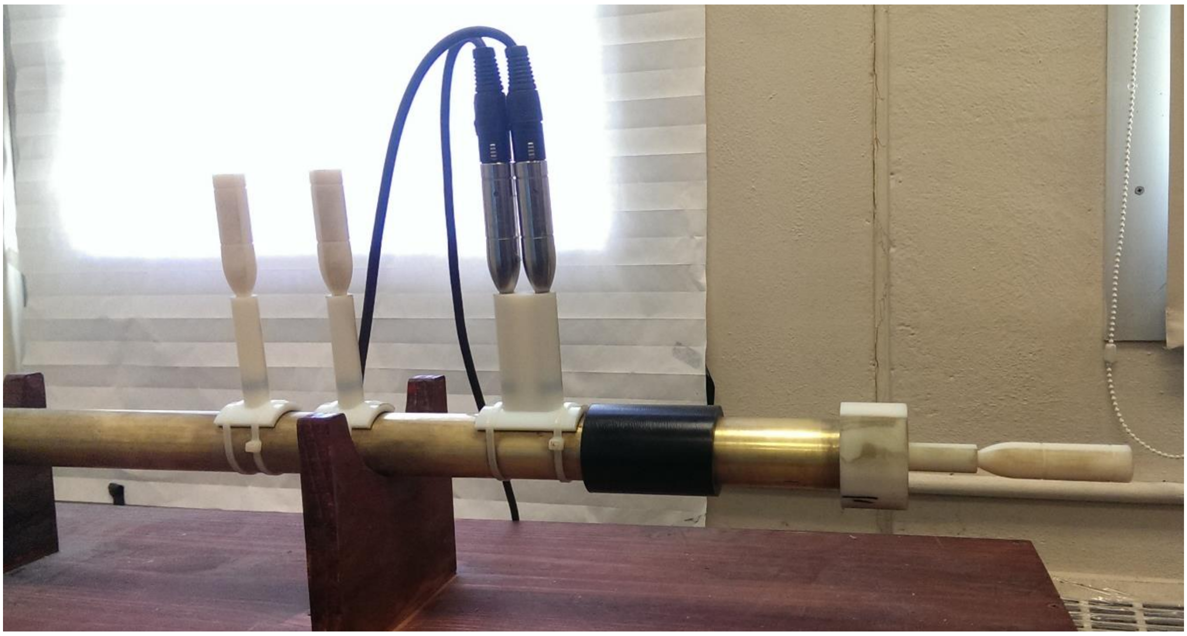


Figure 3.5 Three Microphone Tube

Two different lengths of brass tube sections were cut to act as the two locations that the microphone would be positioned if it was moved along the tube in the piston designed. The two sections were cut to a length of 4.25 and 5 inches. The microphone holder was designed from the other microphone mounts to include a gasket inside. Since the holder also acts as the cap for the tube, a block of aluminum was attached to the cap with a cut out of the microphones face to sit flush. Figure 3.6 shows the three microphone holder and cap.

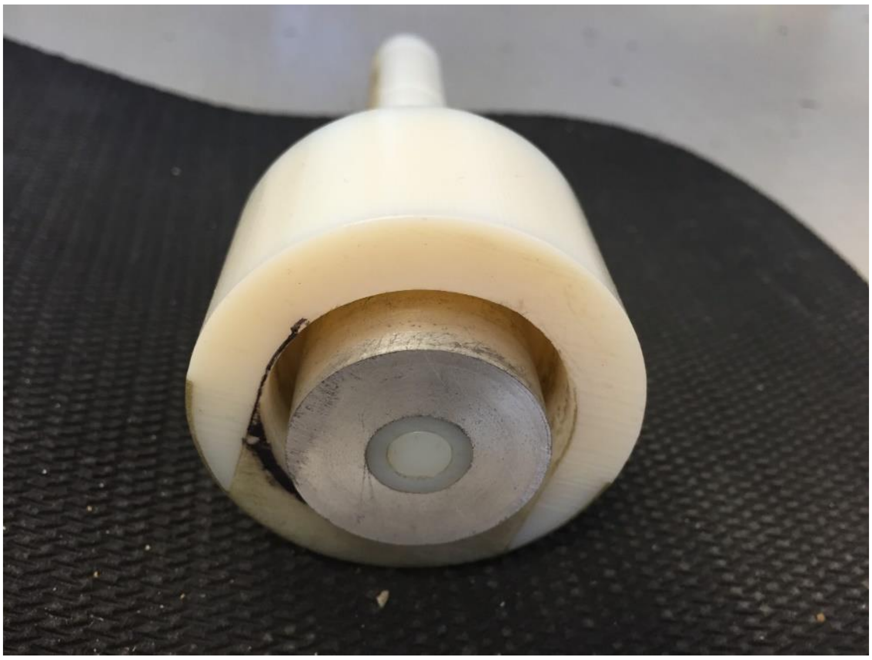


Figure 3.6 Three Microphone Holder and Cap

The actual distance from the end of the tube to the face of the aluminum back plate is measured using calipers before each test to ensure an accurate distance. The cap is used for both of the copper sections of tube. When inserting and removing the microphone from the microphone holder, extra care should be taken so that the cap is not moved. The sample can be put in the upward or downward section of the tube. This depends on how thick the sample is though. Additional sections can be made that are longer, if larger samples are needed to be tested.

Chapter 4.
Design of Experiments

Though standards ASTM E1050 and ISO 10534-2 provide ranges for microphone spacing, distance from microphone to sample, and speaker volume, which the values differ by spacing and which standard used, greater than 0.75 in for a flat specimen, and volume greater than 10dB respectively, the hypothesis of this thesis is that variable of the variables within the specified ranges will lead to variability in the measured results. A design of experiments was designed and conducted to determine how these design parameters affect the measured acoustic properties. The parameters that were thought to have the most impact to the design are provided in Table 4.1. Figure 4.1 shows the impedance tube system with the inputs and outputs along with the parameters that are controllable and uncontrollable for this design.

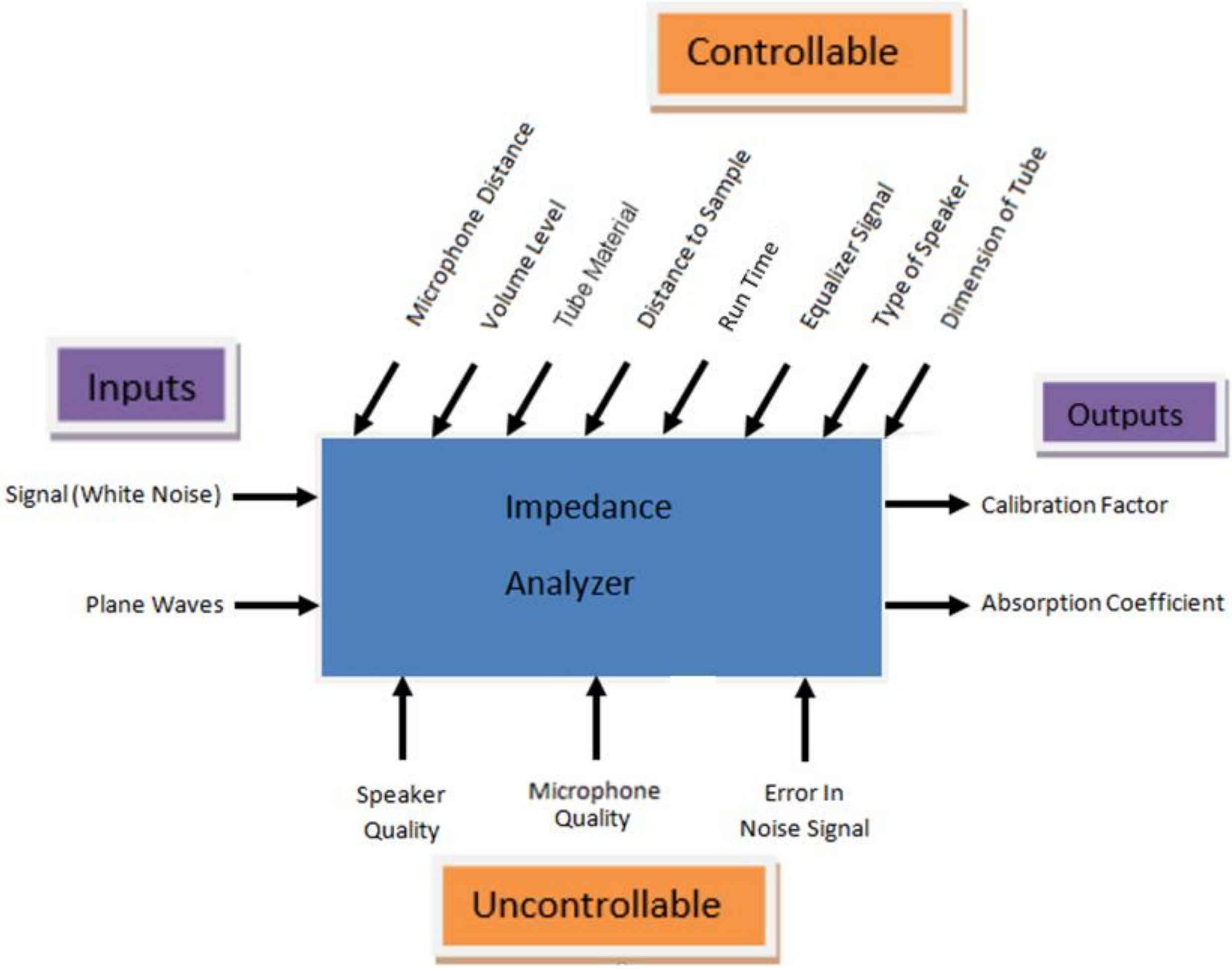


Figure 4.1 Impedance Analyzer System Diagram

The system diagram shows the controllable parameters of the tube that can be changed. Each of these parameters will be changed to show their effect on the absorption coefficient. The test factors and designed runs are shown in Table 4.1 and 4.2. Additional tests were added to the distance to sample and run time tests then the three that are shown in Table 4.2 since they took minimal time to run but are not easily displayed on the table. A polyurethane sample and cotton sample were first tested to examine which provided better results as the microphone calibration sample.

As expected it was found that the higher absorption coefficient cotton sample provided a better calibration like the standards recommend. For this reason all further results shown are with a microphone calibration that was performed using the cotton sample.

Factors	Label	Option 1	Option 2	Option 3
Tube Material	M	PVC	Steel PVC	Brass
Volume Level	V	Low	Medium	High
Distance to Sample	D	Stock	Farther	Nearer
Run Time	R	.1 sec	1 sec	10 sec
Speaker Type	S	Rectangle	Gaussian	Triangle
Signal Equalized	E	On	On	-
Speaker Type	SP	D250-X	DH200E-E	-
Specimen	N	Polyurethane	Pine	-

Table 4.1 Design of Experiment Test Options

Run	M	V	D	R	S	E	SP	N	Note
1	1	1	1	1	1	1	1	1	Stock All Option 1
2	1	1	1	1	1	1	2	1	Speaker Change
3	1	1	2	1	1	1	1	1	Distance Change
4	1	1	3	1	1	1	1	1	Distance Change
5	1	1	2	1	1	1	1	2	Distance Change with Pine
6	1	1	3	1	1	1	1	2	Distance Change with Pine
7	1	2	1	1	1	1	1	1	Medium Volume Level
8	1	3	1	1	1	1	1	1	High Volume Level
9	1	1	1	2	1	1	1	1	1 Second Runtime
10	1	1	1	3	1	1	1	1	10 Second Runtime
11	1	1	1	1	2	1	1	1	Old Speaker
12	1	1	1	1	1	2	1	1	Equalizer Off

Table 4.2 Design of Experiment List of Runs

As a step before the design of experiments, the effect of the microphone spacing on absorption coefficient was studied. As a datum Table 3.2 created from the ASTM standards for the microphone distances (provided in Chapter 2) showed that the upper frequency was highly dependent on the microphone spacing.

4.1. Lower Frequency Limit of Standards

Looking at Table 3.2, it was seen that the values for the lower frequency limit calculated for each microphone spacing differed by a factor of 5 between the ASTM and ISO standards. To test the effect of different microphone spacing's, a tube with 8 microphone holders was built and the polyurethane foam sample was tested. Figure 4.2 shows the curve for all seven microphone position tests sectioned into one curve. The lower frequency for each position was chosen to coincide with the upper frequency of the next microphone positions upper frequency limit.

