UNIVERSITY OF MIAMI
COLLEGE of ENGINEERING

Department of Electrical and Computer Engineering

SENIOR DESIGN PROJECT

Automated Acoustic Impedance Tube

By

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ABSTRACT

The acquisition of acoustic properties such as absorption and transmission-loss coefficients is necessary for the analysis and synthesis of acoustic materials. Because of its accuracy, the impedance tube is the preferred method of measurement. However, current implementations require a strong technical background in acoustics and considerable time to produce results. The system described in this work will use a microcontroller to automate the measurement process and expedite the delivery of information to a non-expert end-user.

Keywords—impedance tube; acoustics; absorption; reflection; microcontroller; automation;

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1. INTRODUCTION

1.1 Problem Statement

One of the main challenges regarding modern impedance tubes is to facilitate ease of use, especially for those without engineering backgrounds. In settings such as construction and education, use of an impedance tube may be required by those without a strong background in engineering or acoustics. Two current configurations for impedance tubes are a single movable microphone which is adjusted depending on the frequency of the loudspeaker and a microphone array which uses interpolation to interpret acoustic coefficients. The first configuration requires calculations of wavelength to determine where to place the microphone for each frequency emitted by the loudspeaker exciting the tube. The adjustments must also be done manually to obtain a full range of coefficients for a given material. The second configuration requires interpolation to be performed on the data after recording and before the data is of any use. Both configurations require significant background knowledge in acoustics, signal processing, and mathematics.

The second issue with current implementations of impedance tubes is cost, which can total thousands of dollars. This limits the usage of a tube to operations with a large budget. A low cost realization of an impedance tube will allow the acoustic coefficients of a material be used in many more applications.

1.2 General Information

To address the first issue stated in 1.1, the first objective of this project is to realize an impedance tube that allows non-technical users of any educational background to produce accurate results in a timely manner. Through the use of a microcontroller, the proposed device can drive different frequencies, change microphone position, and post-process the data to write to external memory or to display on a computer. At the push of a button, the loudspeaker will generate each frequency and the microphone will gather data along its range of motion. Data will be recorded and post-processing will be performed, allowing coefficients to be obtained without additional calculations done by the user.

The cost issue will be addressed by using generic parts that are shaped and tuned for the tube, instead of manufacturing materials for the specific application. The tube can be shaped from PVC plumbing and the microphone and speaker can be of non-ideal frequency response that can be compensated by using signal processing in the microcontroller. The frequencies of interest are below 5kHz, so an electret microphone and paper cone driver will have comparable results to a high-end reference microphone and driver.

The automation of the impedance tube will be carried out in two phases: (i) semi-automated and (ii) fully automated. This project will document the development and implementation of semi-automation, which includes an Arduino-controlled test signal generator whose frequency must be manually changed with a switch, and manually moving the microphone and pushing record in a software interface. The biggest improvement over the current standards for this phase is that many recordings are taken at many different positions for each frequency, with the included MATLAB-developed application finding the minimum and maximum mic positions. This eliminates the need for calculation of where to place the microphone beforehand. Components of full automation were developed and tested, including Arduino code which cycles through frequencies and automatically moves the microphone to various frequencies, as well as moving all processing from software to the Arduino.

Some of the issues that this project addresses are outlined below:

Economical - Part of the goal of the project is lowering the cost of the tube, making low-cost construction of acoustically sound spaces more feasible.

Environmental - While PVC pipe (used to construct tube) is cheap and accessible, it is not an environmentally friendly substance.

Social - One use of the automated impedance tube is for educational purposes, creating the possibility of people being more aware of the acoustics of spaces in general.

Ethical - Aside from environmental concerns, the most important ethical concern is the consequences if the results of the tube are inaccurate. The worst case scenario would be designing of spaces with undesirable acoustics where music/speech cannot be properly perceived because of incorrect attenuation.

Manufacturability - The use of generic PVC and fittings make the project easily reproducible. The most complex and expensive component would be the semi-automatic version’s requirement of a computer.

1.3 Specifications & Goals

Goals:

- Automate overly technical and mathematical components of the impedance tube to make usable by those of any background.
- Provide tube at cost that is significantly lower than that of currently available products.

Specifications:

- Must return absorption coefficients with less than 10% error rate from accepted values.
- Tube must be airtight (little to no acoustic leakage).
- Microphone must be able to sweep across the tube waveguide
- Construction material must have low sound absorption properties
- Tube walls must be rigid enough to prevent sound propagation through vibration of the walls.
- Microphone and loudspeaker must be physically isolated from each other.
- Test material must be same length and shape as the tube aperture.

2. DESIGN PROCEDURE

2.1 Design Decisions

This design process was focused around cost efficiency, and automation to reduce complexity for the user. The two questions we wanted to answer were:

I. Can the tube be manufactured easily and at a lower cost than currently available models?

II. Will our version of the tube expedite the process of obtaining absorption/transmission coefficients?

The main ethical consideration was that the tube functions as expected so that if used to for acoustic design in a building, there will not be problems associated with bad acoustics due to use of our tube. This ties into health, where if acoustics are especially bad, hearing damage could result.

2.2 Applicable Engineering Standards

There are multiple standards for measuring the absorption coefficients of a sample building material. The following influenced the approach of this design:

Standard C384-04 : This ASTM standard for the use of an impedance tube, alternatively called a standing wave apparatus, for the measurement of impedance ratios and the normal incidence sound absorption coefficients of acoustical materials. The standard is an outline for the procedure and equations needed to design a standing wave apparatus and includes a diagram [6].

Standard E1050-12 : This ASTM standard provides a more in-depth example of an impedance tube using two microphones and a digital frequency analysis system. The standard includes diagrams for examples of proper speaker housing and microphone mounting [7].

Standard E2611-09 : This ASTM standard expounds upon the aforementioned ones in that it uses a transfer matrix method to also analyze the transmission loss coefficients of test materials. The method uses microphones on both sides of a test material and the example is well documented with figures and equations [8].

This design measurement method was largely based off of the Standing Wave Method, taken from Standard C384-04. This was based on trying to keep the cost of the unit as minimal as possible, and to gain more data sets to determine the maximum and minimum RMS pressure values with increasing accuracy.

2.3 Equations / Simulations / General Circuits/ Architecture/(High-Level) System Diagrams

I. Applied Concepts of Acoustics

Impedance tubes work on an understanding of basic acoustic principles of wave propagation. In order to best collect accurate measurements from an acoustic impedance tube, the waves traveling through the waveguide must be acting in plane-wave propagation.

$$P = p_0 e^{-jk_y y}(e^{-jk_x x} + re^{jk_x x}) \quad \text{Eq. 2.1}$$

Equation 2.1 defines the constraints for plane-wave propagation in a lossless tube across the x-axis, which can be illustrated as the propagation along the length of the tube. Making sure that the tube can accommodate plane-wave propagation for the highest frequency and lowest frequency is also crucial to capture accurate measurements of the compressions and rarefactions in air along the tube.

$$f_n = x_n \frac{c}{2\pi a} \quad \text{Eq. 2.2}$$

Note that x_n is the Bessel function value (selected to be 1.841). The constant *c* is the propagation of speed (~= 341 m/s), and *a* is the diameter of the waveguide. Determining the lower limit of the the frequency range to test the samples in the impedance tube will affect the proper length to encapsulate at least a full wavelength of the lower frequency, whereas the determining the upper limit of the frequency range will affect the how wide the tube must be in order to keep plane-wave propagation consistent across all frequencies.

In this case, for a given diameter of 3 inches or 7.62 centimeters, the upper bound limiting frequency for plane-wave propagation is 1311.21Hz.

$$L > \frac{\lambda}{2} \Rightarrow L > \frac{1}{2}\frac{c}{f_{min}} \quad \text{Eq. 2.3}$$

In designing an impedance tube, the tube should be able to have at least one maxima and one minima. Equation 2.3 represents a self-imposed condition to determine the minimum length of the tube by taking the lowest frequency band tested into account.

At the moment the incoming source wave hits the sample, a reflected wave comes back at an inverted 180 degree phase shift. The interaction of both the reflected wave and source wave results in a smaller residual signal that can best be illustrated by Figure 2.1 below. The only difference to take into consideration is that the wall at the closed end of the tube will hold the sample of the building material in question, and affect the strength of the reflected wave.

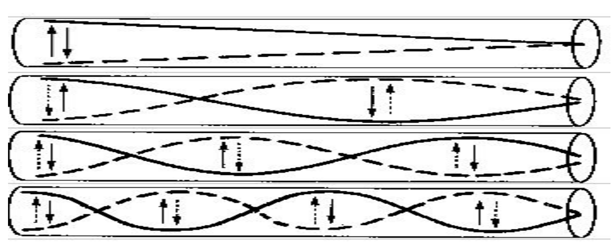


Figure 2.1 *Ideal wave reflections in a half open tube [13]*

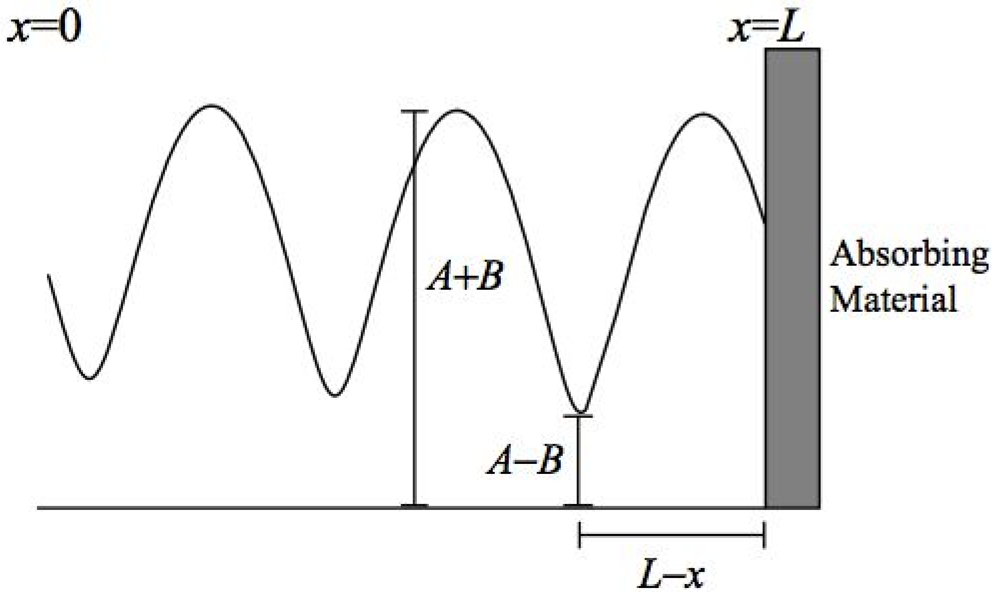


Figure 2.2 *The residual signal in closer view. The upward slope of the signal is defined by the energy lost to the tube walls [13].*

The residual signal shows that there are still maximum and minimum compressions of air pressure defined by the amplitudes *(A+B)* and *(A-B)* respectively. Let the two mentioned amplitude values be redefined as p_{min} and p_{max}. The absorbent coefficients per frequency band are defined as ratios of RMS pressure values in Equation 2.4:

$$\mu = \frac{p_{min}}{p_{max}} \qquad \text{Eq. 2.4}$$

By calculating Equation 2.4, the reflection coefficient R can be determined as well. This value is then used to determine the absorbent coefficient α or the power loss factor β [8]. Both are identical by mathematical definition, and define the extent to which a wall material can absorb energy from a given frequency band or across the noise spectrum.

$$R = \frac{1-\mu}{1+\mu} \qquad \text{Eq. 2.5}$$

$$\alpha = \beta = 1 - |R|^2 \qquad \text{Eq. 2.6}$$

$$x_{max} = x_{min} \pm \frac{\lambda}{4} \quad \text{Eq. 2.7}$$

$$\phi^{o} = 180(\frac{x_{min}}{4} \pm 1) \quad \text{Eq. 2.8}$$

Even if it has already been determined through the Standing Wave Method, the position of maximum RMS pressure can be verified by taking the minimum RMS pressure value and shifting it a quarter of the wavelength in the RMS representation of the residual signal. With the same input information, the phase can be calculated as well at different points across the frequency spectrum. This can be better illustrated by Figure 2.3, which shows the RMS pressure signal representation of the residual wave.

