Acoustic characterization and optimization of a Krannert Center dance studio

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The Krannert Center for the Performing Arts at the University of Illinois Urbana-Champaign serves as a showcase for fine arts and a place for students to learn about music, dance, and performance. Room DRK2500, a dance rehearsal hall, suffers from a long reverberation time, which makes course instruction and music rehearsal difficult. The goal of this project is to reduce the reverberation time of the room using equipment made in-house and acoustic foam paneling provided by the Krannert Center. This report outlines some basic theory of acoustics, instrumentation, data analysis methodology, and describes an attempt at using acoustic panels to reduce the reverberation.

I. INTRODUCTION

A. DRK2500

Opened in 1969, the Krannert Center for the Performing Arts (KCPA) is the University of Illinois Urbana-Champaign's (UIUC) performing arts complex. It has several state-of-the-art performance facilities, as well as many workshops and rehearsal spaces. Previously, University of Illinois physics students measured the acoustic properties of the Foellinger Great Hall, the primary performance space. The Great Hall serves as an example of a performance area with ideal acoustic properties for its purpose.

DRK2500, however, is the subject of many complaints regarding its acoustical qualities, namely that its reverberation time is too long. Located in the basement of the KCPA (Figs. 1-3), this room is primarily used as a rehearsal hall for dance instruction, but it is also used as a performance space, often with live music from a piano and drum kit. The long reverberation time causes a significant loss in speech clarity while music and/or dancing is occurring. Thus, the goal of this project is to reduce the reverberation time of the room to a more acceptable level.

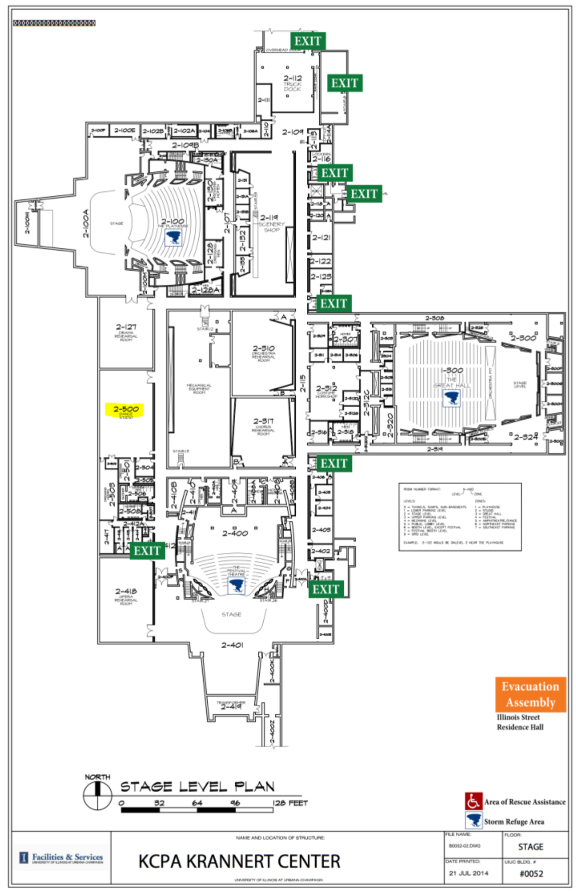

FIG. 1. Floor plan of the Krannert Center stage floor including dance studio DRK2500 (highlighted in yellow) [1].

B. SPL and RT60

The ISO (International Organization for Standardization) standard 3382 lays out several parameters for quantitatively describing the acoustics of a room. A room's reverberation time is quantified by its RT60, which is defined as the time required for the sound pressure level (SPL) of a tone to decrease by 60 dB [2].