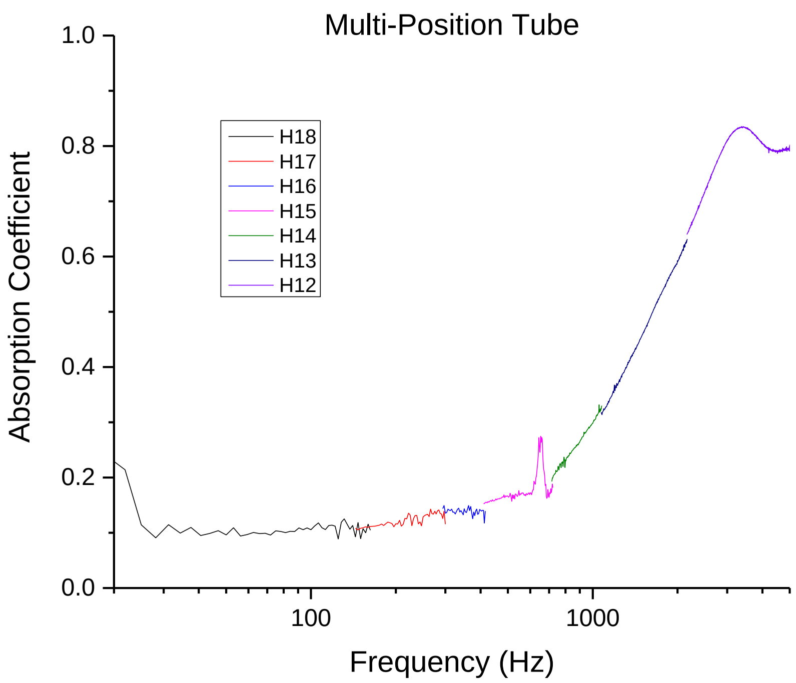
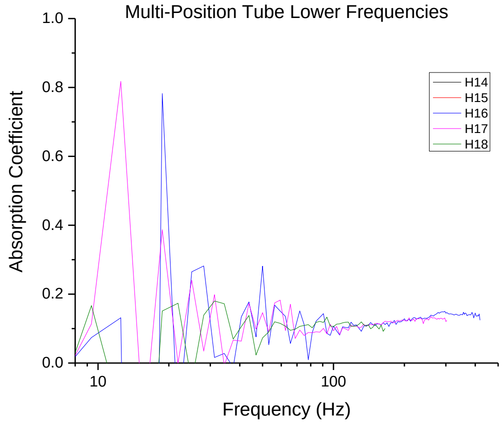


Figure 4.2 Multi-Position Test with Polyurethane Sample on Logarithmic Scale

The distances of H12, H13, H14, etc. coincide with the values highlighted in Table 3.2. H12 corresponds to the smallest microphone spacing while H18 is the largest. As you transition though each microphone position, the curve continues with minimal discontinuity. H15 to H16 shows the most discontinuity which can be attributed to being too close to the half wavelength distance that the standards are attempting to prevent. This can be removed by extending H15 to a lower frequency since it can still be used for those frequencies. The spike in H15 shows up on every curve up to H12. This spike was due to one of the microphones needing to be fixed. Once the microphone was fixed this spike was removed from further results. Figure 4.3 a,b show the results for the lower frequencies at larger microphone positions.



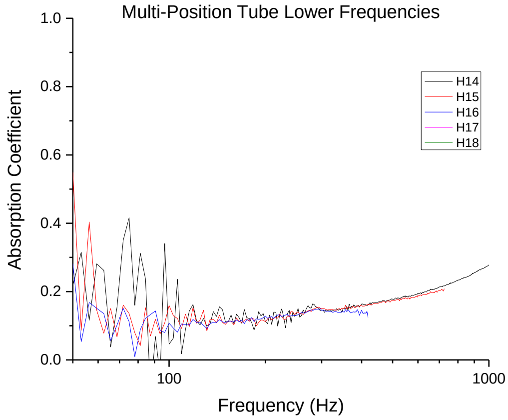


Figure 4.3 a,b Low Frequency Multi-Position Test with Polyurethane Sample on Logarithmic Scale

Figure 4.3a shows that below around 50Hz the results are inaccurate. This is not due to the microphone spacing however. This is due to the limitations in the driver that is being used. In general drivers are not very good at such low frequencies. To improve in this frequency range subwoofer would be better. When looking above 50Hz it can be seen that the noise is reduced as the microphone spacing increases. This is as expected since the larger spacing provides more accurate results at lower frequencies. When looking at Figure 4.3b it can be seen right away that H14 at frequencies below 120Hz does not agree with the other positions and also shows increasing noise below 300Hz. The ASTM standards for this microphone spacing, state that the lower frequency limit should be at 27 Hz which these results contradict. The ISO standards state that the lower frequency limit is 135 Hz. This value much closer to the 120Hz limit that was observed. To further look at the comparison between ISO and ASTM lower frequencies limits, Figure 4.4 shows a higher frequency range.

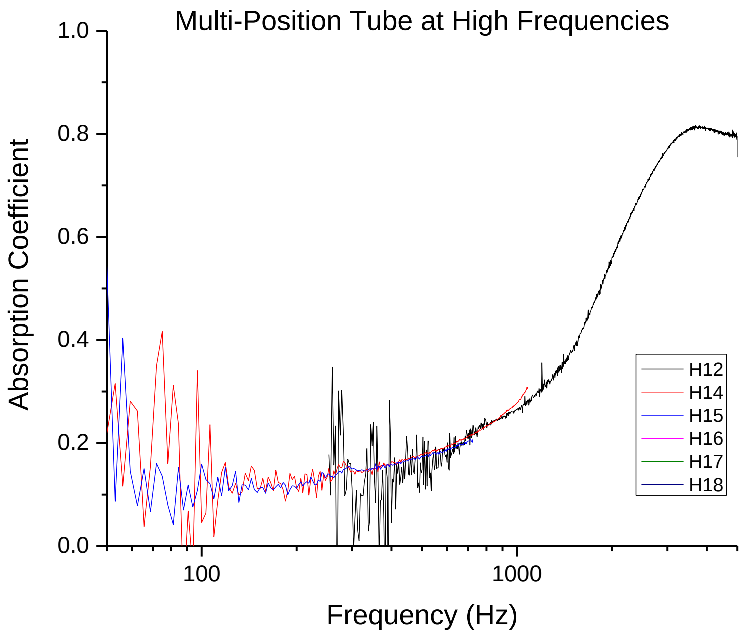


Figure 4.4 High Frequency Multi-Position Test with Polyurethane Sample on Logarithmic Scale

From Figure 4.4 it can be seen that the curve for H12 begins to deviate from H14 and H15 around 700Hz. The data below 250Hz was removed since it made it difficult to view the lower frequencies. The lower limit calculated for this microphone spacing is 158Hz for the ASTM standards and 791 Hz for the ISO standards. This shows again that the ISO standards provide a better lower limit for microphone spacing. The upper frequency limit was not studied since the two standards are within 10% of each other and no definitive conclusion can be made as to which value is better.

4.2. Tube Material and Length

When first testing the design of the two microphone impedance tube there were questioned to whether building the tube out of conventional PVC would affect the results. PVC is not perfectly straight, especially over long distances. While machining tubes out of PVC it was found that over seven feet the PVC can have a displacement of as much as an inch. Since the ASTM standards requires the tube to be perfectly straight, there was a hypothesis as to whether this was a critical factor in building an impedance tube. The ASTM standards also note that the interior of the walls must be perfectly smooth. The inside of the PVC tube showed slight imperfections in the wall because it is not molded as a perfect circle. Although the ASTM standards allow for plastic tubes, the ISO standards do not. It was possible that PVC is not dense enough to be used as a building material. A denser metal would be better at preventing acoustic loss through the tube wall. To test the effect of the tube's material on the absorption coefficients, a tube was built out of PVC, brass, and a PVC tube wrapped with sheet of stainless steel. The ASTM standards suggest wrapping a less dense material with a denser material like steel will reduce attenuation losses. As another test, to show the effect of sound attenuation caused by the length of the tube, several tubes were built and tested out of PVC cut with different lengths between the speaker and the first microphone. Figure 4.5 shows the absorption coefficient of a polyurethane foam sample in the three different tube materials.

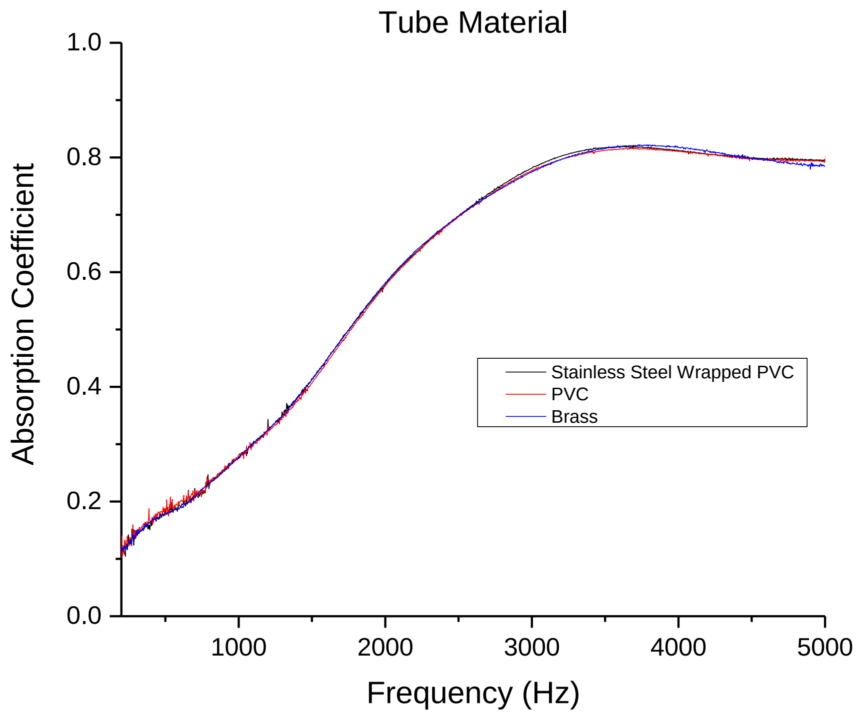


Figure 4.5 Tube Material Effect on Absorption Coefficient for Polyurethane Foam from 200-5000Hz

The three tubes are in agreement with each other over the entire frequency range of interest. There are small differences at higher frequencies and there seems to be more noise at lower frequencies. To view these differences the lower and upper frequencies are plotted again in Figure 4.6 a,b.

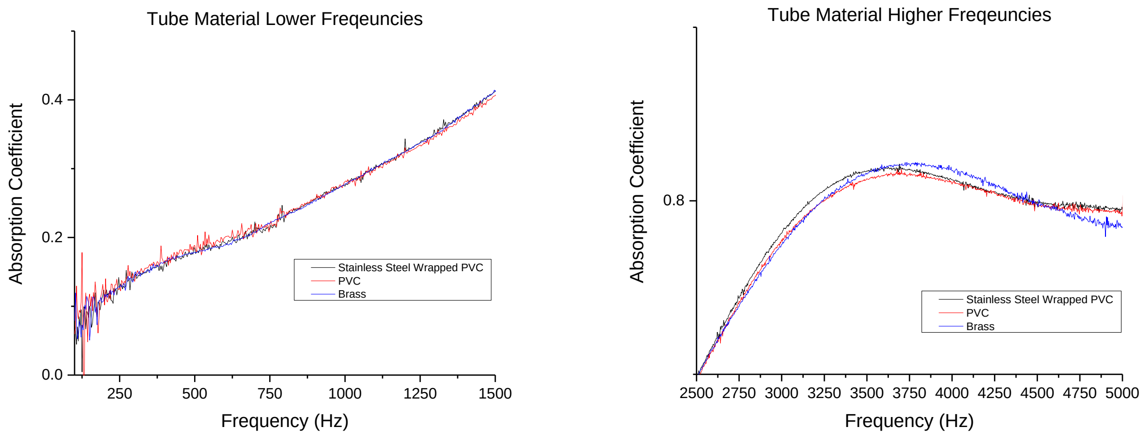


Figure 4.6 a,b Tube Material Effect on Absorption Coefficient for Polyurethane, Low and High Frequencies

It can be seen from Figure 4.6a that at low frequencies the three curves agree with each other. What is different about the curves is that the tube made from brass shows the least amount of noise in the results. The stainless steel wrapped PVC tube is the second best in noise while the PVC tube shows the most noise. This noise can be attributed to the attenuation loss, since the PVC less dense. The stainless steel wrapped tube's noise could be from the imperfections in the surface of the interior wall of the tube. This has less of an affect then that of the attenuation loss from the material.