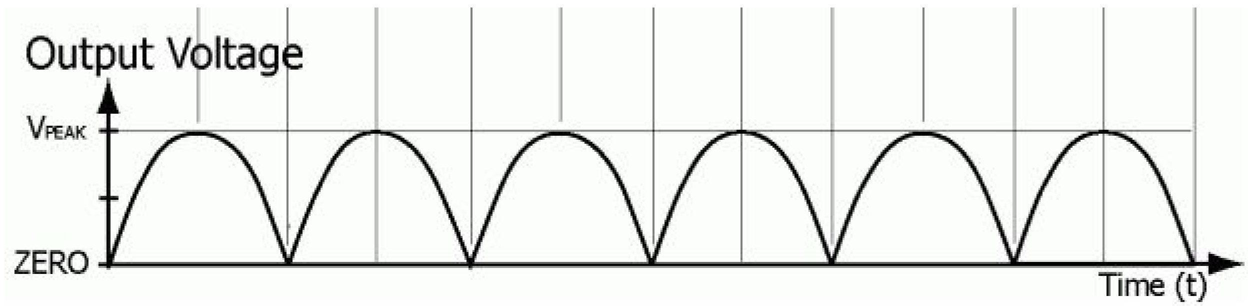


Fig 2.3 *RMS Voltage Representation of a recorded residual pressure signal with full reflection in time. The position of the maximum RMS voltage V_{peak} is a quarter-wavelength away from any neighboring minimum RMS voltage value [13].*

II. Applied Circuit Design

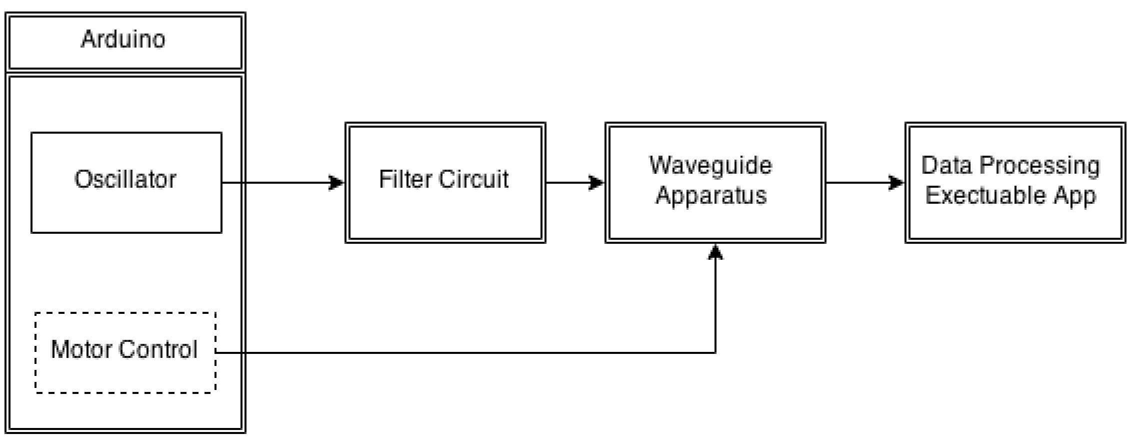


Figure 2.4 *Level 1 architecture of the automated impedance tube.*

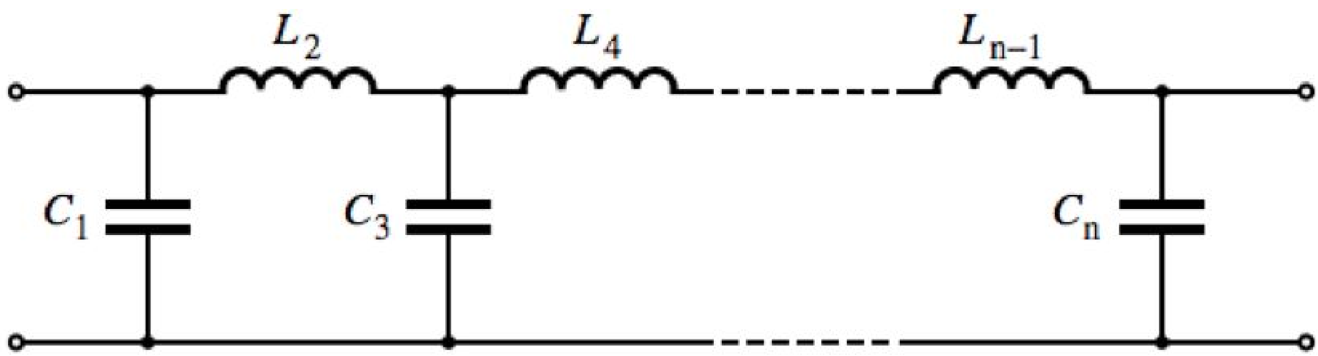


Fig 2.5 *N^{th} Order Chebyshev Low Pass Filter using PI section Cauer Topology [18]*

For an Nth order Type I Chebyshev Filter, the generalized transfer function can be defined as an all-pole system as displayed in Equation 2.1 below:

$$H(s) = \frac{1}{2^{N-1}\varepsilon s} \prod_{m=1}^{N} \frac{1}{(s - s_m^{-})} \qquad \text{Eq. 2.9}$$

In analog electronics, Chebyshev low pass filters are best designed by arranging a defined amount of reactive components (inductors and capacitors), in which the total amount of said reactive components determines the order *N* of the transfer function and the number of resonances that are determined by the order N of the function.

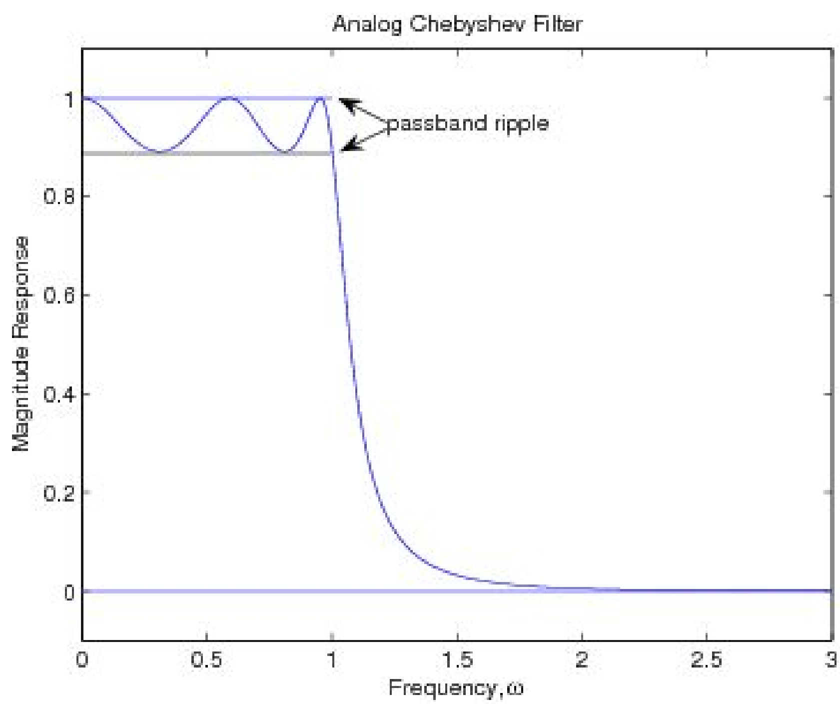


Figure 2.6 *Magnitude Response of a 5th order Type I Chebyshev filter [17]*

Chebyshev filters usually have a drawback in which their passband region usually has a ripple in magnitude that causes ringing in the said region. However they are also usually developed to have a steep rolloff regions between the passband and stopband regions in comparison to others such as Butterworth or Thomson filters in digital or analog implementation.

3. DESIGN DETAILS

3.1 Components

Arduino - Motor – Mic Positioning
Arduino – Oscillator
Arduino - Recording/Processing
Speaker Circuit
Waveguide Tube
MATLAB Generated .exe file - Processing

3.2 Implementation Diagrams (So someone can reproduce your work)

I. Loudspeaker

In order to obtain accurate absorption coefficients, many specifications had to be satisfied when generating a test signal. The generated wave has to be perfectly sinusoidal, so the coefficient generated for each frequency is purely generated from that frequency, and not its overtones. Generating a sine wave without external software and only using microcontrollers/analog circuits is a multi-step process, where care must be taken at each step to avoid distortion. Distortion at any step will cause additional harmonics which cause inaccurate results.

A. Pulse Wave Modulation and Filtering

The Arduino Mini used in this implementation has a bit depth of 8 bits, which only allows a maximum of 256 values to represent a sine wave. This rounding due to low bit depth is known as quantization error. While the problem could be avoided by using a higher bit-depth processor, one objective of the project was minimizing cost, and using a lower bit-depth processor and compensating for the error using a low cost passive filter is cheaper than using a 12 or 16 bit processor.

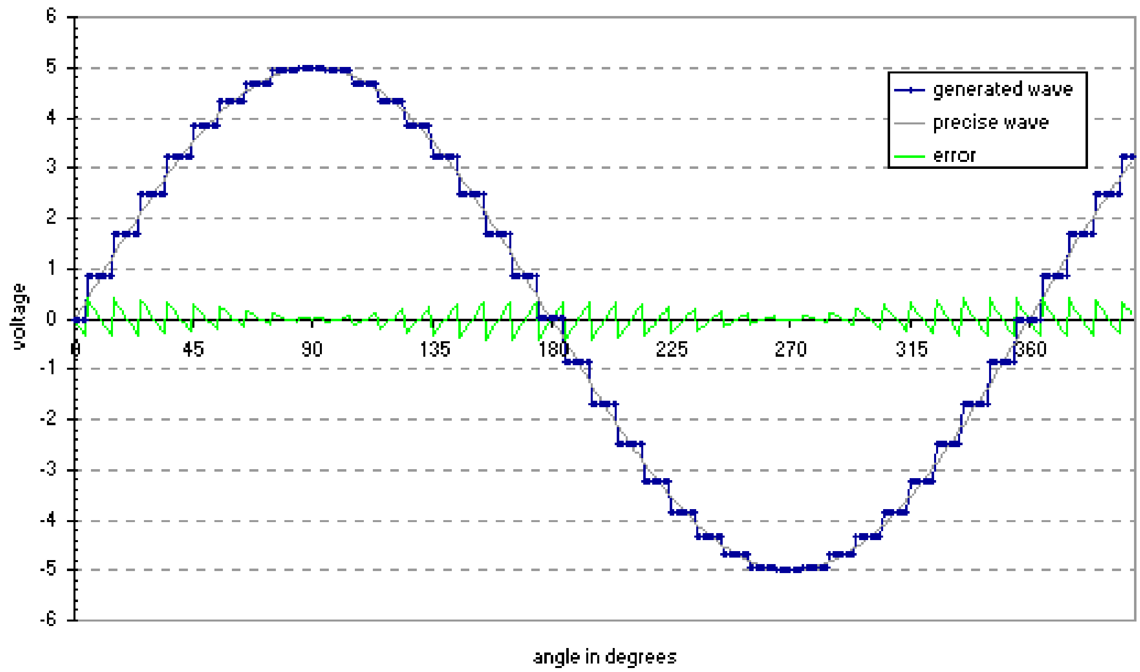


Figure 3.1 *Example of quantization error and its effects on a signal [14].*

By using a highly quantized sine wave source, more high frequency overtones are added to the signal.

$$f(x) = \sum_{n=-\infty}^{\infty} A_n(e^{jnx}) \qquad \text{Eq. 3.1}$$

The Fourier Series (Eq. 3.1) states that every signal can be represented as a sum of sine waves. As harmonics are added, the wave becomes more “square”. Therefore, by properly tuning an analog low pass filter, the higher frequency overtones introduced by quantization error can be eliminated.