SPL is a logarithmic scale of an acoustic wave's pressure and is given by

$$L_p = 20\log_{10}(\frac{p}{p_{ref}}), \quad p_{ref} = 20\,\mu Pa \tag{1}$$

where L_p is the SPL of the wave and p is the root mean square (rms) acoustic pressure of the wave measured in Pascals (Pa). Acoustic pressure is defined as

$$p = p_{total} - p_{atmospheric} \tag{2}$$

and is typically between 20×10^{-6} Pa and 60 Pa. When finding the difference in SPL, a reference pressure is un-

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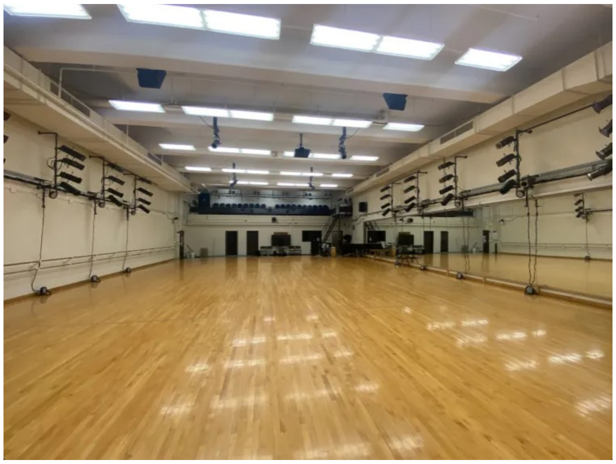

FIG. 2. DRK2500 as seen from a wall opposite to the observation deck.

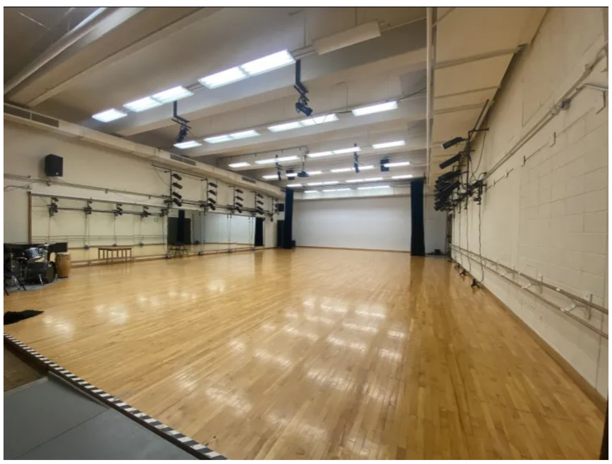

FIG. 3. DRK2500 as seen from the entrance, which can be seen on the left side of Fig. 2.

necessary since the properties of logarithms simplifies the equation to

$$\Delta L_p = 20\log_{10}(\frac{p_{final}}{p_0}), \tag{3}$$

where p_0 is the inital acoustic pressure (Pa). This simplification also allows peak or rms values to be used in the calculation of ΔL_p, since

$$\frac{p_{final,rms.}}{p_{0,rms}} = \frac{p_{final,peak}}{p_{0,peak}}, \tag{4}$$

as long as the choice is consistent between measurements. In this project, peak values are used for simplicity. [3]

One method for calculating the RT60 of a room relies on fitting the decay of the acoustical pressure to an exponential, which is true in an ideal setting. The equation for such a setting is

$$p = p_0 e^{k(t-a)}, \tag{5}$$

where a (s) is the time the decay starts, t (s) is the time passed since the start of the decay, and k ($\frac{1}{s}$) is the decay constant. Plugging Eq. 5 into Eq. 3 and setting SPL to -60 dB gives

$$-60 = 20\log_{10}(e^{k(t-a)}). \tag{6}$$

Solving for t, which is relabeled as T_{60} to represent the RT60 time, gives

$$T_{60} = \ln\left(\frac{0.001}{k}\right). \tag{7}$$

From the propagation of uncertainty principles, the uncertainty of the RT60 time, σ_{RT60}, can then be calculated from the variance of k, σ_k^2, which is the element $pcov_{1,1}$ in the covariance matrix returned by the `curve_fit` function:

$$\sigma_{\mathrm{RT60}} = \sqrt{\left(\frac{\partial T_{60}}{\partial k}\right)^2 \sigma_k^2}. \tag{8}$$