When looking at Figure 4.6b, the differences between the three curves are apparent once magnified. The variation between the three curves is less than a maximum of 1% difference. Although this is very low the differences between the three curves can be attributed to a combination of both the interior wall and the attenuation loss through the tube wall.

4.3. Position of Samples

The ASTM and ISO standards recommend specific distances from the face of the specimen to the first microphone. This distance depends on the type of surface the material has. A perfectly flat surface can be a minimum of half the diameter of the tube. This distance increases to as much as twice the diameter if the specimen is asymmetrical. In general, the samples tested in this paper have a flat surface. A test to measure the effect that specimen distance causes on the measured absorption coefficient was performed for a polyurethane foam sample along with a sample of pine. The results for the polyurethane sample are shown in Figure 4.7.

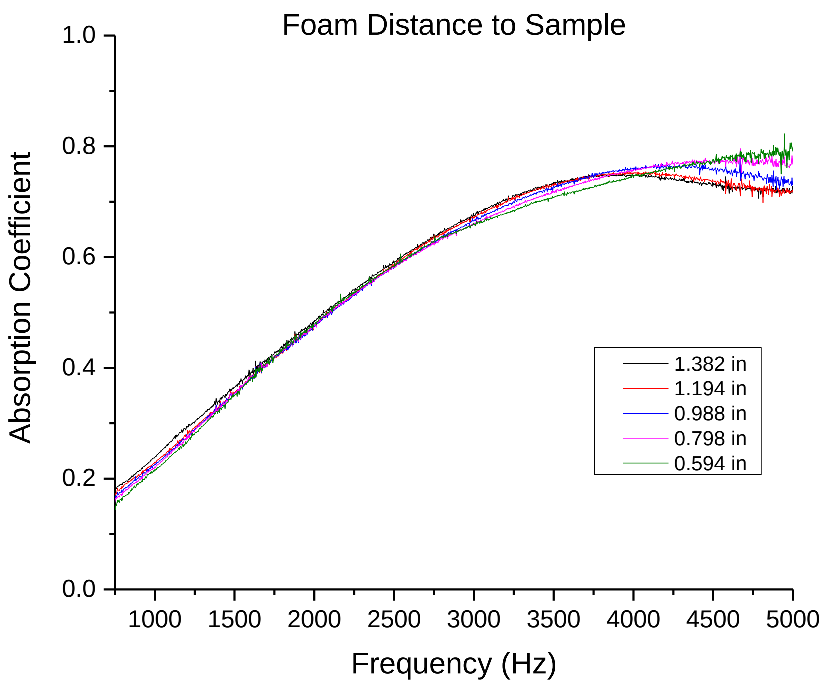


Figure 4.7 Sample Distance Effect on Absorption Coefficient for Polyurethane Foam

For an impedance tube diameter of 1.25 in, a sample with a flat surface should be able to be placed 0.75 in away from the first microphone. For the polyurethane sample, the absorption coefficient should match at for all each distance above 0.75 in. When looking at Figure 4.7 it can be seen that although 0.798 in is further than the required distance, the absorption coefficient at larger frequencies is higher than when the samples is placed further away. Even the distance of 0.988 in has an absorption coefficient higher than the distances above 1in where the values match. This means either the distance recommended by the standards is incorrect or this foam sample is not considered to have a flat surface. The distance may not be far enough for planes waves to fully develop. This implies that when building an impedance tube, consider placing the samples further from the first microphone, as to avoid inconsistencies in results. The effect that distance away from the microphone on a pine wood sample was also tested. The results are shown in Figure 4.8 a,b.

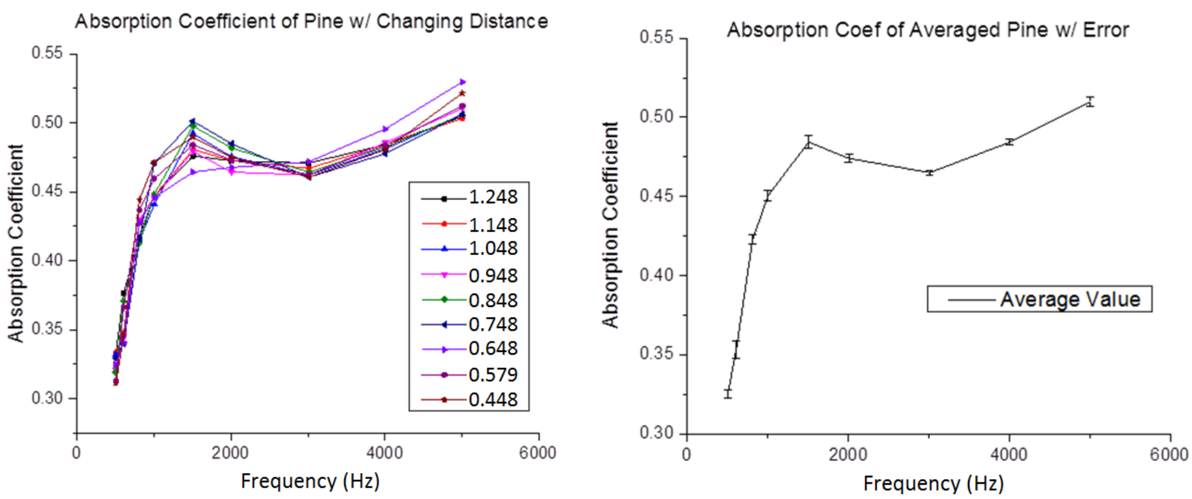


Figure 4.8 a,b Sample Distance Effect on Absorption Coefficient for Pine

The pine sample also shows variations in absorption coefficient with the change in distance from microphone to sample. Unlike the polyurethane sample, there are variations throughout the lower frequencies as well as the higher frequencies. Like the polyurethane sample at high frequencies, the three closest positions showed the highest rise in absorption coefficient. At around 1800Hz the absorption coefficient varies with little direct correlation to the distance but the largest variation is the second closes position. At most the error is in the range of 2.5%. Like the polyurethane sample, the variation in absorption coefficient can be attributed to plane waves in the tube not fully developing. The pine sample shows more variation, especially at lower frequencies because it is denser and less absorptive. This will in turn cause the sample to reflect more.

4.4. Signal, Equalizer, and Speaker

Another important area to look into when designing the impedance tube is how the actual signal/sound wave is created. This is can be affected by the actual signal that is created and how the signal is interpreted by the speaker. Since the signal created does have a uniform spectral density, ideally the actual sound wave created from the speaker should match. This is usually not the case because the speaker's response is not uniform and will depend on frequency. To fix this an equalizer may help in adjusting the response so that the sound wave will have a uniform spectral density. The test signal was tested as rectangle, Gaussian, and triangle white noise. Sine sweep was also tested but the result output was erroneous so it was not included. An actual different speaker may also affect the results because their responses will be different. The signal is the first topic that will be explored and the comparison between the three signals is shown in Figure 4.9.

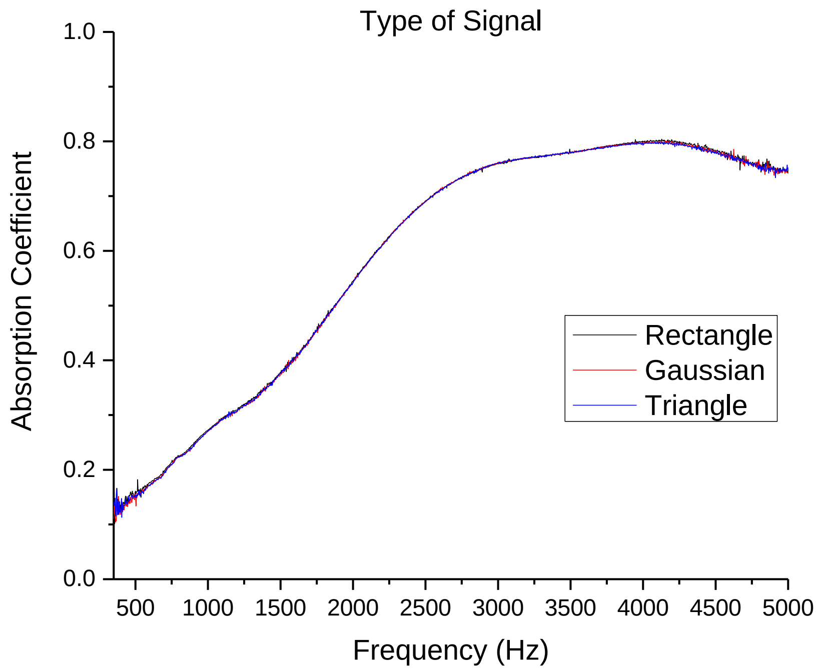


Figure 4.9 Signal Type Effect on Absorption Coefficient for Polyurethane

The rectangle and triangle white noise signal agree with each other the frequency band of interest. The Gaussian white noise signal show slightly more noise around the upper and lower frequency bands which is caused by the signal now being uniform spectral density. Although this is a requirement of the ASTM and ISO standards, it was tested to show discrepancy the type of signal would have on the absorption coefficient however minimal it is. The rectangle white noise signal was chosen to be used in all other testing because it showed less noise at lower frequencies. Figure 4.10 shows the effect of using an equalizer on the signal to form the signal into a uniform spectral density.

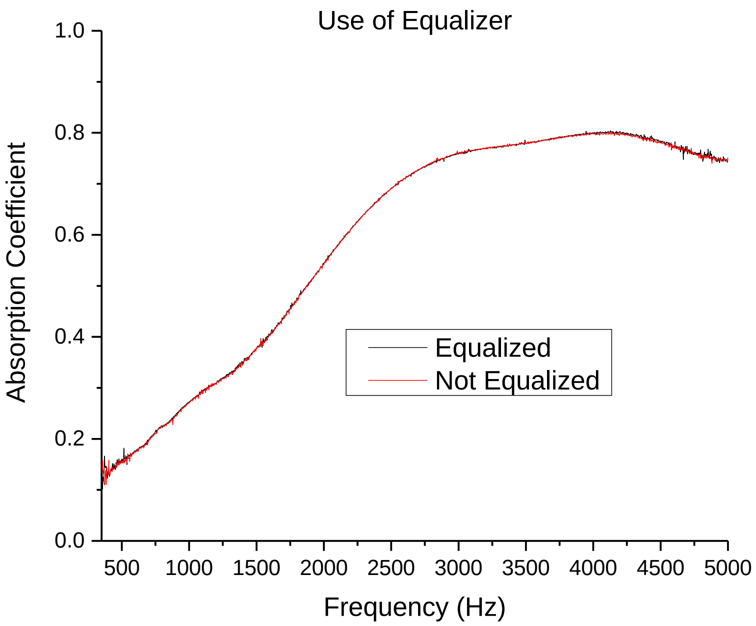


Figure 4.10 Equalizers Effect on Absorption Coefficient for Polyurethane

Looking at Figure 4.10 it can be seen that the results from using an equalizer show little difference from those without the using the equalizer for most of the frequency bandwidth. It does however help slightly with lower frequencies. This is due to the speaker not having a good frequency response at lower frequencies. Increasing the amplitude at lower frequencies using the equalizer makes sure the signal to noise ratio at these frequencies is high enough to achieve better results. The benefits to using an equalizer are only minimal though, so using one with the configuration is not required.

The original speaker chosen to run tests in the impedance tube was the Selenium DH200E-E. This speaker showed good results at higher frequencies, but at lower frequencies noise was found. The Selenium D250-X was purchased because it had a recommended lower frequency limit of 500Hz compared to the DH200E-E's 1,500 Hz. The comparison between the two types of drivers is shown in Figure 4.11.