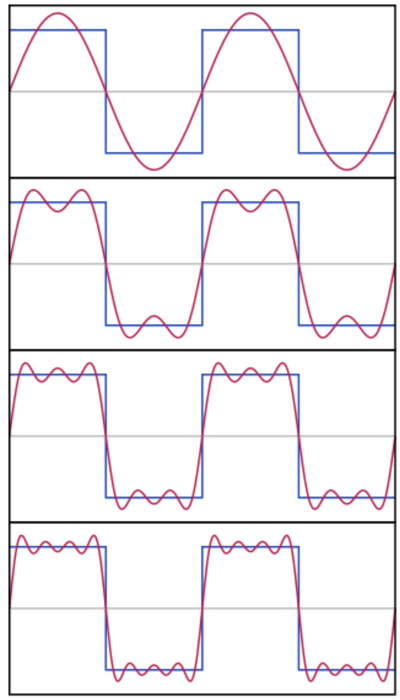


Figure 3.2 *Resulting waveform shape as additional harmonics are added to a single sine wave [15].*

Additionally, the Arduino does not output a true analog signal, but instead represents the 256 analog values using pulse wave modulation (PWM). This is a method of encoding signals using square waves with varying pulse widths that are defined by binary values, but varying lengths of the signal being high or low to represent the amplitude of the signal.

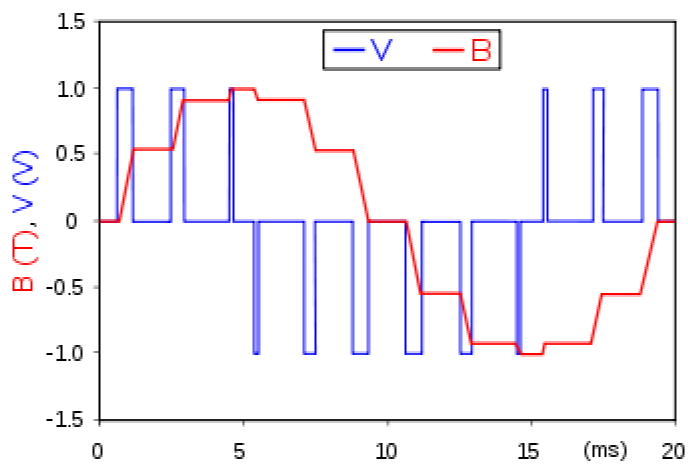


Figure 3.3 *Example of Pulse Wave Modulation (blue) encoding being performed on an analog signal (red) [16].*

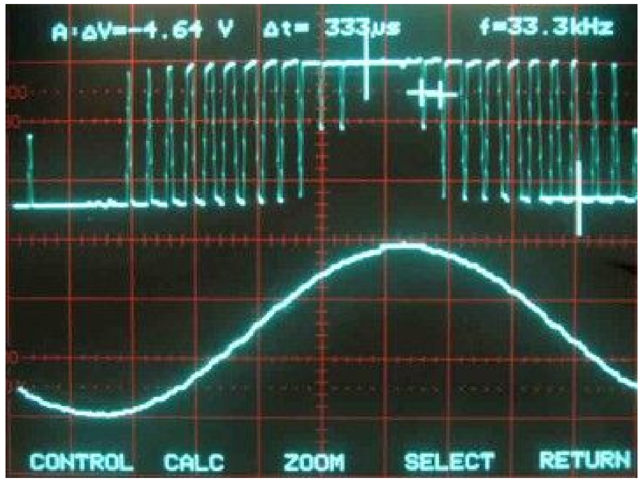


Figure 3.4 *Pulse wave modulated output from Arduino (top) and filtered output (bottom) [12].*

However, even with pulse wave modulation, low pass filtering should still properly remove the high frequency components and result in only the desired fundamental frequency. For this particular project, a 5th order Chebyshev low pass filter was used to remove high frequency components and output the pure sine wave with a cutoff frequency of 12.5kHz. The filter order must be relatively high to remove all necessary overtones while not attenuating the fundamental frequency.

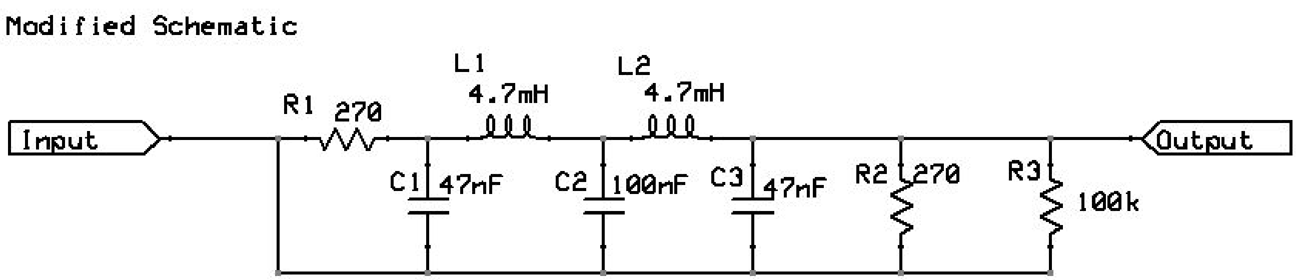


Figure 3.5 *Schematic for the 5th order Chebyshev low pass filter [12].*

B. Power Amplification

In order to properly amplify the output of the Arduino to drive the loudspeaker, a power amplifier needed to be incorporated into the design. Because of the low impedance of loudspeakers, they draw a significant amount of current. While the Arduino's power supply can generate 5 volts, it is not capable of generating enough current to power the loudspeaker. In order to properly power the loudspeaker, a 9V source consisting of 6 AA batteries in series was used, along with a JRC MJM386 power amplifier circuit. The AA's in series will be able to generate slightly more current than a single 9V battery, and the JRC chip is designed specifically for power amplification purposes. Other configurations such as a single 9V and an NE5532 generated distortion of the sine wave without any added gain. This was likely caused by sagging of the voltage source due to too much current being drawn, causing the amplifier to distort.

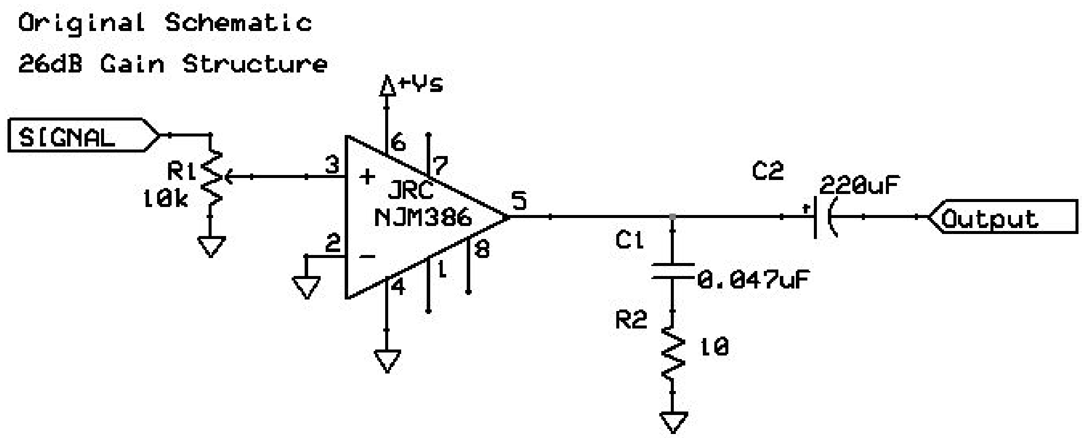


Figure 3.6 *Schematic for the power amplification stage [Appendix A].*

C. Voltage Division

To ensure the input of the power amplifier is sufficiently small to avoid clipping, a voltage divider is put in series before the filter and power amplifier. This stage must be done in the analog and not in the Arduino to avoid increased quantization error. By decreasing the amplitude of the signal in the Arduino, fewer of the possible 256 values are used, effectively increasing quantization error. However, the issue with this voltage divider is that it changes the cutoff frequency of the low pass filter. To solve this issue, a buffer is placed between the voltage divider and low pass filter stages. This buffer must be single supply biased , so the resulting circuit is much more complex than a dual supplied buffer. The buffer must be single-supply biased for a simple connection to *one* single battery pack as opposed to a dual-supply which requires two sources to provide the same swing voltage.

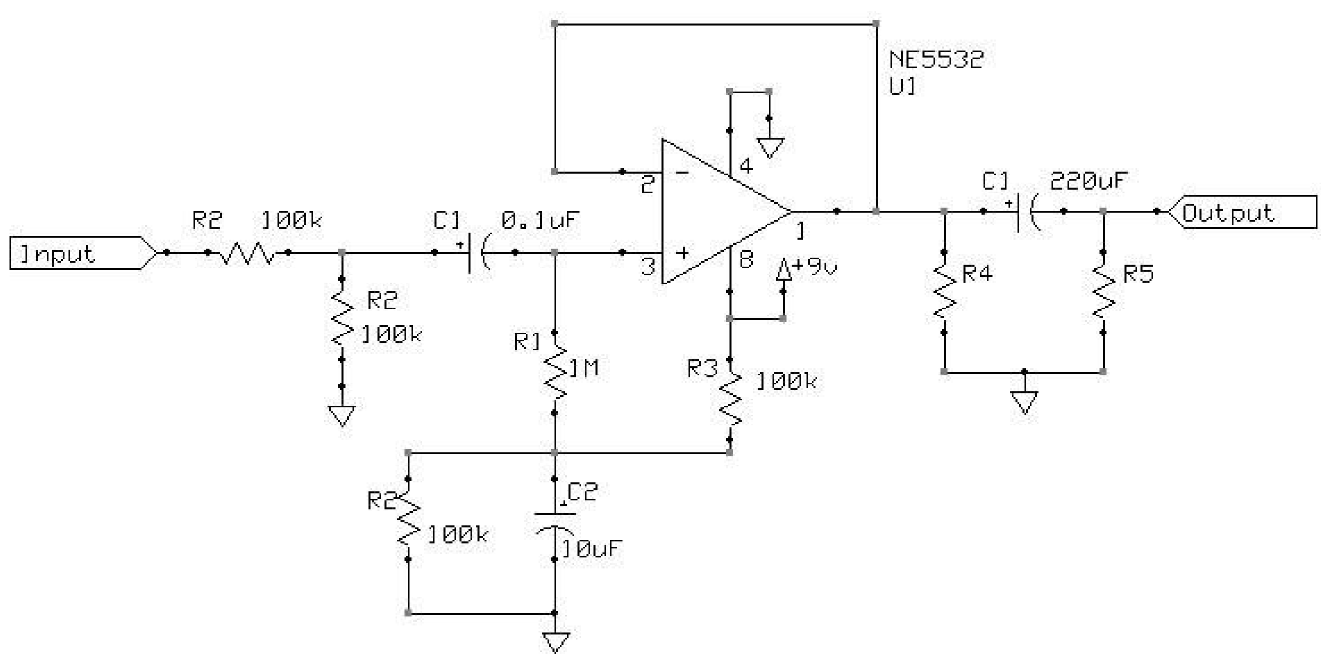


Figure 3.7 *Schematic for voltage division and single supplied unity gain buffer [11].*

D. Final Implementation

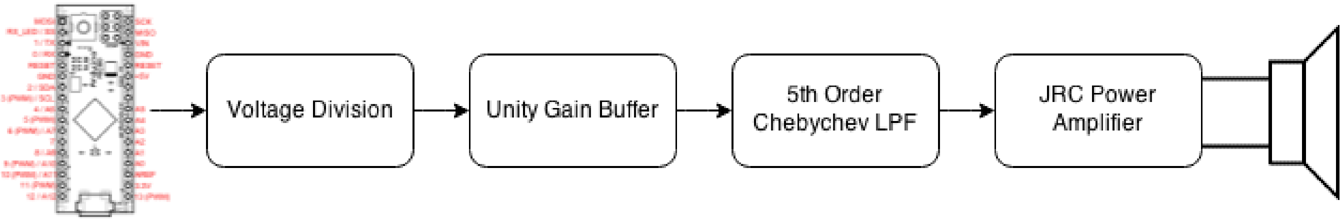


Figure 3.8 *Block diagram for loudspeaker component.*

II. Microphone

In order to obtain proper coefficients, the microphone must be both small enough to be placed in the tube without interrupting the plane wave propagation, and also must have a relatively flat frequency response. The microphone chosen for this implementation was a SparkFun MEMS microphone. The MEMS has a footprint of 4.72 mm × 3.76 mm × 1.0 mm, so it easily fits in the tube without interrupting plane wave propagation. The microphone has a built in amplifier with a gain of 67, allowing a signal that is usable for most applications. The bandwidth is also from 150 Hz to 15kHz which includes the entire range that an impedance tube would be used to measure.