Applying the partial derivative:

$$\frac{\partial T_{60}}{\partial k} = -\frac{\ln(0.001)}{k^2}. \tag{9}$$

Thus, the uncertainty on RT60 is:

$$\sigma_{\mathrm{RT60}} = \left|\frac{\ln(0.001)}{k^2}\right| \sqrt{\sigma_k^2}. \tag{10}$$

Another method for calculating the RT60 of a room relies on finding the absorption constants of the materials in the room. In a room, the decay of sound is typically facilitated by absorption

$$A = \sum S_i \alpha_i, \tag{11}$$

where A is the total absorption of the room measured in sabins, S is the area of an absorbing surface in the room, α is the surface's absorption coefficient, and the summation is over all of the different absorbing surfaces in the room. This can be used to empirically determine the RT60 of a room using Sabine's formula

$$T_{60} = 0.161\frac{V}{A}, \tag{12}$$

where V is the volume of the room in meters (a different constant is necessary for different units). [4] Using the dimensions of the room and properties of the walls, one can obtain an estimate of the RT60 time.

DRK2500 is 13.55 meters wide, 22.45 meters long, and 6.01 meters in height. The room also includes an observation deck, which includes two rows of seats for instructors and students to watch practice sessions. It has a height of 2.58 meters, a width of 13.55 meters, and extends 2.48 meters beyond the rest of the hall. These measurements were done using a laser distance meter.

Treating the dance hall as a perfect rectangular prizm (ignoring the observation deck), assuming the entire floor is made of wood, and that the remaining walls are made of concrete, the RT60 time of the room should be roughly 7.46 seconds. While the 'ideal' RT60 time is a subjective matter, typical RT60 times for multi-purpose auditoriums are 1.5-1.8 s. [5] Qualitative analysis of the room makes it evident that the RT60 time is not nearly as bad as the estimate suggests, likely due to unconsidered geometries of the room, additional absorbing surfaces (such as cushioned chairs), and lower absorption coefficients than estimated.

Two approaches are used to determine the RT60 time in this work. The first method involves recording the reverberation from a single frequency tone emitted by a speaker. The second method is the industry standard, which is outlined by Schroeder [6], and involves providing an impulse and measuring the response. In this case, a balloon pop in the center of the room is used to emulate a Dirac-delta function source.

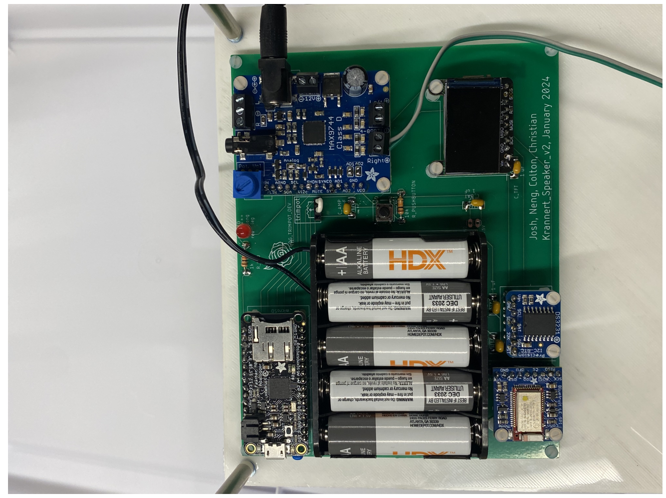

FIG. 4. The PCB of the speaker board is attached to a 3D printed box with four legs.

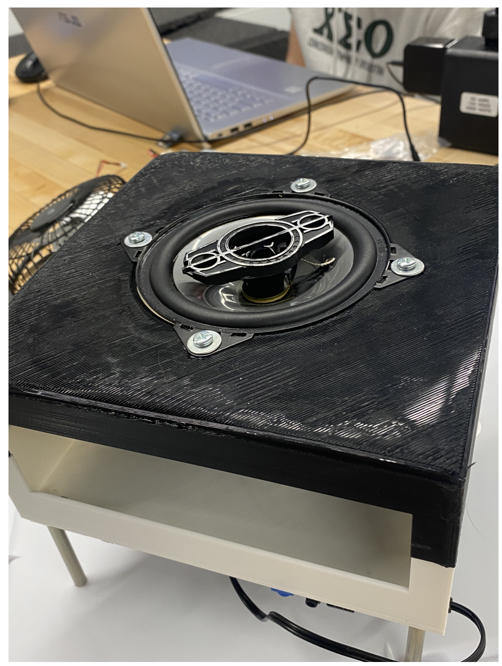

FIG. 5. The speaker is screwed into the top of the 3D printed box such that the cone is facing outward.