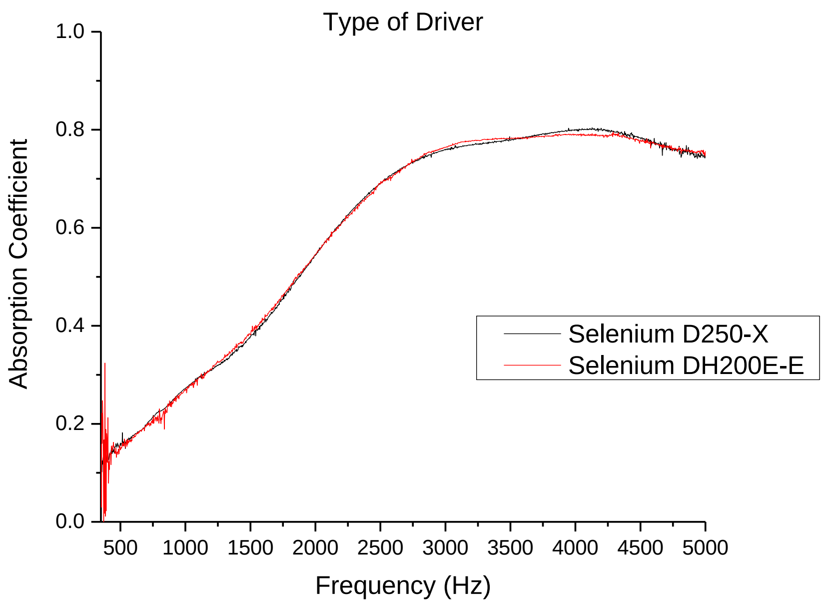


Figure 4.11 Speakers Effect on Absorption Coefficient for Polyurethane

Looking at Figure 4.11 it can be seen the both speakers exhibit the same general curve with small variations in-between the range of 2,500 Hz to 4,500 Hz. This may be caused by the fact that the DH200E was much louder outside of the tube. It could also be imperfections in the speaker itself since the D250-X has a higher build quality than the DH200E. At the lower end of the frequency range for the DH200E, the noise increases around 1,000 Hz and continues to the lower frequencies before it becomes too large to discern the actual curve. As expected the D250-X shows better performance at these lower frequencies. For these reasons the D250-X was used for test runs used inside the tube.

4.5. Run Time and Speaker Volume

The final tests ran to test the design of the tube were on the test run time length and on the speaker volume. The standards describe a time length that only needs to be greater than a few seconds. During initial testing the question arose to whether increasing the run time length would help improve lower frequency noise. To test this several runs were performed with varying sample time length. This test is shown in Figure 4.11.

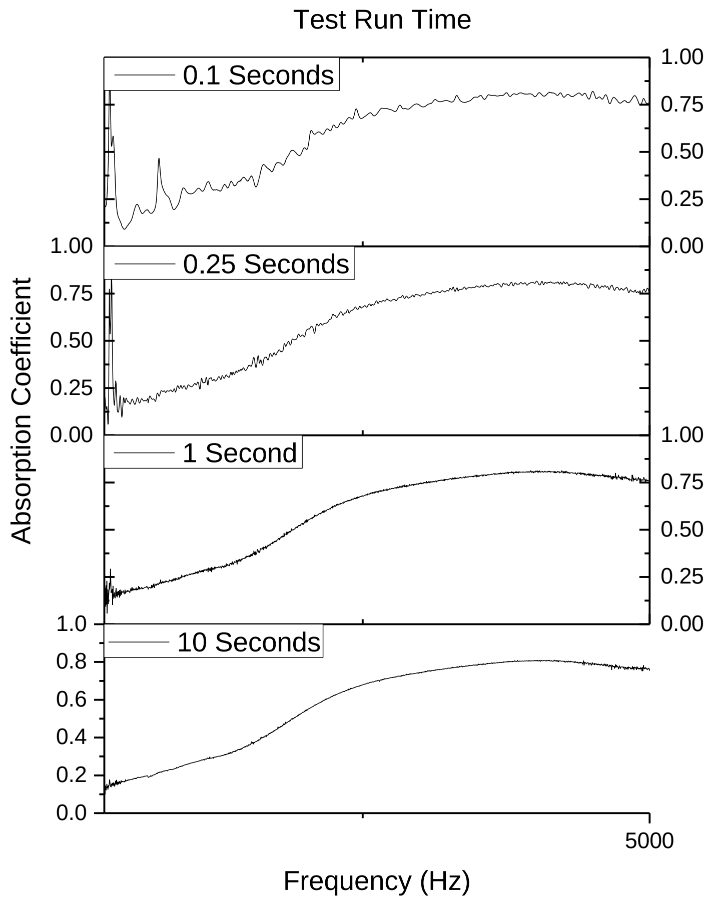


Figure 4.12 Run Time Effect on Absorption Coefficient for Polyurethane

It can be seen from the Figure 4.12 that the even at only 0.25 seconds are drastically better than those from 0.1 seconds. The major peaks and noise have been smoothed out. As the time increases to 1 second most of the noise has been removed. When increasing the time from 1 second on provides somewhat of a diminishing returns. At 10 seconds the noise at lower frequencies is removed. Longer than 10 seconds did not show any noticeable change between absorption coefficient, so it was not shown. Since the acoustic properties are calculated using transfer functions, lengthen the run time acts as averaging the data which in turns removes the noise from the curves. Subsequent and further tests performed on the tube will be performed at 10 seconds because of the less noise at lower frequencies.

The volume of the speaker can also have an effect on the acoustic properties of materials tested inside of the tube.   Several tests were performed at speaker voltage of 10dB to 28dB. Below 10mV, the signal to noise ratio too low and provided erroneous results. Tests were also not performed above 28dB for fear of damaging the speaker. The results are shown in Figure 4.13.

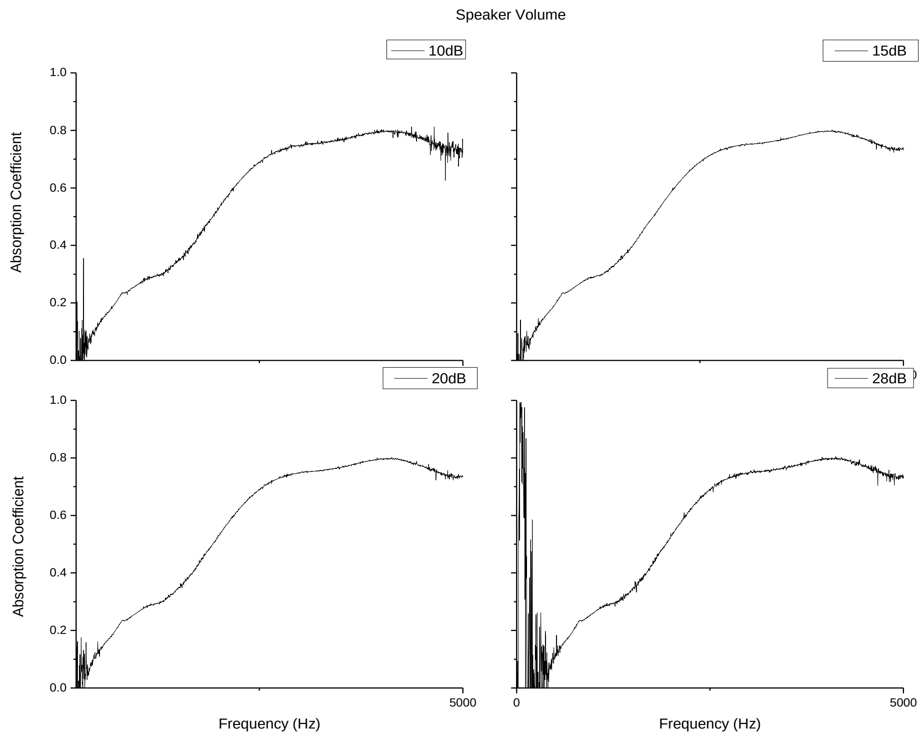


Figure 4.13 Speaker Volumes Effect on Absorption Coefficient for Polyurethane

Looking at the four cases above, it can be seen that each shows the general same absorption coefficient curve. The different comes down to which one has the least noise. When looking at the extreme cases each shows more noise than the other two curves. At 10dB the noise is just caused by not having a high enough signal to noise ratio. At 28dB the noise is caused by over saturating the microphones. This will cause microphone clipping which will cause both error and could damage the microphones. Between 15 dB and 20 dB they are very similar, but the 15 dB provides less noise at low and high frequencies. 15 dB is used as the volume level for both subsequent and further tests run. Using the results from this section, different materials can now be tested.

Chapter 5.
Tube Comparison Results

The design of experiments provided a standard for test parameters when running the tube for material testing. Keeping these parameters consistent between tests removes the possibility for errors. Using these parameters several different materials were chosen to be tested. The materials were chosen so that a wide range of absorption coefficients was tested. Table 5.1 contains the materials tested.

Table 5.1 Tested Materials and Size

Material Type	Size (inches)
Melamine	0.776
Denim	1.150
Fiberglass	0.925
Acrylic	0.993
Pine	0.973
Green Foam	0.995
Polyurethane	0.926
Foam Matrix	1.000
Cotton	2.001

The denser samples were cut using a hole saw, and then sanded to a diameter of

1.25 inches. All other samples were cut using a hole punch. Care must be taken when using both techniques because the hole punch can cause the samples to be trapezoidal and be smaller than 1.25 inches. Samples cut using the hole saw need to be carefully sanded so that the edge is perfectly round. If the samples are not sanded correctly they will cause gaps between the samples edge and the tube wall which can cause errors in the measured acoustic properties. The cotton sample was made from compressing three cotton balls together. The cotton sample was used during the calibration procedure. Figure 4.1 shows each test sample.

Figure 5.1 Samples Tested (from left to right, Foam Matrix, Polyurethane, Melamine, Green Foam, Cotton, Fiberglass, Denim, Pine, Acrylic)

The samples were chosen to be approximately 1 inch so that they could be compared to one another. Since not every material came in this size, some of the samples are a different size. The two microphone tube is the first tube to be tested with each of these materials since it is used in both the four and three microphone set up. It was first calibrated using the cotton sample and then each of the samples were tested. The results are shown in Figure 5.2.

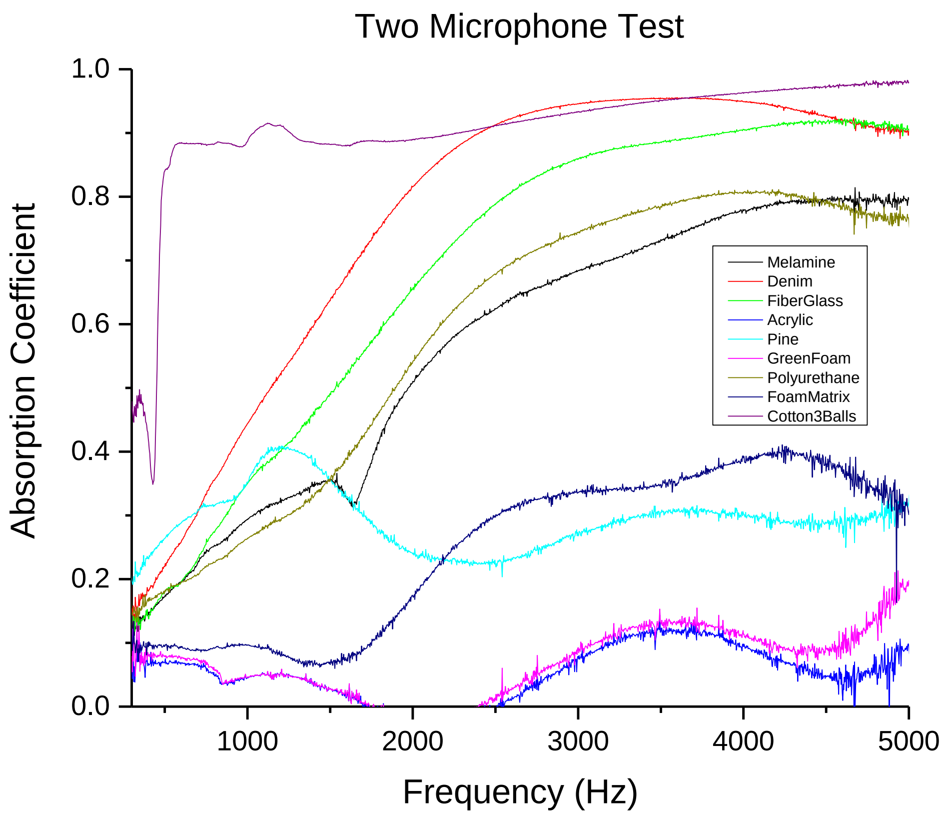


Figure 5.2 Two Microphone Sample Test

Looking at Figure 5.2 it can be seen that the cotton has the highest absorption coefficient over the desired frequency range. It is also about double the thickness of the other materials so it makes sense that this is true. Melamine which is known for its high acoustic absorption properties performs almost equal to that of polyurethane even though it is not at think. Figure 5.3 shows the sample of melamine compared to a manufactures specification on its melamine properties. The acrylic and green foam sample both had very low absorption coefficients. They also have a negative absorption coefficient around 2000Hz which is not possible. This error is caused by both samples having a highly reflective surface. This would cause the closer microphone to pick up a distorted sound. It is possible that moving the sample further from the microphone would allow for plane waves to develop and remove this error. Looking further into the melamine sample, Figure 5.3 compares the sample to one approximately the same thickness manufactured by BASF. Our sample of melamine was not manufactured by BASF, so there could be some differences in the actual material.