III. Movement of Motor

While most implementations of an impedance tube use one or two fixed microphones and extensive use of interpolation and estimation to find the minimum and maximum points, this implementation records the test signal at multiple positions across the loudspeaker, using whichever recorded position is the minimum or maximum. The accuracy of the calculated pressure minimum and maximum increases as the number of positions increases, and this method avoids the estimation required for most impedance tube implementations. There are two possible implementations: a semi-automated implementation and a fully automated implementation.

A. Semi-Automated Implementation

The semi-automated implementation was used for prototyping and testing of the impedance tube. For the semi-automated implementation, the frequency of the loudspeaker is controlled by a 3-bit switch, hooked up to the inputs of the Arduino. The mic inputs are read into a program developed in MATLAB, and recordings for each mic position and frequency are triggered manually. The microphone position is moved by hand. The microphone is inside of the tube, with two wires extending through holes outside of the tube attached. By moving the position of the wires by a fixed distance, the microphone position is changed within the tube. Overall, the process for the semi-automated implementation is as follows:

(1) Secure sample in tube
(2) Set frequency of loudspeaker using switch
(3) Begin recording using “Record” button in software interface
(4) Move microphone position by fixed amount
(5) Repeat steps 3 and 4 for number of mic positions chosen for frequency.

(6) Repeat steps 2 through 5 for number of frequencies being calculated.
(7) Process data using “Process” button in software interface.

Although this is an involved, multi-step process, even the semi-automated process avoids extensive engineering calculations and knowledge. The semi-automated implementation is ideal for educational purposes by avoiding extensive setup and calculations, but still providing a degree of user interaction. The semi-automated implementation also served as a stepping point to the proposed fully automated implementation, and was beneficial for testing the accuracy of the tube.

B. Fully Automated Implementation

The goal of the proposed fully automated implementation is to simplify the entire process of acquiring absorption coefficients to two steps:

(1) Place sample material in tube
(2) Press “record” button to run test and calculate coefficients.

This implementation is much more costly and requires much more hardware interfacing, but is ideal for real world testing by users of any educational level. Instead of moving the microphone by hand, the microphone position is controlled by a motor attached to a rack and pinion assembly, similar to the apparatus used to control printer cartridge position. The entire tube is controlled by a second Arduino, which has control over the loudspeaker frequency and microphone position. The microphone will move N positions for every M frequencies being used, resulting in a total of M x N recordings. In order to take the tube from the classroom to the actual work field, the software user interface is converted to processing code written to the Arduino which will write coefficients to an SD card, eliminating the need of a computer.

Although the extent of this project was to obtain accurate test results from the semi-automated process, some steps were taken towards realizing a fully automated process. This included a centralized Arduino code which controls frequency, position, and processing of data, as well as testing of the horizontal drive motor attached to the tube to move the microphone. The main challenges to overcome involving the fully automated process is implementing it on a processor which can accurately interpret input audio signals (the Arduino reads analog signals as pulses, just as it writes), and setting up a system of pulleys and wires so that the drive motor can accurately control the position of the microphone within the tube.

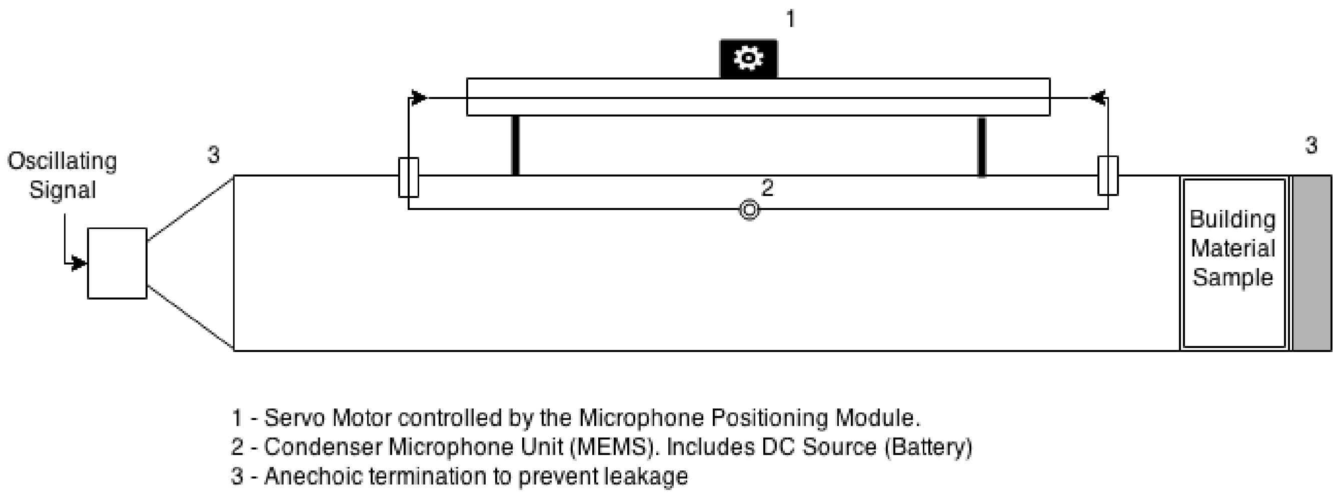


Figure 3.9 *Impedance Tube/Waveguide Apparatus*

IV. MATLAB User Interface

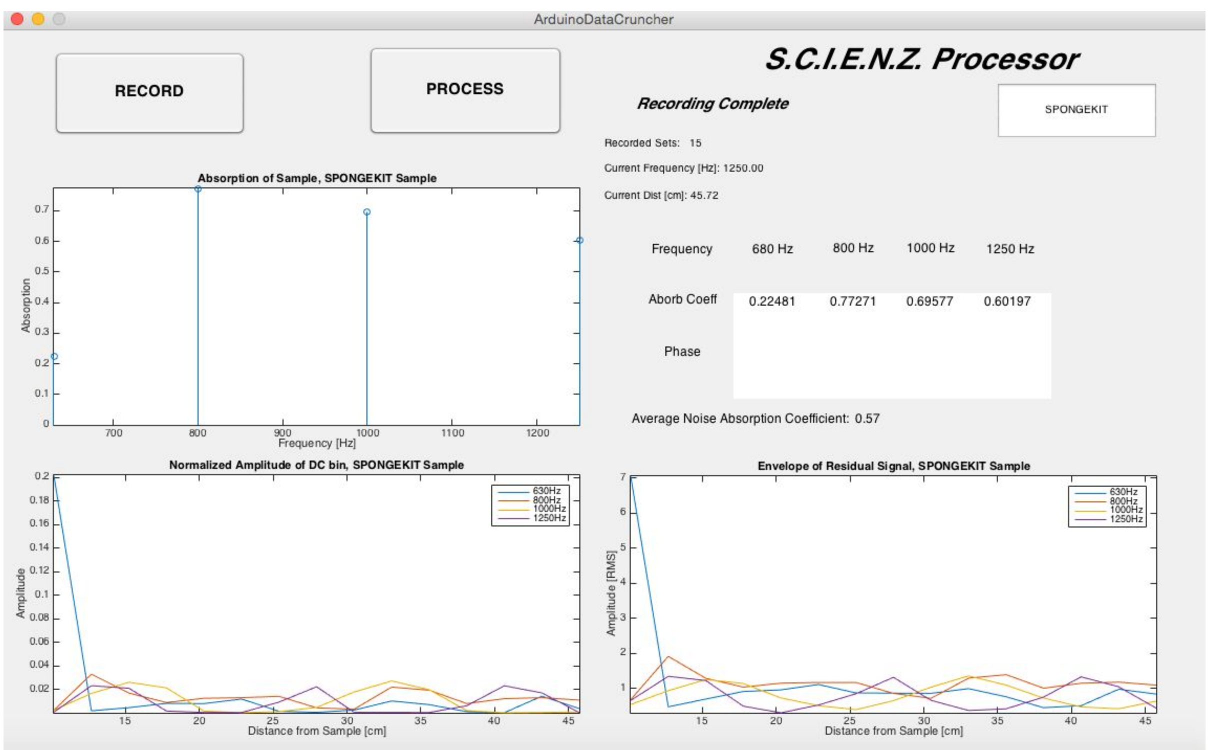


Figure 3.10 *S.C.I.E.N.Z Processor Application*

A. Basic Implementation

For the semi-automated implementation, MATLAB was used to develop a standalone program which records data and then processes the recordings into coefficients. For N microphone positions and M frequencies, an MxN cell array of audio recordings is created for an entire test sequence. The program will wait for the user to click “record”, allowing time between recordings to manually move the microphone and change the frequency.

Once all the recordings have been taken, the user must press the "Process" button, which will automatically convert the MxN cell matrix of audio vectors into an Mx1 set of absorption coefficients. The first step of this process is converting the MxN cell matrix of audio vectors into an MxN matrix or RMS coefficients. The RMS power level is used to select which microphone positions have the highest and lowest power levels. These values will be used in the rest of the calculations covered in section 2.3.I.

B. Mic Response Compensation

Due to the low-cost, space-efficient nature of the MEMS microphone, there is a significant low-frequency rolloff associated with the MEMS microphone. For properly obtaining the maxima and minima, successful recording of the low frequencies of the residual wave are extremely important.

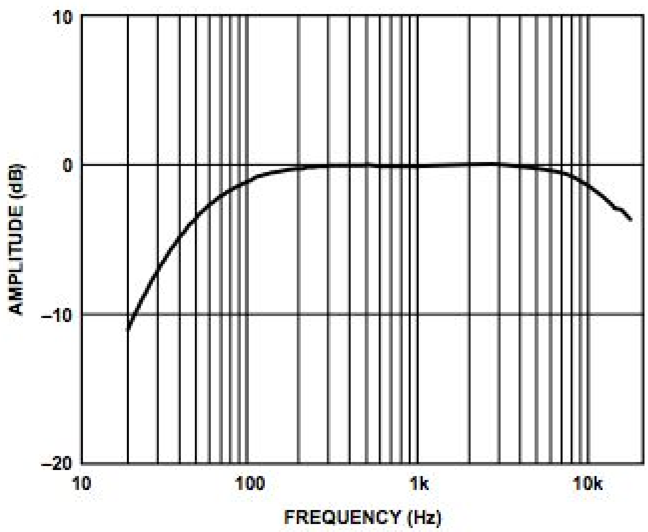


Figure 3.11 *Frequency response of INMP 401 (used in MEMS) [Appendix B].*

As can be seen from the datasheet, there is a significant low frequency rolloff associated with the MEMS microphone. With the low frequency response of the microphone too low to properly detect the pressure maxima and minima, a DC boosting factor (ξ), is multiplied with the lowest frequency bin in the frequency domain.

$$
\begin{aligned}
X(k) &= F\{x[n]\} \\
Y(k) &= X(k) \\
Y(1) &= \xi X(1) \\
y[n] &= F^{-1}\{Y(k)\}
\end{aligned}
\qquad \text{Eq. 3.2}
$$

This emphasizes the DC pressure offset created by the residual signal, making difference between pmin and pmax more pronounced. To compensate for the response of the MEMS, an ξ of 100 is used. Without ξ , the calculated coefficients are consistently too high, but with the adjustment factor, they fall into a much more acceptable range. While this issue would ideally be solved with a higher end microphone, this fix allows accurate results with a lower budget, as solving the issue in software has little to no additional cost.