II. INSTRUMENTATION

Two printed circuit boards (PCBs) designs are used in this project: a speaker board, which is used to produce the single frequency tone used in the first method, and a microphone board, which is used to record data in both methods.

A. Speaker Board

The speaker board (Figs. 4 & 5) includes an Adafruit Bluefruit low energy (LE) serial peripheral interface (SPI) Friend, an Adafruit Feather M0 Express Adalogger, an Adafruit MAX9744 Class D amplifier (powered by five AA batteries), and an Adafruit 20 watt 4 Ohm full range speaker, model XS-GTF1027.

The board activates upon receiving a trigger signal from a phone via Bluetooth® using the Bluefruit LE SPI Friend and the Bluefruit Connect phone app. The Adalogger then sends a 220 Hz and 450 Hz tone to the Adafruit MAX9744 amplifier via an I2C protocol with a pause in between the tones. The amplifier's gain is controlled by a built-in potentiometer, though it is possible to use an external potentiometer if digital volume control is desired. Since the speaker's range is from 60 Hz to 24 kHz [7], it is well-suited for this project, given that the human ear's hearing range is from 20 Hz to 20 kHz. [8]

The first iteration of this device produced square waves, which was problematic. Square waves are created by combining the fundamental frequency with many higher order frequencies, which makes it difficult to pick out the desired fundamental. The microphone boards were particularly sensitive to third and fourth order harmonic frequencies present in the square wave. To prevent this, a sine wave is generated in place of the square wave.

Producing a sine wave mostly eliminates the problem of higher order frequencies appearing in a spectrogram. Sine waves, however, are more difficult to produce, since they require an analog approach. The current version of this device simply emulates an approximate sine wave. This is done by filling an array with values of one period of a sine wave at a desired frequency. The array is then read over and the values are output to the digital-to-analog converter.

B. Microphone Board

The microphone board (Fig. 6) is controlled by an Adafruit Feather M4 Express, which utilizes the AT SAM D51 G19A micro-controller. The Feather M4 was chosen to take data over the M0 primarily due to its flash memory. The process of continuously writing data to the SD card over SPI is too slow, so the length of samples is limited by the size of the Ferroelectric random-access memory (FRAM) of the chip in which the data get stored. The M4 has 512 KB of FRAM, allowing the storage of over 2.5 seconds of 12 bit analog input data taken at 37 kHz. The M0 only has 256 KB of FRAM which would halve the data taking duration. In addition, the M4 has a 120 MHz clock allowing it to run considerably faster than the M0, which only has a 48 MHz clock.

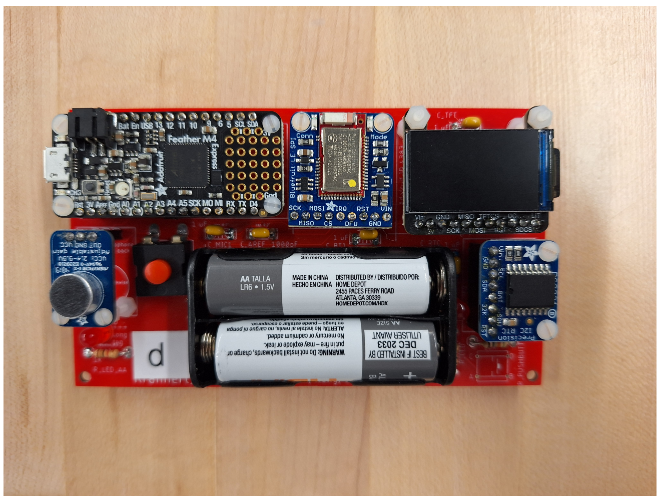

FIG. 6. A picture of a microphone board.