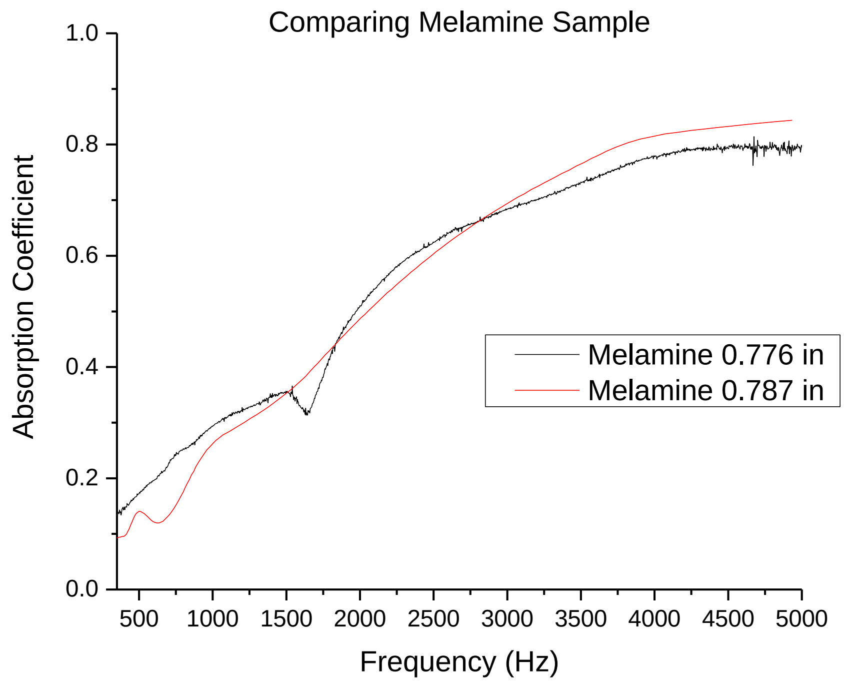


Figure 5.3 Melamine Comparison [23]

Looking at Figure 5.3, the 0.776 in sample is the melamine sample we tested and the 0.787 in is what BASF stated the acoustic properties of a 20mm sample. The two samples agree well with each other. At lower frequencies the BASF sample has a lower absorption coefficient while it has a higher absorption coefficient at higher frequencies. The differences can be attributed to the fact that this isn't a BASF sample. Testing one of their samples would most likely show more comparable results. This does however show that the results from our melamine sample has an absorption coefficient comparable to one of a similar material in the same frequency range.

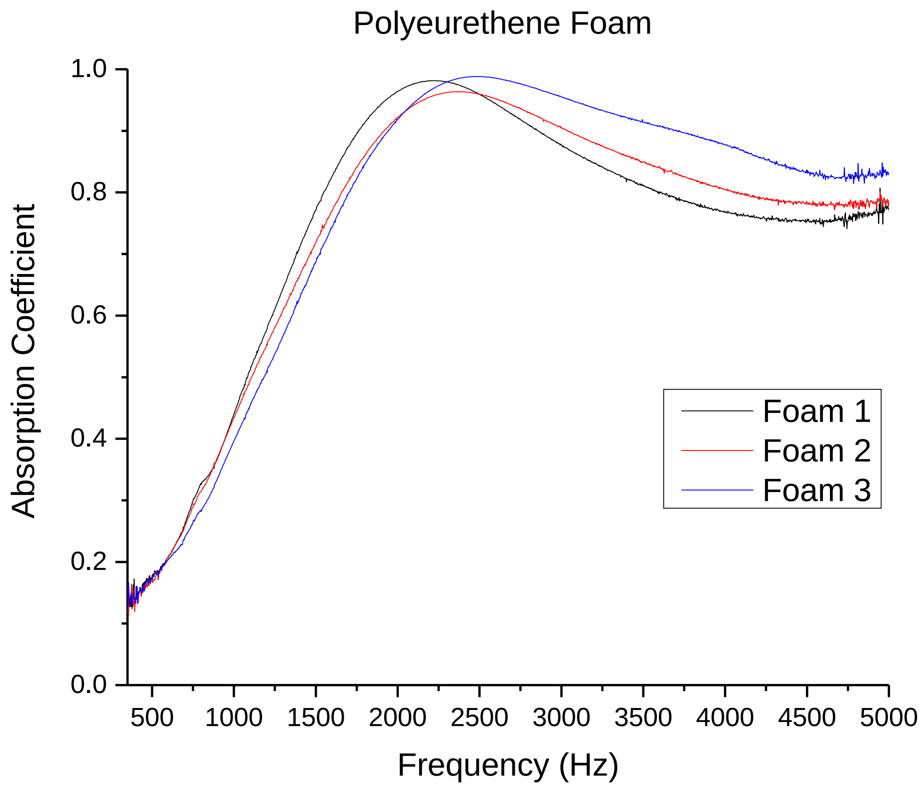


Figure 5.4 Three Different Samples of Polyurethane Foam

The ASTM standards require that at least three test specimens be fabricated and tested to average the results together. Figure 5.4 shows why this is necessary. Although all three of the Polyurethane foam samples are the same thickness, each sample has a different absorption coefficient peak. At higher frequencies one of the samples has a much larger absorption coefficient compared to the other two. This is why three samples should be tested but it would probably be better to test more if it is possible. Next, a few of these materials will be tested with the four microphone tube.

The results are shown in Figure 5.5. Not every specimen was tested since the four microphone set up takes much longer to run with only two microphones.

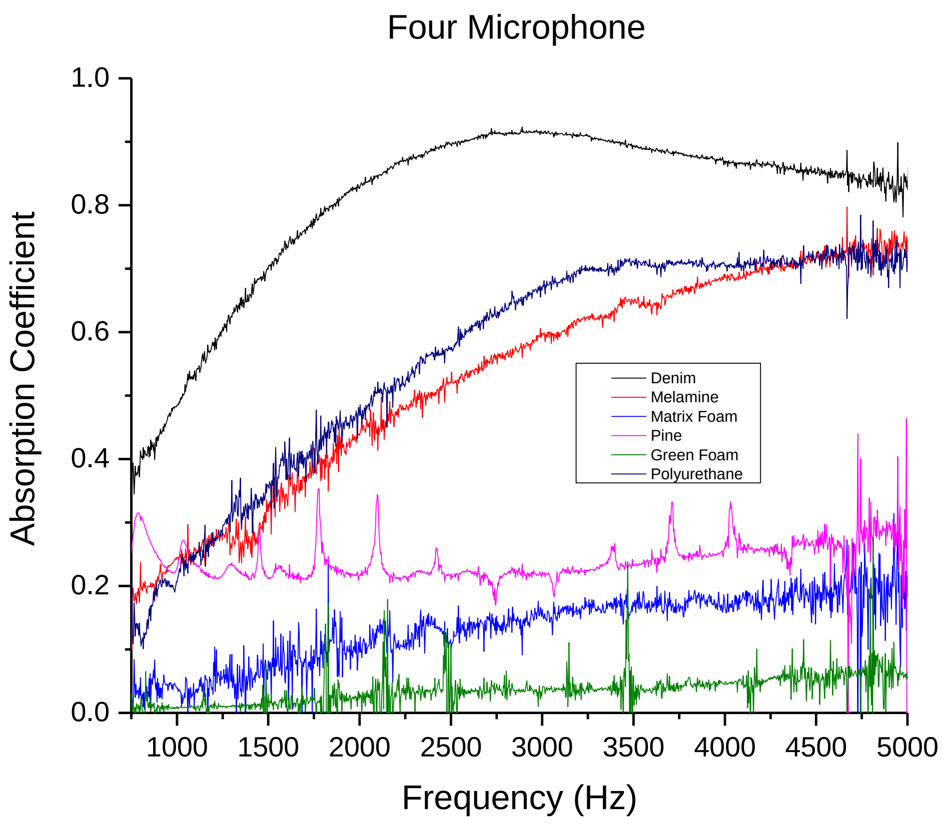


Figure 5.5 Specimen Test with Four Microphone Tube

The denim sample shows the highest absorption coefficient as expected. This was seen in the two microphone tube and it also one of the thicker samples tested. The only other sample that was thicker was the cotton sample which would have the highest absorption coefficient. The sample was tested using the three microphone tube for comparison. The melamine, pine, and matrix foam samples, each show noise which looks like oscillations. It was determined that this was caused by a slight mismatch in the tubes center between the two sections. The inner diameter of the tube is not centered at the same location as the outer diameters center. This causes there to be a mismatch in the inner wall of the tube when the two sections are joined together. This problem could have been prevented if known when fabricating the tube because as long as the two sections are aligned when the microphone holes are drilled, there will be no mismatch in the inner wall. Unfortunately this was not known and there is no way for this to be fixed unless a new four microphone section is fabricated. The impact of the mismatch can be reduced by placing the sample slightly over the crack that separated the two sections.

The pine and green foam samples have large spikes in their curves. These spikes are most likely caused by the back end microphones picking up noise in the tube. Since both samples prevent a majority of the sound from passing through, the back end microphones do not measure much if any of the sound wave. These microphones measure any noise on that end of the tube caused by the environment. These spikes can also be seen when studying the transmission loss of the same samples. The measured transmission loss of these samples can be seen in Figure 5.6.

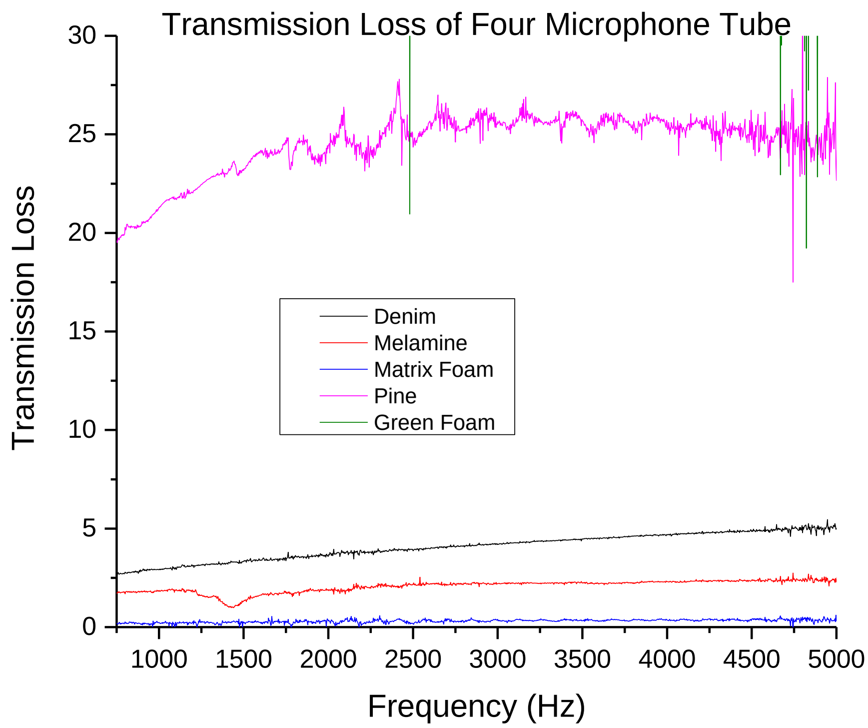


Figure 5.6 Transmission Loss of Four Microphone Tube

The transmission loss of the denim, melamine, and the matrix foam samples are each have an acceptable transmission loss. Since denim had the highest absorption coefficient out of the three samples, it makes sense that the transmission loss would be the same. The matrix foam sample was not a very good absorber of sound and its very large openings allow a majority of the sound wave to pass right through the material. This is why the matrix foam sample has the lowest absorption coefficient. What is interesting about Figure 5.6 is when you look at the curves for the green foam sample and pine sample. These two samples have the highest transmission loss out of the samples tested. The green foam sample has a very reflective surface, which does not allow the sound waves pass through. The pine sample is very dense and also prevents the sound from passing through. Oscillations also shows up in the transmission loss curves which can be attributed to the noise in the two microphones in the back end of the tube. The curves for both samples actually match either which reinforces the notion that the microphones are reaching the point where they can only measure noise. Since this is a result of the material the only ways to prevent this would be to increase the volume, decrease the thickness of the samples, or somehow reduce the noise experienced by the microphones.