C. Plots

At the end of each iteration of the process function, a series of plots and coefficients are generated. The three generated plots are (1) a stem plot giving the values of the absorption coefficients, (2) a plot of the M residual signals as a function of distance, calculated from the N RMS values for each microphone position, and (3) a plot of M DC bins as a function of distance, calculated from the N RMS values for each microphone position. Additionally, the M coefficients and the average coefficient are printed to the console in MATLAB.

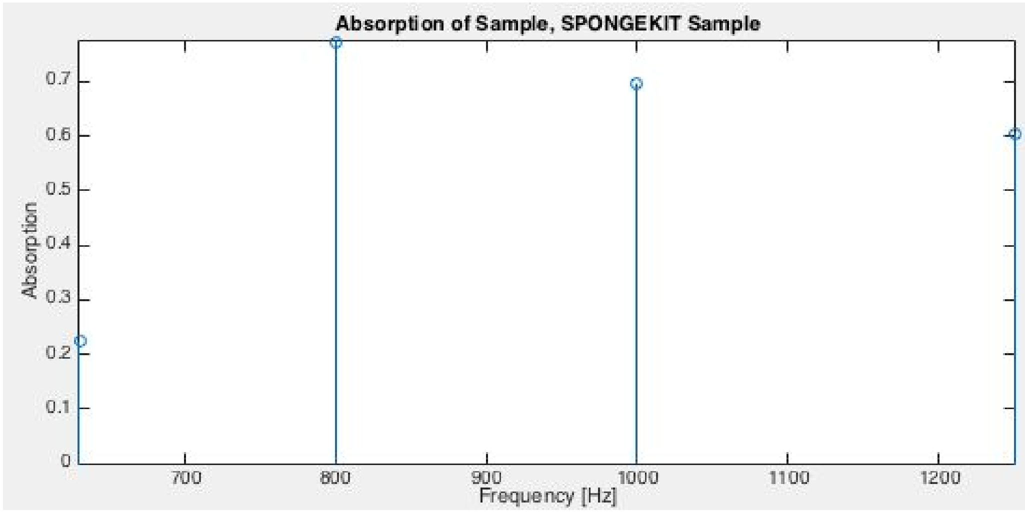
4. DESIGN VERIFICATION

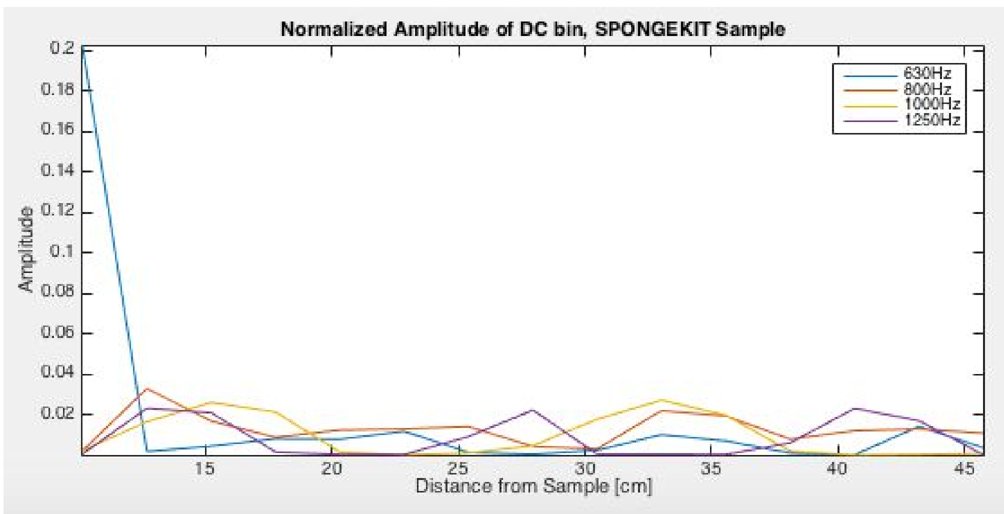
4.1 Testing Procedure

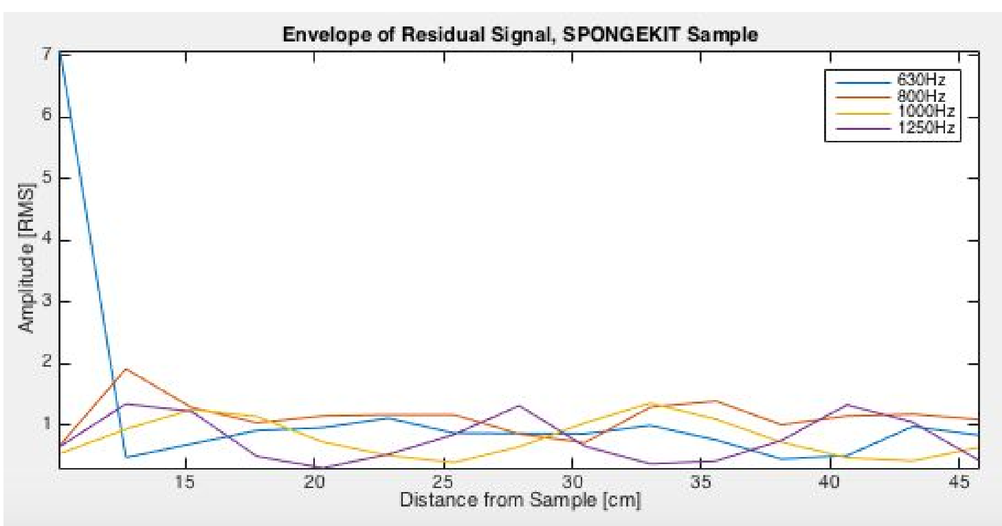
Based on equations 2.1-2.3, the usable range of the impedance tube is from around 680 Hz to 1311 Hz. Using third octave bands, this gives four usable frequencies: 680, 800, 1000, 1250. For each of these four frequencies, 15 positions are used, varying from 4 to 18 inches away from the samples. The test procedure consists of loading the sample, having one person move the microphone and change frequencies, and a second person pressing record for each microphone position. The wire controlling the movement of the microphone in the tube was marked at every inch so that every microphone position moves the microphone the same amount. This also allows the plots for RMS residual signal and DC bins plotted against distance in MATLAB to be accurate. The samples tested were bathroom tile, carpet, cork on wood, kitchen sponge, glass.

4.2 Plots and Tables

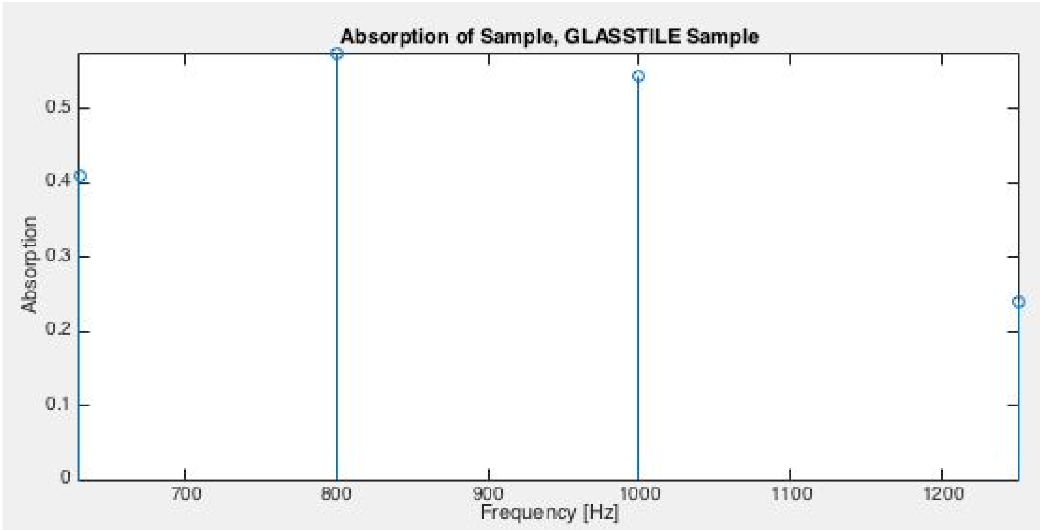
Sponge



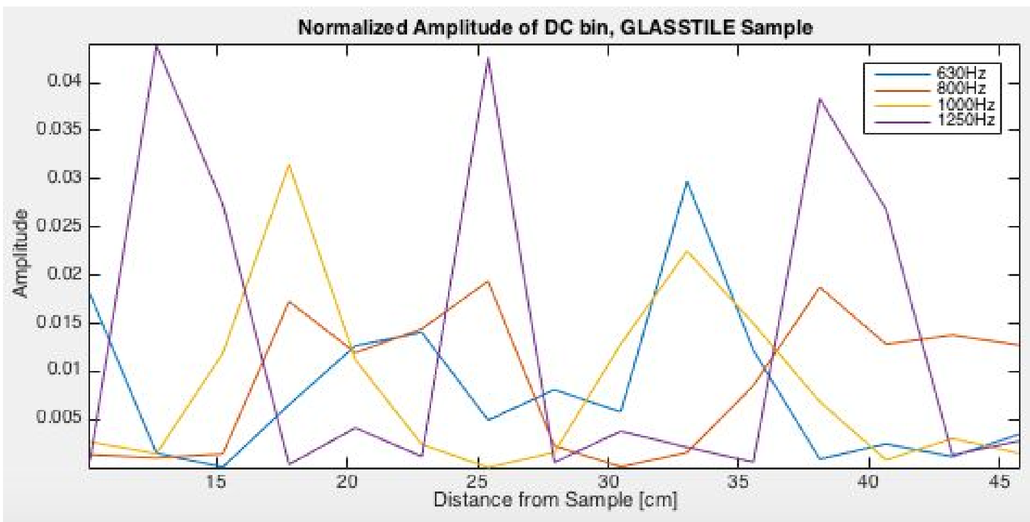




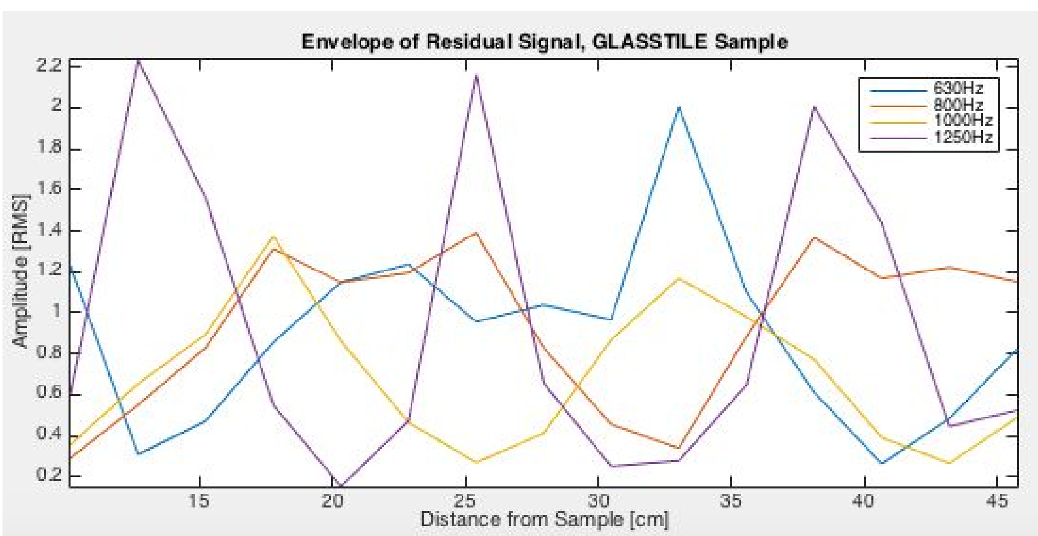
Glass



Absorption of Sample, GLASSTILE Sample
Absorption
0
0.1
0.2
0.3
0.4
0.5
700
800
900
1000
1100
1200
Frequency [Hz]