In practice, the increased speed is not utilized. Analog input data are read using Arduino's `analogRead` function, which is considerably slower than the maximum speed of the analog inputs. It is worth noting that, utilizing timer interrupts, data can be taken at over 44.1 kHz, which is the standard audio format for CDs (known as Red Book Audio). [9] Since this was above the upper frequency of human hearing and the maximum operating frequency of the microphone is only 20kHz, sampling at 44.1 kHz was much faster than what was necessary.

The microphone boards use an Adafruit Electret Microphone Amplifier which is set to the minimum gain, since tests done at higher gains produced clipped data. The Adafruit electret microphone is ideal as it has a large dynamic range, advertised to be 20 Hz to 20 kHz and is inexpensive compared to dynamic microphones. The Adafruit chip also includes a Maxim MAX4466 amplifier, which is an op amp purpose-built for amplifying microphones. This allows the microphone to be plugged directly into the Arduino's analog inputs as shown in Fig. 15.

Similarly to the speaker board, Adafruit's Bluefruit LE SPI Friend breakout board is used to communicate with the device. After receiving the same trigger signal that starts the tone generation, the microphone board waits for two seconds and then starts taking and saving data. This delay gives the speaker plenty of time to fill the room with sound. Data can be taken one trial at a time, or a sequence can be programmed to take multiple sets of data at a range of frequencies. Additionally, many boards can all be started at the same time. The temporal accuracy of starting the devices with Bluetooth was tested using a clap from a wood block equidistant from two devices. They were found to be accurate to 1.5 ms of each other.

The microphone board also contains a thin-film-transistor (TFT) display with a built in SD card reader (the Adafruit 1.14" 240x135 Color TFT Display + MicroSD Card Breakout) to store data sets and provide the user with the device's status. Finally, the board uses a real time clock (the Adafruit DS3231 Precision RTC Breakout) to provide data files with helpful names.

III. INVESTIGATING ACOUSTIC PRESSURE

To estimate the time at which a signal reaches -60 dB in pressure, two different curve fitting equations that model the signal behavior in their respective domains are used in this method. The first is a linear equation applied when the signal is kept at reference pressure level P_0, and the second is an exponential decay function applied when the signal falls below P_0. This can be treated as a piece-wise function,

$$y = \begin{cases} P_0 sin(ft) & \text{for } t \leq t_0 \\ P_0 sin(ft) e^{k(t-a)} & \text{for } t > t_0, \end{cases} \tag{13}$$

where t_0 (s) is the time at which the sound is cut off and f (Hz) is the frequency of the wave. The RT60 time of the room can then be calculated using Eqs. 6 & 7.

This method was first tested in a small chamber made out of acoustic paneling. These tests were able to determine an RT60 time as shown in Fig. 7, showing that this method has merit when the room is geometrically simple and the decay time falls within several periods of the tone.

In the dance hall, the speaker board was placed on a drum seat in the middle of the room. The seat also conveniently served as a way to dampen the noise from the speaker's vibrations, since in testing the speaker's uneven legs caused it to vibrate when producing sound, which created noticeable noise when placed on a hard surface. The four microphone boards were placed near the corners of the wooden floor of the dance hall, roughly six feet from the walls.

Fig. 8 shows data from one microphone. Some interesting artifacts appear in the data. For one, there is a peak which appears before the decay. The source of this peak is currently not known, as it cannot be consistently reproduced over multiple trials. There is also a