Like the four microphone tube, the three microphone tube can be used to measure both the absorption coefficient and the transmission loss. The absorption coefficient of specifically selected samples is shown in Figure 5.7.

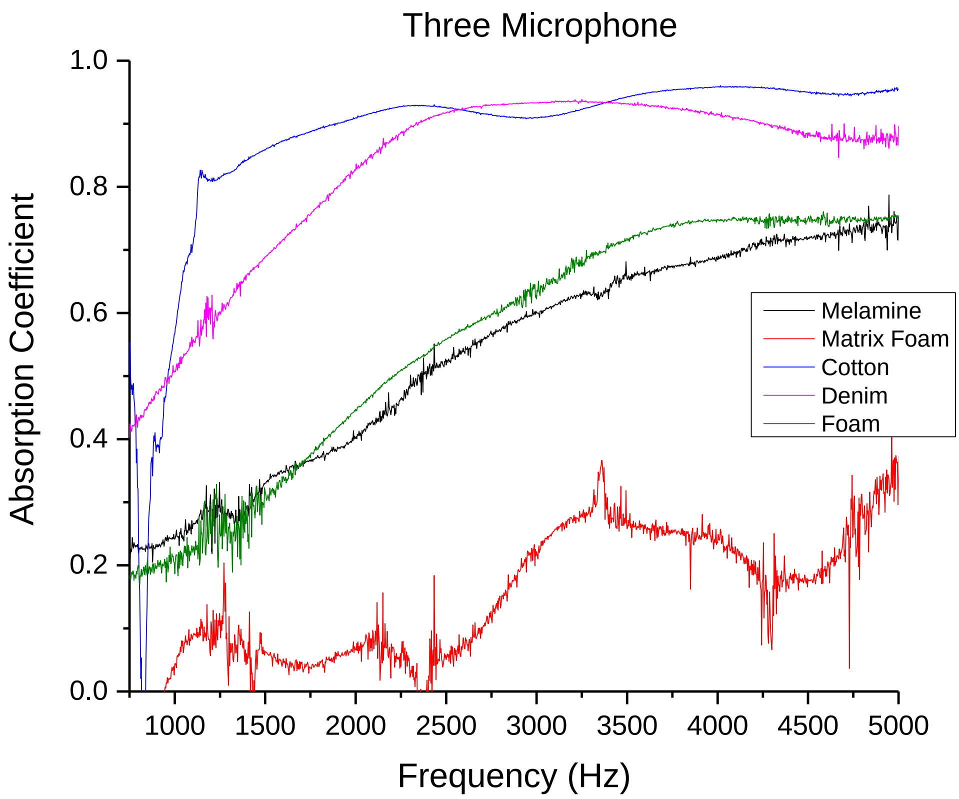


Figure 5.7 Absorption Coefficient using Three Microphone Tube

When studying the curves in Figure 5.7, it can be quickly noticed that the cotton sample and denim sample continue the trend of having the highest absorption coefficient. Then followed by the foam and melamine sample. The matrix foam sample shows the most amount of noise over the entire curve. This is most likely due to both the high reflective aluminum backing around the third microphone. Since in the three microphone tube, the third microphone does not require a hole in the tube wall. This means that the mismatch in tube center is able to be fixed by lining up the first and second section together as when it was a solid brass tube. There is also noise seen at around 1250 Hz, that can be seen in every curve but the cotton sample. Since it affects most of the samples the error is not material dependent and must be caused by the tube. The third microphone mount can also be assigned to this since the high absorption material cotton, is not affected which could be because less sound passes to the back end. More samples would have been tested but the three microphone holder broke and the easy fix of gluing caused the tube to measure erroneous results. A new microphone holder is required if the three microphone tube is going to be used. The transmission loss for what materials were tested before the microphone holder broke can be seen in Figure 5.8.

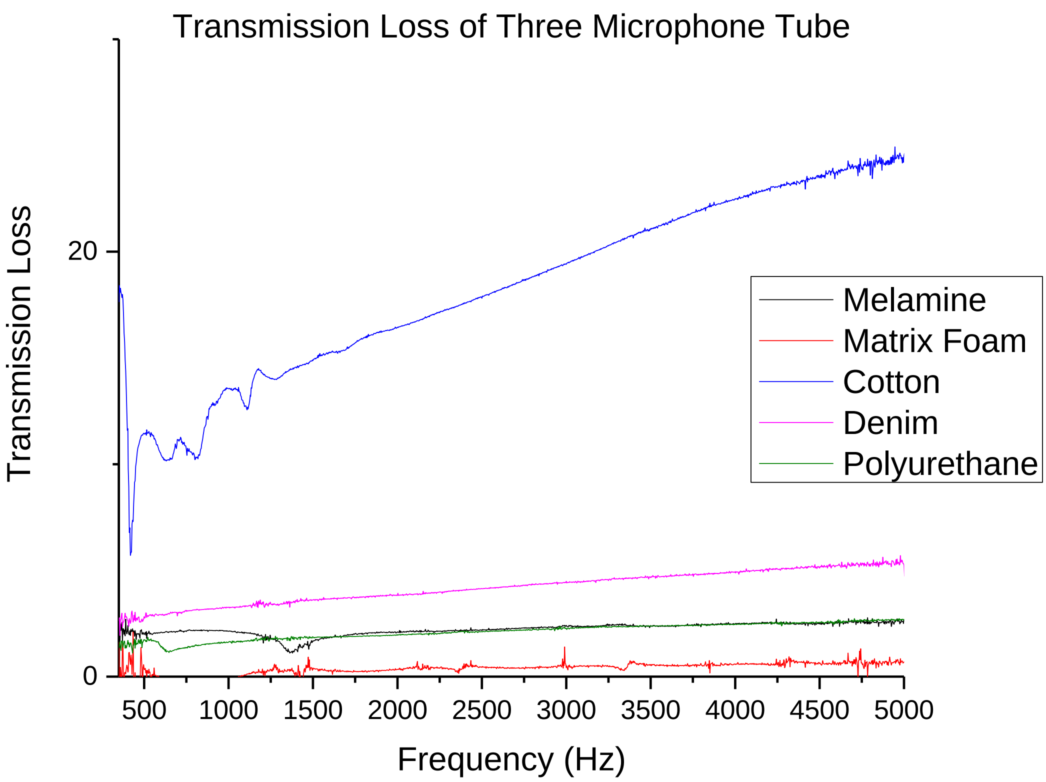


Figure 5.8 Transmission Loss of Samples in Three Microphone Tube

The transmission loss of the denim, melamine, and polyurethane each have reasonable results between each other. Out of the three samples the Denim would be the sample with the highest transmission loss. The melamine and polyurethane are around the same, especially at higher frequencies. It is important to remember that the melamine sample is thinner than the polyurethane sample. This means that the melamine sample has similar transmission loss values to the thicker polyurethane sample. Which in an aerospace application would mean it will take up less space on the aircraft but provide comparable transmission loss. This is why melamine is widely used commercially as in acoustic sound absorption and transmission loss applications. The cotton sample had the highest absorption coefficient of all the samples. Under 1500 Hz, there were some fluctuations in the transmission loss value, which could be attributed to the fact that the cotton sample is made up of three cotton balls. This means there are some gaps in between the balls, which could cause some anomalies in the results. The matrix foam sample was expected to show a very low transmission loss because of the large air gaps in the sample and its very low absorption coefficient properties. The hypothesis was true for majority of the frequency range of interest, except for cases under 1000 Hz. In this section, the transmission loss was calculated to be negative which can only be attributed to noise caused by the high reflectivity of the back plate and possibly the third microphone holder.

When looking at each tube individually, it is possible to compare the different materials to each other. This is beneficial to ensure that the tube is both working properly and providing the expected results. Since all three of the impedance tubes are able to measure absorption coefficients, ideally they should all measure the same absorption coefficient for the same sample. Since the denim sample provided a consistent and high absorption coefficient, it will be the material to study between the three tubes. This can be seen in Figure 5.9.

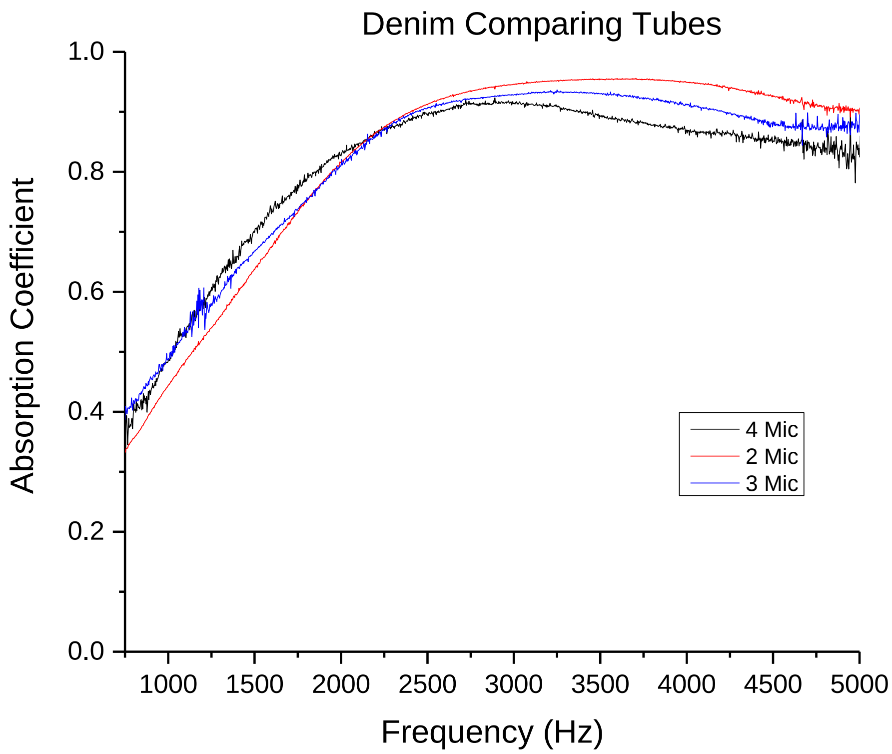


Figure 5.9 Absorption Coefficient of Denim Sample for All Three Tubes

Overall the three curves reasonably agree with each other. The two microphone tube does show the least amount of noise over the entire frequency range. This is probably due to the three and four microphone tubes connection not meeting up exactly centered. The two and three microphone tube curves are closer to each other than the two microphone tube which may also be caused by the center mismatch. Since the four microphone tube has a worse mismatch than the three microphone tube. It is the one that shows a larger deviation from the two microphone results. Over the entire frequency range it can be seen at most there is a 10% difference between the two and four microphone tubes. This is definitely something to consider when testing different materials because comparing two different materials each tested in a different tube, could potentially give you a 10% windows of error. So to stay consistent it is better to test materials in the same tube.

Since the four microphone and three microphone tubes are both able to measure the transmission loss of a material, their results can be compared to one another. The transmission loss of the denim, melamine, and polyurethane samples is shown in Figure 5.10. Not every material that was tested is shown because not every material was tested in each of the tubes.