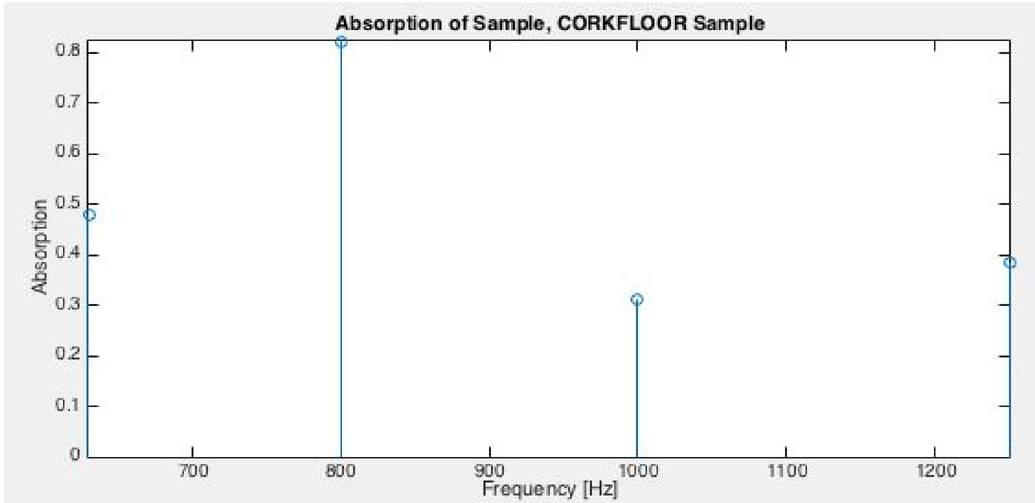


Normalized Amplitude of DC bin, GLASSTILE Sample
Amplitude
0.005
0.01
0.015
0.02
0.025
0.03
0.035
0.04
15
20
25
30
35
40
45
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz

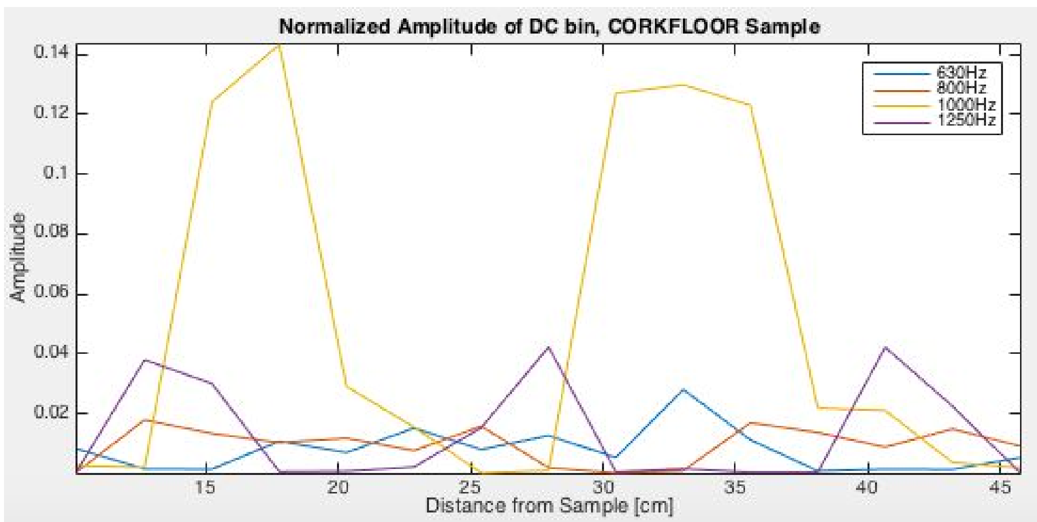


Envelope of Residual Signal, GLASSTILE Sample
Amplitude [RMS]
0.2
0.4
0.6
0.8
1
1.2
1.4
1.6
1.8
2
2.2
15
20
25
30
35
40
45
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz

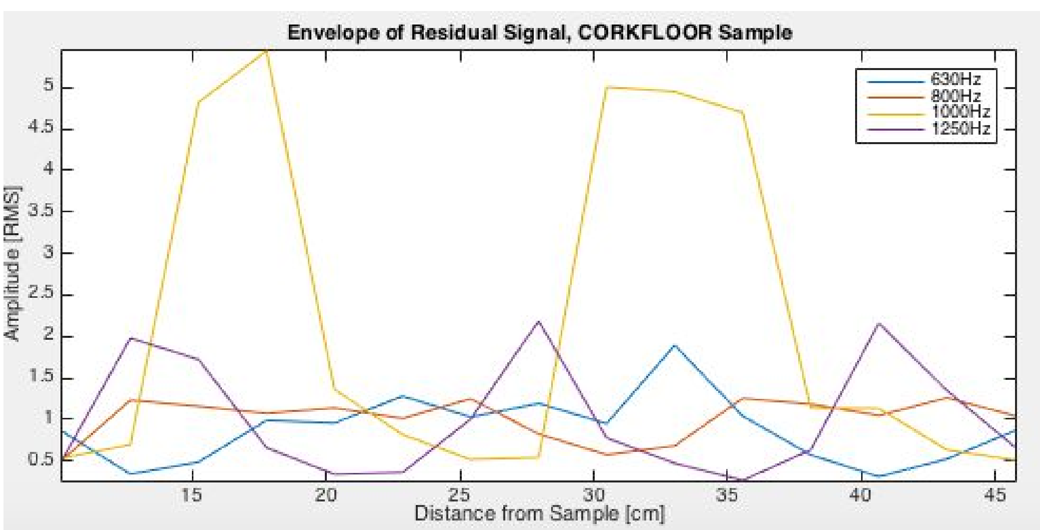
Cork



Absorption of Sample, CORKFLOOR Sample
Absorption
Frequency [Hz]

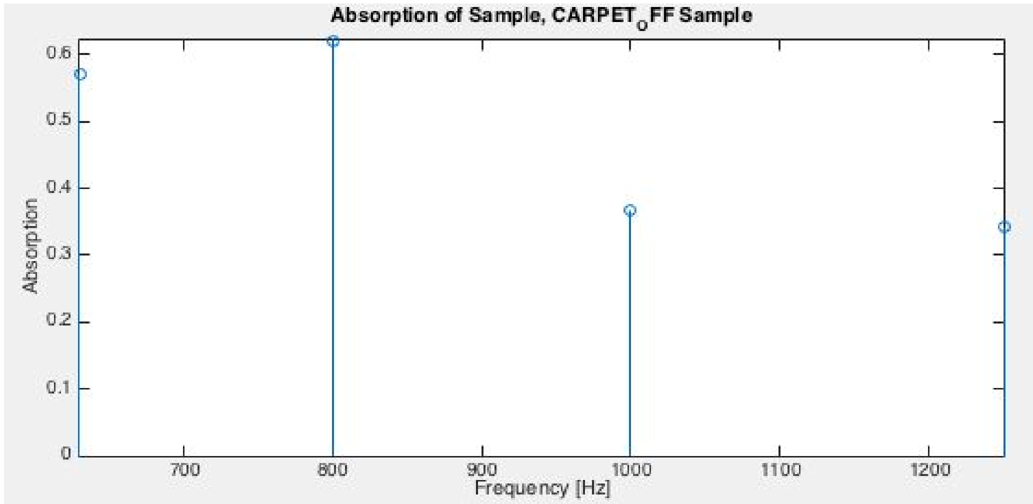


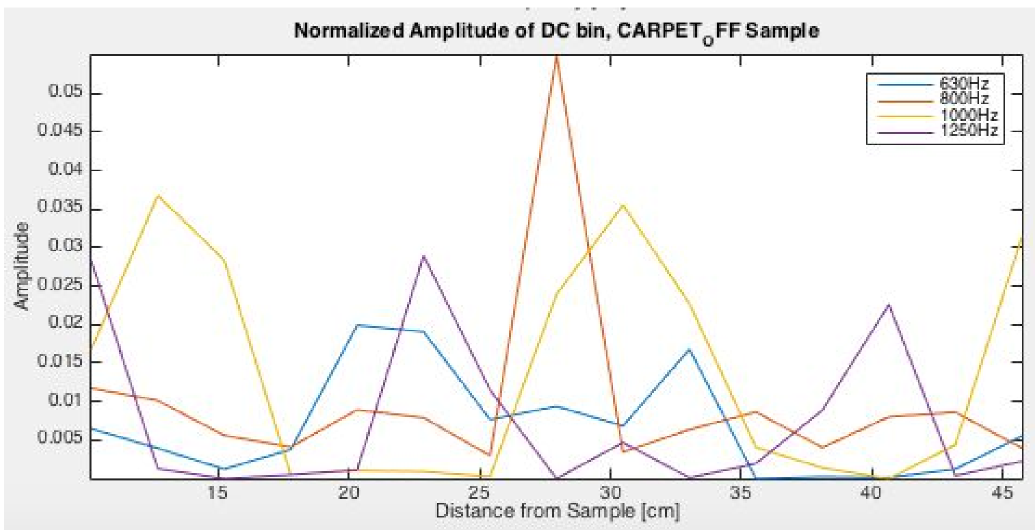
Normalized Amplitude of DC bin, CORKFLOOR Sample
Amplitude
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz

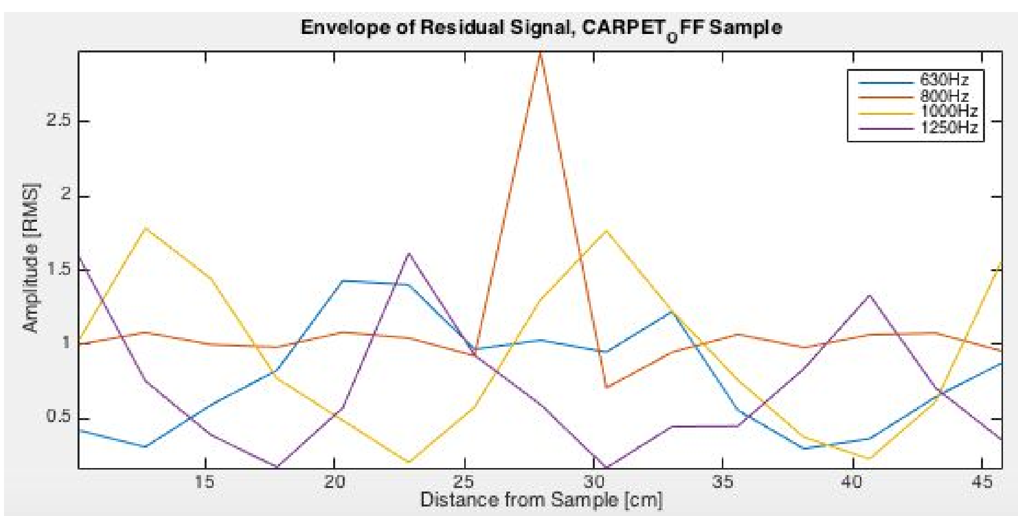


Envelope of Residual Signal, CORKFLOOR Sample
Amplitude [RMS]
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz

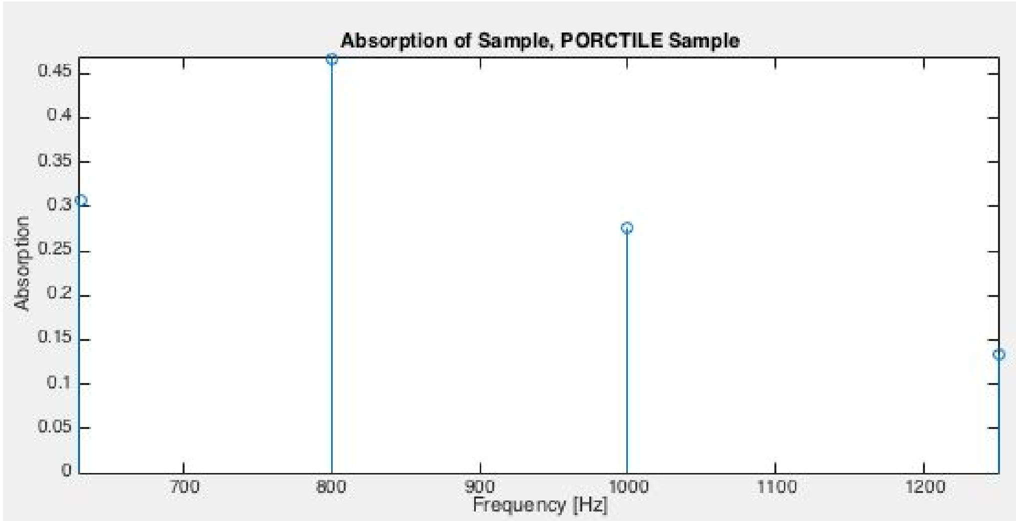
Carpet

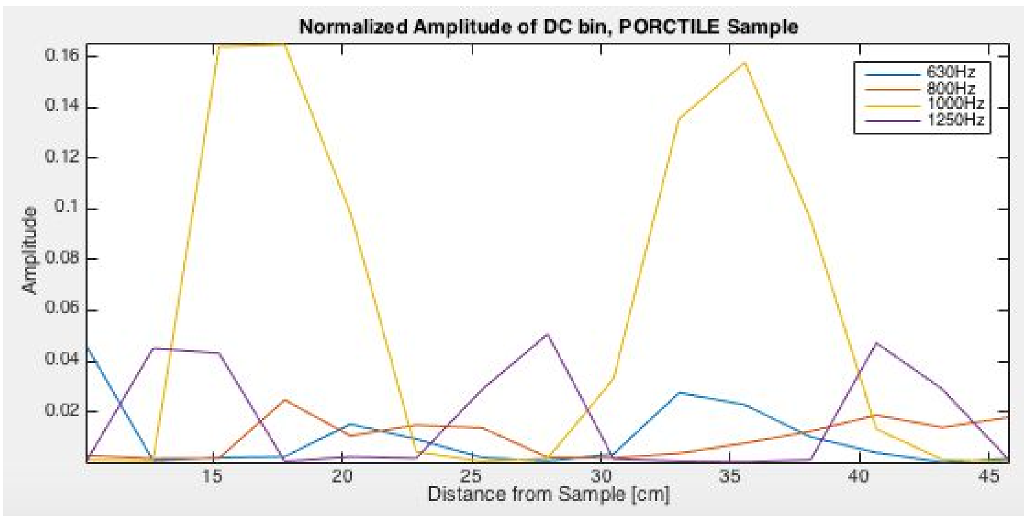

Absorption of Sample, CARPET_OFF Sample
Absorption
Frequency [Hz]
0
0.1
0.2
0.3
0.4
0.5
0.6
700
800
900
1000
1100
1200

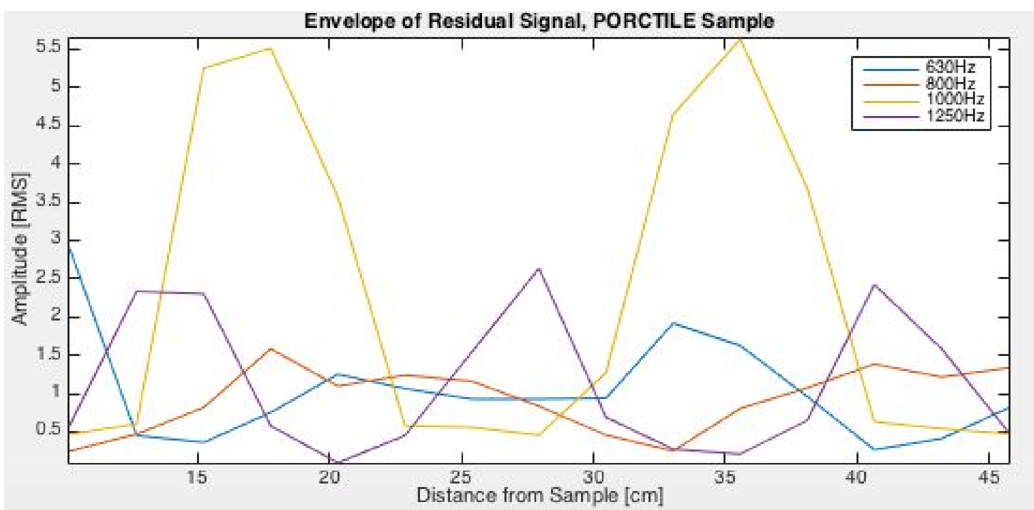

Normalized Amplitude of DC bin, CARPET_OFF Sample
630Hz
800Hz
1000Hz
1250Hz
Amplitude
Distance from Sample [cm]
0.005
0.01
0.015
0.02
0.025
0.03
0.035
0.04
0.045
0.05
15
20
25
30
35
40
45


Envelope of Residual Signal, CARPET_OFF Sample
630Hz
800Hz
1000Hz
1250Hz
Amplitude [RMS]
Distance from Sample [cm]
0.5
1
1.5
2
2.5
15
20
25
30
35
40
45

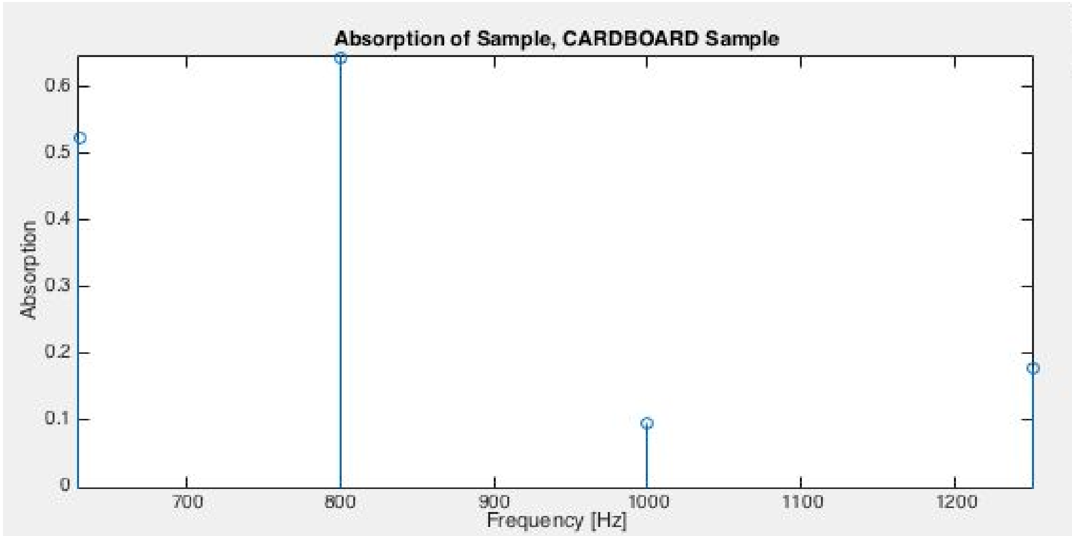
Tile


Absorption of Sample, PORCTILE Sample
Absorption
Frequency [Hz]
0.45
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
700
800
900
1000
1100
1200

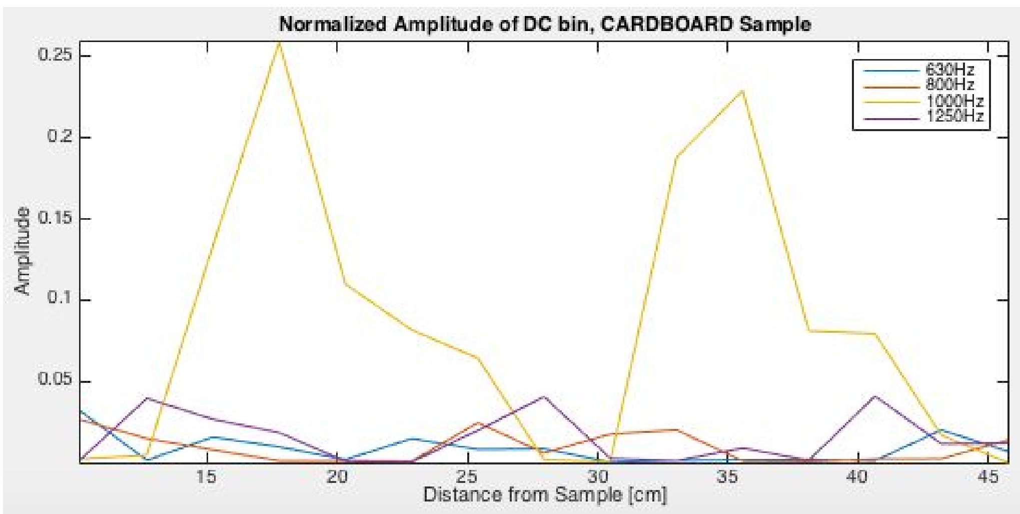

Normalized Amplitude of DC bin, PORCTILE Sample
Amplitude
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz
0.16
0.14
0.12
0.1
0.08
0.06
0.04
0.02
15
20
25
30
35
40
45


Envelope of Residual Signal, PORCTILE Sample
Amplitude [RMS]
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz
5.5
5
4.5
4
3.5
3
2.5
2
1.5
1
0.5
15
20
25
30
35
40
45

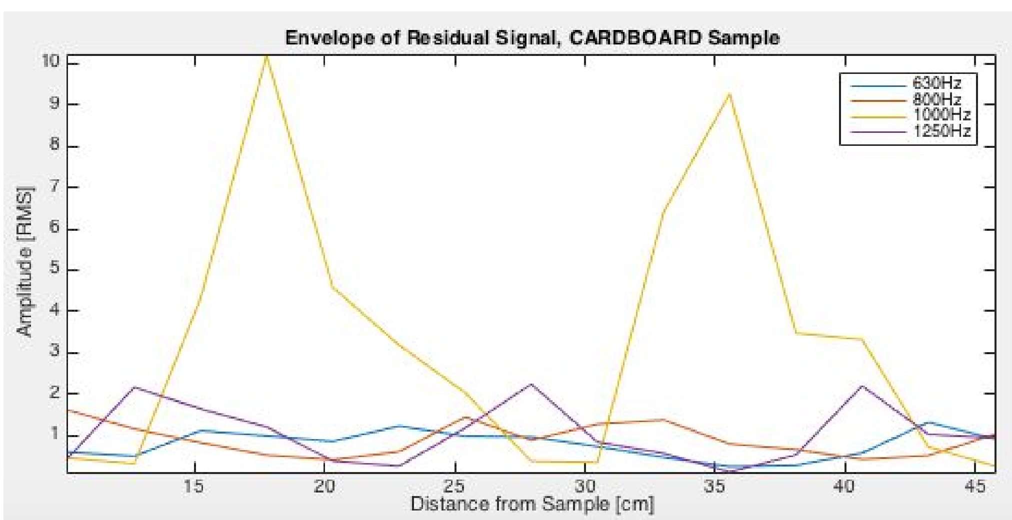
Cardboard



Absorption of Sample, CARDBOARD Sample
Absorption
Frequency [Hz]