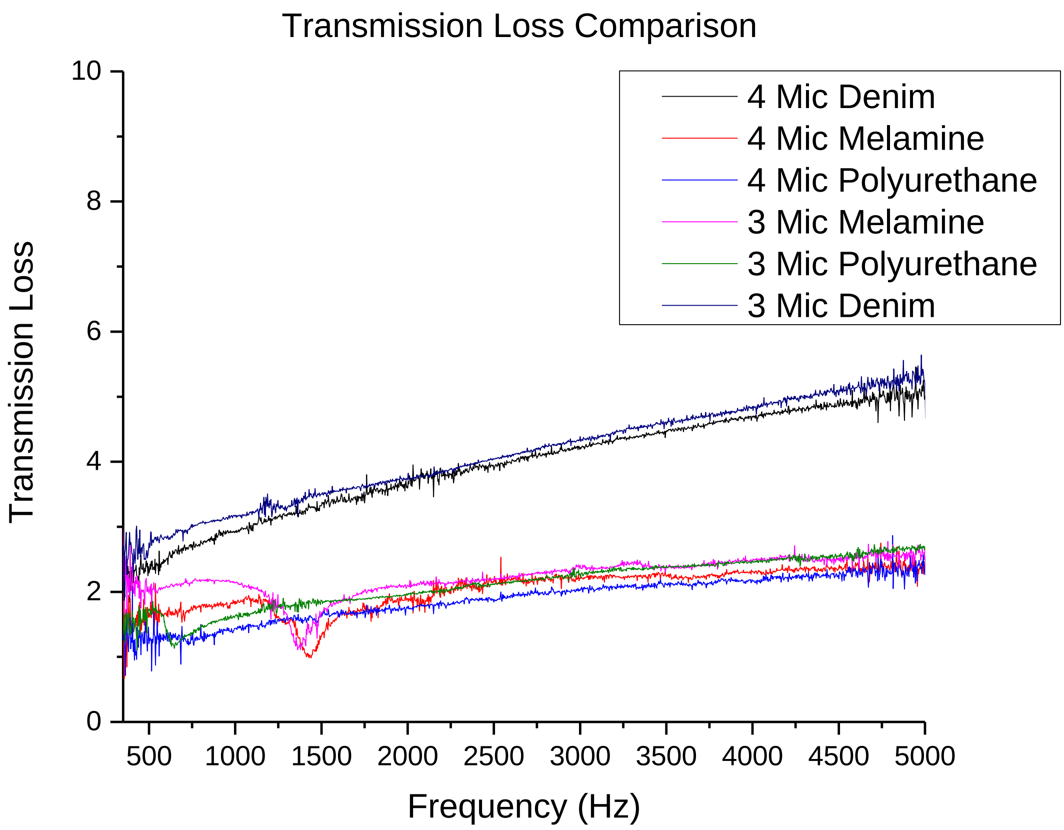


Figure 5.10 Transmission Loss of Samples in Three and Four Microphone Tubes

First taking a look at the denim sample, the transmission loss between the three and four microphone tube does not show much difference. The lower frequency range shows a difference of at most around 10%. This can be seen at the lower frequency range. When looking at the melamine sample, at lower frequencies there is a slight difference but both show a distinct dip in the curve around 1400 Hz. This is a typical trend that melamine shows and its thickness usually determines where this dip is. Similar to the other two samples tested, the polyurethane sample shows a consistent similarity between the transmission loss curve that was measured in the three and four microphone tubes. Again there is around a 10% difference in the value measured between the two tubes. Unlike the denim sample this difference is uniform over the frequency range after around 750 Hz. These results do show though, that the three and four microphone tube are measuring similar values to one another. Again though, I would recommend using the same tube when comparing test samples. This removes any error that could be caused by tube differences which can result in as much as 10%.

Chapter 6.
Conclusions and Future Work

6.1. Conclusions

In our present work the design and fabrication of a two, three, and four microphone impedance tube analyzer was performed. A design of experiments was conducted on the two microphone impedance tube to study how the design or test parameters that can affect the results to some degree during an experiment. As expected the microphone spacing was shown to be directly related to the frequency range that can be studied. It was found that the ISO standards recommendation for the lower limit value was near to the experimental result than the ASTM standards. An impedance tube fabricated from brass was shown to have a 1% improvement on attenuation loss though out the tube. The smoother interior wall provided less noise in test results. It was shown that the position of sample relative to the first microphone could cause an error that is as high as 2.5%. The rectangle white noise test signal showed the least amount of noise. An equalizer was showed minimal improvements on signal noise during testing and would not be mandatory in a configuration. The Selenium DH200E's higher build quality and lower frequency limit showed improvements in noise, especially at lower frequencies. An experiments time length was directly related to the inversely related to the noise in the results. There was no noticeable improvement longer than 10 seconds though. In the final experimental run it was shown that the optimal speaker volume needs to be determined to show the least amount of noise results.

The two microphone impedance tube provided adequate results for most of the materials tested. Highly reflective materials like the acrylic and the green foam samples provided erroneous results that. The results from these types of materials should studied more carefully to determine the correct way to measure their acoustic properties. The results may improve if the sample was moved further from the microphones. These reflective materials were also shown to provide poor results in the three and four microphone tube as well. Little sound reached the downward end of the tube which didn’t allow the microphones to measure anything but noise. It was shown that the three samples of the same material does not provide the exact same absorption coefficient. So at least three samples should be tested and then averaged.

The three different tubes were compared by measuring the absorption coefficient of different materials. It was shown that there can be a 10% difference between the three tubes when testing the same sample. This is a considerable amount if two different materials are going to be compared. The 10% difference was also seen when comparing the transmission loss compared between the three and four microphone tubes. To avoid introducing any error between the tubes, the same tube should be used when comparing different test specimens.

Only two different microphones were used when testing the three and four microphone tube. It would be easier and faster to run experiments if more microphones were used. This is especially the case if several microphone positions would be tested. It is suggested that the three microphone tube should not be performed with two microphones because you can introduce error in the microphone position when removing and inserting the third microphone into its holder. Since the results between the three and four microphone were comparable to each other, the three microphone tube would be preferred since it was faster and required less transfer functions.

6.2. Future Work

The future work will be to purchase at least one more microphone for the three microphone and two microphones if the four microphone is used. The four microphone will require a new downward end to be fabricated to fix the miss alignment in the center of the brass tube. The third microphone holder for the three microphone tube needs to be reprinted since it cracked during testing. Another design of experiments should be conducted on the three and four microphone tube to study what test parameters can affect results. A study into the affect that highly reflective samples has on results should be performed so to ensure how to obtain adequate results.

Piezoelectric samples will need to be fabricated so that their acoustic properties can be measured in the impedance tubes. A new mounting bracket may need to be created if the samples need to be electrically driven. A finite element model of the materials that were tested should be created to validate the results for the materials tested in this paper.

References

[1] A. Smith, "Aircraft and road traffic noise and children's cognition," *The Lancet,* vol. 366, pp. 715-716, 2005.

[2] *Airport Noise and Emissions Regulations: Newark Liberty International Airport,* F.A.A. Administration, 2011.

[3] L. Goines and L. Hagler, "Noise pollution: a modem plague," *South Med J,* vol. 100, pp. 287-94, Mar 2007.

[4] S. A. Stansfeld and M. P. Matheson, "Noise pollution: non-auditory effects on health," *British Medical Bulletin,* vol. 68, pp. 243-257, December 1, 2003 2003.

[5] W. Neise and L. Enghardt, "Technology approach to aero engine noise reduction," *Aerospace Science and Technology,* vol. 7, pp. 352-363, 7// 2003.

[6] C. Lu and P. Morrell, "Determination and Applications of Environmental Costs at Different Sized Airports – Aircraft Noise and Engine Emissions," *Transportation,* vol. 33, pp. 45-61, 2006/01/01 2006.

[7] F. A. Administration, "Noise Control and Compatability Planning for Airports," U. S. D. o. Transportation, Ed., ed, 1983.

[8] *Adoption of Statutory Prohibition on the Operation of Jets Weighing 75,000 Pounds or Less That Are Not Stage 3 Noise Compliant,* D. O. TRANSPORTATION 14 CFR Part 91, 2013.

[9] *NOISE STANDARDS: AIRCRAFT TYPE AND AIRWORTHINESS CERTIFICATION*, 2014.

[10] A. Kempton, "Acoustic liners for modern aero-engines," 2011.

[11] B. Dominik, M. Rudolf, and P. Reinhard, "The Virtual Scarf Inlet," in *13th AIAA/CEAS Aeroacoustics Conference (28th AIAA Aeroacoustics Conference)*, ed: American Institute of Aeronautics and Astronautics, 2007.

[12] A. Kempton, "Acoustic liners for modern aero-engines," Rolls-Royce, Ed., ed, 2011.

[13] J. Blauert, *Spatial hearing: the psychophysics of human sound localization*: MIT press, 1997.

[14] P. o. Oakland, "Aircraft Noise (Excerpt from the Oakland International Airport Master Plan Update)," P. o. Oakland, Ed., ed, 2006.

[15] S. W. Smith, *The scientist and engineer's guide to digital signal processing*: California Technical Publishing, 1997.

[16] V. D'Alessandro, G. Petrone, F. Franco, and S. De Rosa, "A review of the vibroacoustics of sandwich panels: Models and experiments," *Journal of Sandwich Structures and Materials,* vol. 15, pp. 541-582, 2013.

[17] P. R. Peters, Shankar; Nutt, Steven, "Sound transmission loss of damped honeycomb sandwich panels " presented at the INTER-NOISE, HONOLULU, HAWAII, USA, 2006.

[18] S. K. Ramanathan, "Sound transmission properties of honeycomb panels and double-walled structures," 2012.

[19] C. F. Ng and C. K. Hui, "Low frequency sound insulation using stiffness control with

honeycomb panels," *Applied Acoustics,* vol. 69, pp. 293-301, 4// 2008.

[20] H. Wen-chao and N. Chung-fai, "Sound insulation improvement using honeycomb sandwich panels," *Applied Acoustics,* vol. 53, pp. 163-177, 1// 1998.

[21] U. Biernat, "BASF foam Basotect in the interior insulation of new Boeing aircraft," in *BSAF The Chemical Company*, ed, 2007.

[22] G. Amaral-Labat, E. Gourdon, V. Fierro, A. Pizzi, and A. Celzard, "Acoustic properties of cellular vitreous carbon foams," *Carbon,* vol. 58, pp. 76-86, 7// 2013.

[23] H. Baumgartl, "Lightweight, Versatile All-rounder," in *Kunststoffe International*, ed, 2006, p. 3.

[24] Basotec, "Basotect® The versatile melamine resin foam," ed, 2014.

[25] Hilton, "B400 Impedance Tube Price," 2015.

[26] B. Kjaer, "B&K Impedance Tube Price," ed, 2015.

[27] A. S. C384, "Standard Test Method for Impedance and Absorption of Acoustical Materials by Impedance Tube Method," ed. West Conshocken, PA: ASTM International, 2004.

[28] A. S. E1050, "Standard Test Method for Impedance and Absorption of Acoustical Materials Using a Tube, Two Microphones and a Digital Frequency Analysis System," ed. West Conshocken, PA: ASTM International, 2010.

[29] A. S. E2611, "Standard Test Method for Measurement of Normal Incidence Sound Transmission of Acoustical Materials Based on the Transfer Matrix Method," ed. West Conshocken, PA: ASTM International, 2009.

[30] Iso, "ISO 10534-1, Acoustics - Determination of sound absorption coefficient and impedance in impedance tubes - Part 1: Method using standing wave ratio. International Standards Organisation," ed, 1996.

[31] Iso, "ISO 10534-2, Acoustics - Determination of sound absorption coefficient and impedance in impedance tubes - Part 2: Transfer-function method," ed, 1998.

[32] Y. Salissou, R. Panneton, and O. Doutres, "Complement to standard method for measuring normal incidence sound transmission loss with three microphones," *The Journal of the Acoustical Society of America,* vol. 131, pp. EL216-EL222, 2012.

[33] C. Kang, J. Matsumura, K. Oda, 順. 松村, 一. 小田, ジ. マツムラ*, et al.*, "A comparison of the Standing Wave and Two Microphone Methods in Measuring The Sound Absorption Coefficient of Wood," *Journal of the Faculty of Agriculture, Kyushu University,* vol. 51, pp. 1-4, 2006/02 2006.

[34] J. Y. Chung and D. A. Blaser, "Transfer function method of measuring in‐duct acoustic properties. I. Theory," *The Journal of the Acoustical Society of America,* vol. 68, pp. 907-913, 1980.

[35] I. L. V. Leo L. Beranek, *Noise and Vibration Control Engineering: Principles and Applications*, 1 ed.: Wiley-Interscience, 1992.

[36] B. H. Song and J. S. Bolton, "A transfer-matrix approach for estimating the characteristic impedance and wave numbers of limp and rigid porous materials," *The Journal of the Acoustical Society of America,* vol. 107, pp. 1131-1152, 2000.

[37] O. S. Doutres, Yacoubou; Atalla, Noureddine; Panneton, Raymond, "Evaluation of the acoustic and non-acoustic properties of sound absorbing materials using a three-microphone

impedance tube," *Applied Acoustics,* vol. 71, pp. 506-509, 2010.

[38] Y. Salissou, R. Panneton, and O. Doutres, *Three-microphone two-cavity method for measuring sound transmission loss in a modified impedance tube* vol. 39, 2011.

[39] T. Iwase, Yzumi, Y. and Kawabata, R. , "A new measuring method for sound propagation constant by using sound tube without any air space back of a test material," presented at the Internoise 98, New Zealand, 1998.

[40] Y. Salissou and R. Panneton, "Wideband characterization of the complex wave number and characteristic impedance of sound absorbers," *The Journal of the Acoustical Society of America,* vol. 128, pp. 2868-2876, 2010.

[41] O. P. Doutres, Raymond; Salissou, Yacoubou, "An additional configuration to standard ASTM E2611-09 for measuring the normal incidence sound transmission loss in a modified impedance tube," in *Acoustic 2012 Nantes Conference*, Nantes, France, 2012.