Normalized Amplitude of DC bin, CARDBOARD Sample
Amplitude
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz



Envelope of Residual Signal, CARDBOARD Sample
Amplitude [RMS]
Distance from Sample [cm]
630Hz
800Hz
1000Hz
1250Hz

	680 Hz	800 Hz	1000 Hz	1250 Hz	Avg
Sponge	0.22481	0.77271	0.69577	0.60197	0.57
Glass	0.40975	0.57297	0.54337	0.23865	0.44
Cork	0.48019	0.82234	0.31275	0.3827	0.50
Carpet	0.56995	0.62172	0.36855	0.3419	0.48
Tile	0.30767	0.46705	0.27674	0.13375	0.30
Cardboard	0.52355	0.64525	0.096346	0.1772	0.36

	Calculated	Real Value	Error
Sponge	0.57	.0702 [5]	0.50
Glass	0.44	.12 [19]	0.32
Cork	0.50	.3 [19]	0.20
Carpet	0.48	.35 [19]	0.13
Tile	0.30	.1 [19]	0.20
Cardboard	0.36		

4.3 Discussion of Results

Although the results show some degree of error between the calculated results and documented results, there are several explanations that were outside of the scope of the project that may account for this. First of all, for every type of material, there are several potential variations on the material that may affect the results. It is impossible to tell if the material you are performing calculations on is the same exact material that others have calculated coefficients for. The obtained samples may also not have been pure, and some were only ¼” thick, which differs from ½” called for in the standards. One other issue is that many coefficients are calculated for different frequency ranges than done in this project, so the comparisons are really only an estimate. Lastly, sealing of the tube was not perfect, as the test signal could be heard from outside of the tube. Due to these factors, an error of 0.2 or less could be considered reasonable. However, the values generated by the tube seemed to be consistently high. In all of the residual and DC plots, the minima and maxima can always be seen, indicating that the overall process is functioning well enough for a standing wave to be detected. Although the resolution is not smooth because there are only 15 points per frequency, the peaks can still be clearly seen, so resolution is probably not too large of an issue. Further adjustment of the DC magnitude factor ξ may be necessary to further emphasize the standing wave and lower the coefficient values to closer to standard values.

5. COST

5.1 Parts

Table 5.1 Costs Analysis.

Part	Cost	Number Used	Total Cost
Motor Assembly	0.00	1	0.00
24” of 3” diameter PVC pipe	6.98	1	6.98
3” PVC Toilet Flange	3.75	3	11.25
3” PVC Termination Cap	3.95	1	3.95
Dayton Audio RS75-4 3" Driver	21.55	1	21.55
Sparkfun MEMS Microphone Assembly	9.95	1	9.95
20 AWG Wire	2.49	1	2.49
Arduino UNO	24.95	0	0.00
Arduino Nano	10.95	1	10.95
Passive Electronics Components (Inductors, Capacitors, Resistors)	5.00	1	5.00
JRC386 Current Driver IC	1.25	1	1.25
Breadboard	10.00	1	10.00
Alligator Test Leads	2.95	1 - pack of 10	2.95
CR2032 Battery and Holder	3.00	1	3.00
9V - 6xAA Battery Holder	1.00	1	1.00
Fasteners	10.00	1	10.00
Wood Supports - ¾” thick	1.50	3	4.50
Sample Materials (Carpet, Cork, Wood panelling, Granite)	Donated by Home Depot		

5.2 Labor

The fabrication of the impedance tube involved modification of readily available materials in order to keep cost at a minimum. The tube, sample housing, and driver enclosure were fashioned from 3” diameter PVC plumbing. The driver housing was filled with hydrophilic sponge and sealed with a PVC end cap in order to provide an anechoic termination. The sample housing was manufactured from two PVC toilet flanges that were drilled and fitted with fasteners in order to create an interlocking mechanism for an airtight seal. The stands for the impedance tube were manufactured from ¾” thick plywood in a U-shape with 3¼” spacing between the walls. The circuits were individually assembled and adjusted on breadboard before being reassembled on perforated circuit board. The rack and pinion motor assembly was recycled from a Hewlett-Packard inkjet printer.

6. CONCLUSIONS

This project served as a proof of concept for a fully-automated acoustic impedance tube.

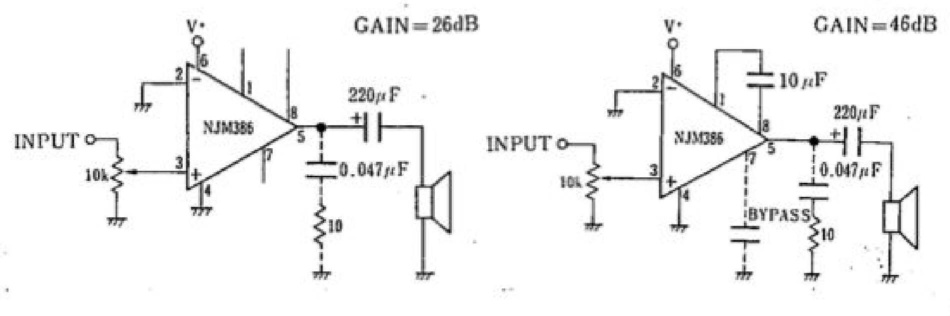
Through the realization of this project, we reinforced many concepts that were taught over the course of our undergraduate careers. Engineering Acoustics and Circuit Theory courses were the fundamentals of the knowledge set utilized in this project. Building a real world application using the concepts from academic courses presented a new set of challenges that cannot be effectively presented in a classroom. The teamwork and organization skills that were developed in the first semester of Senior Design helped us realize our goal by organizing our thoughts and setting up deadlines.

This project served as a proof of concept and the lessons learned can be applied to further the utility of the apparatus. A tube of longer length can provide a lower frequency range while a smaller diameter can provide a higher frequency limit. A larger number of recording distances can provide coefficients and residual plots of better resolution. A linear motion from a motor assembly can provide another step of automation. A SD card and a touchscreen shield using an Arduino, Raspberry Pi, or other microcontroller can also eliminate the dependency on using an external computer for the calculation of coefficients. A termination can be built with another microphone assembly in it to measure transmission loss through the sample material.

APPENDIX A: JRC 386 Datasheet Excerpts

Excerpt from JRC 386 datasheet displaying a typical configuration for a power amplifier as used in the loudspeaker amplifier.

■ **TYPICAL APPLICATION**



APPENDIX B: INMP401 (MEMS Microphone) Datasheet Excerpts

Excerpt from datasheet of the INMP401 microphone, which is used on the MEMS microphone circuit. These excerpts give dimensions and frequency response, specifications which are both highly relevant to the project.

InvenSense

INMP401

Omnidirectional Microphone with Bottom Port and Analog Output

GENERAL DESCRIPTION

The INMP401* is a high-quality, high-performance, low-power, analog-output bottom-ported omnidirectional MEMS microphone. The INMP401 consists of a MEMS microphone element, an impedance converter, and an output amplifier. The INMP401 sensitivity specification makes it an excellent choice for near-field applications. The INMP401 has a wideband frequency response, resulting in natural sound with high intelligibility. The specially designed low frequency cutoff reduces wind noise. Its low current consumption enables long battery life for portable applications.

The INMP401 complies with the TIA-920 *Telecommunications Telephone Terminal Equipment Transmission Requirements for Wideband Digital Wireline Telephones* standard.

The INMP401 is available in a 4.72 × 3.76 × 1.00 mm surface-mount package. It is reflow solder compatible with no sensitivity degradation. The INMP401 is halide free.

**Protected by U.S. Patents 7,449,356; 7,825,484; 7,885,423; and 7,961,897. Other patents are pending.*

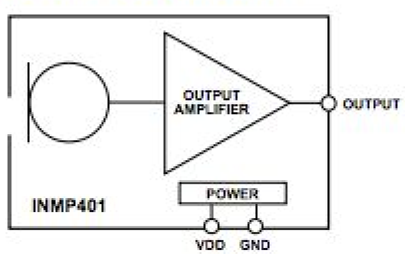
APPLICATIONS

- Mobile Devices
- Teleconferencing Systems
- Headsets
- Security Panels
- Intercom Devices

FEATURES

- 4.72 × 3.76 × 1.00 mm Surface-Mount Package
- SNR of 62 dBA
- Sensitivity of −42 dBV
- Flat Frequency Response from 60 Hz to 15 kHz
- Low Current Consumption: <250 µA
- Single-Ended Analog Output
- High PSR of 70 dB
- Compatible with Sn/Pb and Pb-Free Solder Processes
- RoHS/WEEE Compliant

FUNCTIONAL BLOCK DIAGRAM



ORDERING INFORMATION

PART	TEMP RANGE
INMP401ACEZ-R0*	−40°C to +85°C
INMP401ACEZ-R7†	−40°C to +85°C
EV_INMP401	—
EV_INMP401-FX	—

* – 13" Tape and Reel

† – 7" Tape and reel to be discontinued. Contact sales@invensense.com for availability.

TYPICAL PERFORMANCE CHARACTERISTICS

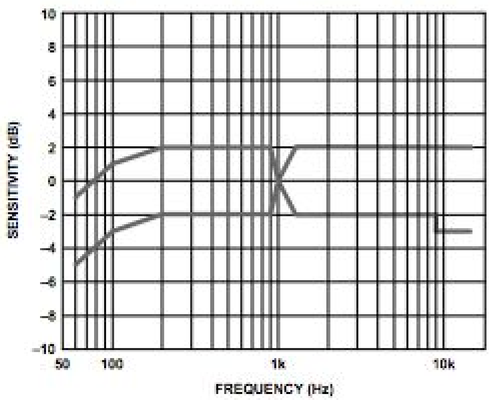


Figure 3. Frequency Response Mask

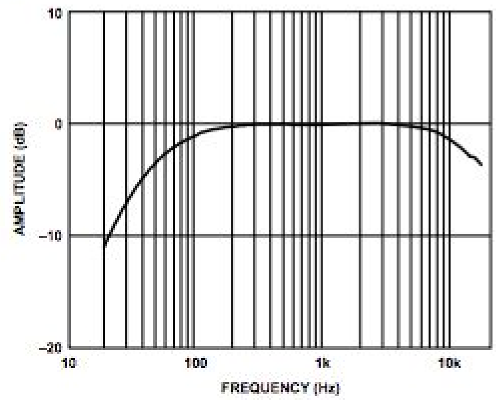


Figure 4. Typical Frequency Response (Measured)

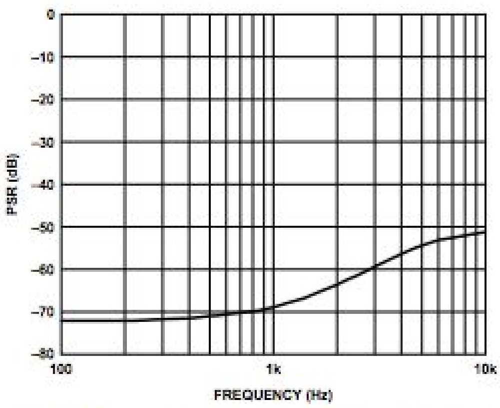


Figure 5. Typical Power Supply Rejection Ratio vs. Frequency

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