[42] Y. Salissou and R. Panneton, "A general wave decomposition formula for the measurement of normal incidence sound transmission loss in impedance tube," *The Journal of the Acoustical Society of America,* vol. 125, pp. 2083-2090, 2009.

[43] M. Suhanek, K. Jambrosic, and M. Horvat, "A comparison of two methods for measuring the sound absorption coefficient using impedance tubes," in *ELMAR, 2008. 50th International Symposium*, 2008, pp. 321-324.

[44] J. Y. Kunio, Taewook; Hou, Kang; Bolton, J. Stuart; Enok, Jan, "A Comparison of Two and Four Microphone Standing Wave Tube Procedures for Estimating the Normal Incidence Absorption Coefficient," presented at the Inter-Noise, Ottawa, Canada, 2009.

[45] M. Suhanek, K. Jambrosic, and H. Domitrovic, "Student project of building an impedance tube," *The Journal of the Acoustical Society of America,* vol. 123, pp. 3616-3616, 2008.

[46] S. P. O'Malley, "Construction and Testing of a Modern Acoustic Impedance Tube," M.S., NAVAL POSTGRADUATE SCHOOL MONTEREY CA, 2001.

Appendices

```
%%%%Two Microphone Impedance Tube
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
FileName='2mic_Foam_RE_H12_0926_3161_188C_3035inhg_200mv_11292014';
PlotName='Foam RE 200mV H12 L=0.926 B=5.101';
SpecimenL=.926;%%Specimen Length in inches
BackplateL=3.161;%%Backplate Length in inches
Temp=19.5; %%Temperature in C
s12 =0.851*.0254; %Distance between 1st and 2nd mic in meters
s14= 7.5*.0254; %Distance between 1st and 3rd mic in meters

%%%%This imports Test data from Matlab File.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
data=dlmread(FileName);
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

%%%%This imports Calibration Microphone data from Excel Files.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
Hc=dlmread('Calibrate_2cot_Brass_H12_0926_3161_249C_2971inhg_100mv_10172014');
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

%%%%This is used to calculate  Absorption Coefficient
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
c=331.4+.6*Temp;%speed of sound
L=(4.021+2.6-BackplateL-SpecimenL)*.0254;%Distance from specimen to first mic in meters
with all specimens

nfft=2^nextpow2(5000);
[H12,Freq]=tfestimate(data(:,1),data(:,2),[],[],nfft,25600,'onesided');%Transfer
Function

H12=H12./Hc;
k=2*pi*Freq/c;
R=((H12-exp(-1i*k*s12))./(exp(1i*k*s12)-H12)).*exp(1i*2*k*(L+s12));
a=1-(abs(R)).^2;%Absorption Coefficent
z=(1+R)./(1-R); %Normalized Impedance

%%%%Four Microphone Impedance Tube
```

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
FileName='4mic_Brass_0926_239C_2988inhg_100mv_10222014';
PlotName='Foam Brass Tube 4mic 100mV H12 L=0.926';
SpecimenL=0.926*.0254;%%Specimen Length in meters
Temp=23.9; %%Temperature in C
Pres=29.88*3.386; %Atmospheric Pressure in Kpa
s2=2.6*.0254;%Distance between front of specimen and 1st mic in meters
s1=0.851*.0254+s2;%Distance between front of specimen and 2nd mic in meters
s3=3.6*.0254;%Distance between front of specimen and 3rd mic in meters
s4=0.851*.0254+s3;%Distance between front of specimen and 4th mic in meters
s12 =0.851*.0254; %Distance between 1st and 2nd mic in meters
s13= 2.5*.0254; %Distance between 1st and 3rd mic in meters
s14= 5.0*.0254; %Distance between 1st and 4th mic in meters
s15= 7.5*.0254; %Distance between 1st and 5th mic in meters
s16= 13*.0254; %Distance between 1st and 6th mic in meters
s17= 18*.0254; %Distance between 1st and 7th mic in meters
s18= 33*.0254; %Distance between 1st and 8th mic in meters

%%%%This imports Test data from
%%%%Matlab File.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
H=dlmread(FileName);
Freq=H(:,1);
H11a=H(:,2);
H11b=H(:,3);
H21a=H(:,4);
H31a=H(:,5);
H41a=H(:,6);
H21b=H(:,7);
H31b=H(:,8);
H41b=H(:,9);

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

c=331.4+.6*Temp;%speed of sound
p=1.290*(Pres/101.325)*(273.15/(273.15+Temp));
k=2*pi*Freq/c-i*0.0194*sqrt(Freq)/c/SpecimenL;

Aa=i*(H11a.*exp(-i*k*s2)-H21a.*exp(-i*k*s1))./(2*sin(k*s12));
Ba=i*(H21a.*exp(i*k*s1)-H11a.*exp(i*k*s2))./(2*sin(k*s12));
```

```
Ca=i*(H31a.*exp(i*k*s4)-H41a.*exp(i*k*s3))./(2*sin(k*s12));
Da=i*(H41a.*exp(-i*k*s3)-H31a.*exp(-i*k*s4))./(2*sin(k*s12));
Ab=i*(H11b.*exp(-i*k*s2)-H21b.*exp(-i*k*s1))./(2*sin(k*s12));
Bb=i*(H21b.*exp(i*k*s1)-H11b.*exp(i*k*s2))./(2*sin(k*s12));
Cb=i*(H31b.*exp(i*k*s4)-H41b.*exp(i*k*s3))./(2*sin(k*s12));
Db=i*(H41b.*exp(-i*k*s3)-H31b.*exp(-i*k*s4))./(2*sin(k*s12));

P0a=Aa+Ba;
Pda=Ca.*exp(-i*k*SpecimenL)+Da.*exp(i*k*SpecimenL);
U0a=(Aa-Ba)/c/p;
Uda=(Ca.*exp(-i*k*SpecimenL)-Da.*exp(i*k*SpecimenL))/c/p;
P0b=Ab+Bb;
Pdb=Cb.*exp(-i*k*SpecimenL)+Db.*exp(i*k*SpecimenL);
U0b=(Ab-Bb)/c/p;
Udb=(Cb.*exp(-i*k*SpecimenL)-Db.*exp(i*k*SpecimenL))/c/p;

T11=(P0a.*Udb-P0b.*Uda)./(Pda.*Udb-Pdb.*Uda);
T12=(P0b.*Pda-P0a.*Pdb)./(Pda.*Udb-Pdb.*Uda);
T21=(U0a.*Udb-U0b.*Uda)./(Pda.*Udb-Pdb.*Uda);
T22=(Pda.*U0b-Pdb.*U0a)./(Pda.*Udb-Pdb.*Uda);

%{ Transfer Matrix for geometry symmetric
T11=(Pdb.*Udb+P0b.*U0b)./(P0b.*Udb-Pdb.*U0b);
T12=(P0b.*P0b-Pdb.*Pdb)./(P0b.*Udb-Pdb.*U0b);
T21=(U0b.*U0b-Udb.*Udb)./(P0b.*Udb-Pdb.*U0b);
T22=(Pdb.*Udb+Pdb.*U0b)./(P0b.*Udb-Pdb.*U0b);
%}

t=2*exp(i*k*SpecimenL)./(T11+T12/p/c+p*c*T21+T22);%Transmission Coefficient
R=(T11-p*c*T21)./(T11+p*c*T21);%Reflection Coefficent
a=1-(abs(R)).^2;%Absorption Coefficent
z=(1+R)./(1-R); %Normalized Impedance




%%%%Three Microphone Impedance Tube
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
FileName='3mic_Brass_2side_Demin_RE_H12_1150_3923 4750_214C_2970inhg_50mv_11062014';
PlotName='Foam Brass Tube 3mic 100mV H12 L=0.926';
SpecimenL=1.15*.0254;%%Specimen Length in meters
```

```
Temp=21.4; %%Temperature in C
Pres=29.70*3.386; %Atmospheric Pressure in Kpa
s2=(2.6*.0254-SpecimenL);%Distance between front of specimen and 1st mic in meters
s1=0.851*.0254+s2;%Distance between front of specimen and 2nd mic in meters
sa=(3.923)*.0254;%-SpecimenL;%Distance between front of specimen and 3a mic in meters
sb=(4.750)*.0254;%-SpecimenL;%Distance between front of specimen and 3b mic in meters
%s3=0.851*.0254+3.6*.0254;
%s4=5*.0254+3.6*.0254;
%s1=-5*.0254+s2
s12 =0.851*.0254; %Distance between 1st and 2nd mic in meters

%%%%This imports Test data from
%%%%Matlab File.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
data=dlmread(FileName);
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%This imports Calibration Microphone data from
%%%%Excel Files.
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
Hc=dlmread('Calibrate_2cot_Brass_H12_0926_3161_249C_2971inhg_100mv_10172014');
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%}

nfft=2^nextpow2(5000);

[H12a,Freq]=tfestimate(data(:,1),data(:,2),[],[],nfft,25600,'onesided');%Transfer
Function
[H13a,Freq]=tfestimate(data(:,3),data(:,4),[],[],nfft,25600,'onesided');%Transfer
Function
[H12b,Freq]=tfestimate(data(:,5),data(:,6),[],[],nfft,25600,'onesided');%Transfer
Function
[H13b,Freq]=tfestimate(data(:,7),data(:,8),[],[],nfft,25600,'onesided');%Transfer
Function
%%%%%%%%%%%%%%%%%
H12a=H12a./Hc;
H13a=H13a./Hc;
H12b=H12b./Hc;
H13b=H13b./Hc;
%%%%%%%%%%%%%%%%%%
```

```
c=331.4+.6*Temp;%speed of sound
p=1.290*(Pres/101.325)*(273.15/(273.15+Temp));
k=2*pi*Freq/c-i*0.0194*sqrt(Freq)/c/SpecimenL;
Z0=p*c;
Zs=Z0;

P0a=-2*i*exp(i*k*s2).*(H12a.*sin(k*(s1))-sin(k*s2))./(H12a.*exp(-i*k*s12)-1);
Pda=-2*i*exp(i*k*s2).*(H13a.*sin(k*(s12)).*cos(k*sa))./(H12a.*exp(-i*k*s12)-1);
U0a=(2*exp(i*k*s2)./Zs).*(H12a.*cos(k*(s1))-cos(k*s2))./(H12a.*exp(-i*k*s12)-1);
Uda=(2*exp(i*k*s2)./Zs).*(H13a.*sin(k*(s12)).*sin(k*sa))./(H12a.*exp(-i*k*s12)-1);
P0b=-2*i*exp(i*k*s2).*(H12b.*sin(k*(s1))-sin(k*s2))./(H12b.*exp(-i*k*s12)-1);
Pdb=-2*i*exp(i*k*s2).*(H13b.*sin(k*(s12)).*cos(k*sb))./(H12b.*exp(-i*k*s12)-1);
U0b=(2*exp(i*k*s2)./Zs).*(H12b.*cos(k*(s1))-cos(k*s2))./(H12b.*exp(-i*k*s12)-1);
Udb=(2*exp(i*k*s2)./Zs).*(H13b.*sin(k*(s12)).*sin(k*sb))./(H12b.*exp(-i*k*s12)-1);

T11=(P0a.*Udb-P0b.*Uda)./(Pda.*Udb-Pdb.*Uda);
T12=(P0b.*Pda-P0a.*Pdb)./(Pda.*Udb-Pdb.*Uda);
T21=(U0a.*Udb-U0b.*Uda)./(Pda.*Udb-Pdb.*Uda);
T22=(Pda.*U0b-Pdb.*U0a)./(Pda.*Udb-Pdb.*Uda);
%{ Transfer Matrix for geometry symmetric
T11=(Pda.*Uda+P0a.*U0a)./(P0a.*Uda-U0a.*Pda);
T12=(P0a.*P0a-Pda.*Pda)./(P0a.*Uda-U0a.*Pda);
T21=(U0a.*U0a-Uda.*Uda)./(P0a.*Uda-U0a.*Pda);
T22=(Pda.*Uda+Pda.*U0a)./(P0a.*Uda-U0a.*Pda);
%}
R=(T11-p*c*T21)./(T11+p*c*T21);%Reflection Coefficent
a=1-(abs(R)).^2;%Absorption Coefficent
t=2*exp(i*k*SpecimenL)./(T11+T12./(p*c)+(p*c).*T21+T22);%Transmission Coefficient
Tln=20.*log10(abs(1./t));
z=(1+R)./(1-R); %Normalized Impedance
```