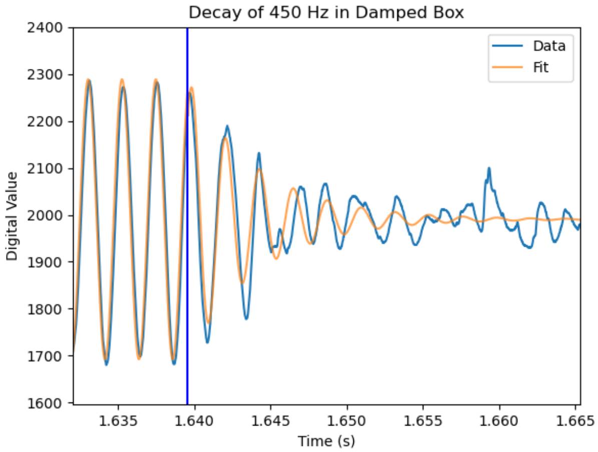

FIG. 7. The decay of a 450 Hz tone in the acoustic panel box. Here the blue line denotes where the tone was switched off. The calculated RT60 time is 0.032 ± 0.001 seconds.

bulge in the data after the drop, which can most likely be explained as an echo.

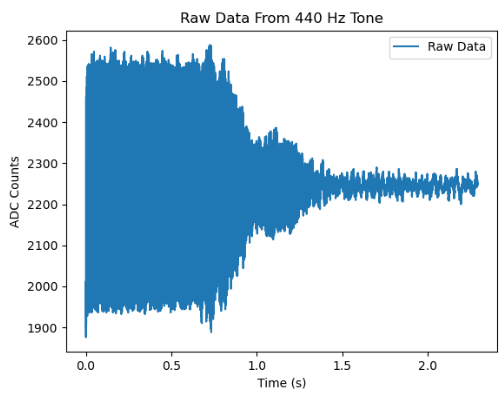

FIG. 8. Recorded data from a device placed near a wall.

Due to the unpredictable nature of these artifacts, an exponential fit was deemed useless. The model does not account for these anomalies. This necessitates the use of a different approach.

IV. SCHROEDER'S METHOD

The 1965 paper by Schroeder introduces the standard procedure for determining reverberation time.[6] Initially, one must gather the room's impulse response. In this project, the signal from a popped balloon is recorded. Next, a band-pass filter is used to focus the impulse response within a specific frequency range. The envelope of the signal is then extracted using a Hilbert transform, followed by smoothing with a moving average filter. The signal is subsequently converted to decibels. Finally, Schroeder's integration technique is applied, and the linear decay portion of the signal is fit to determine the reverberation time. The four methods used to process the raw data described above are:

1. *Band-pass filter*

Band-pass filters allow only a specific range of frequencies to pass through while attenuating or rejecting frequencies outside this range. A Butter-worth filter from the Scipy library is used for data processing. It is designed to attenuate frequencies exponentially outside the specified range. By employing the discrete time Fourier transform, one can calculate the amplitude for each frequency bin. The amplitudes within the desired frequency range can be kept unchanged and the amplitudes outside this range can be set to zero or have an exponential decay applied to them. Finally, the signal is converted back from the frequency domain to the time domain using the inverse discrete Fourier transform. The Discrete-time Fourier transform and Inverse Discrete-time Fourier transform are defined as:

$$f[k] = \frac{1}{N}\sum_{n=0}^{N-1} F[n] \cdot e^{i\frac{2\pi}{N}nk}, \tag{14}$$

$$F[n] = \sum_{n=0}^{N-1} f[k] \cdot e^{-i\frac{2\pi}{N}nk}. \tag{15}$$

- $f[k]$, where k is the frequency index ranging from 0 to $N-1$, represents the Fourier coefficient for the frequency component, encapsulating both amplitude and phase.
- $F[n]$, where n is the time index ranging from 0 to $N-1$, represents the sample of the original signal at time index n.

2. *Hilbert transform*

Many applications involve measurements that result in time signals containing rapidly oscillating components. When finding the reverberation of a signal, any rapid oscillations within the signal (frequency of the tone) are not of interest. Instead it is useful to understand how the amplitude of the oscillations varies over time. The amplitude of the oscillation varies slowly with time, and the shape of this slow variation is called the 'envelope'. By using the Hilbert transform, the rapid oscillations can be removed from the signal to produce a direct representation of the envelope alone. Here are two important features of the signal after the Hilbert transform: first, its removal of the oscillations allows for a detailed study of the envelope; second, the signal after the Hilbert transform is a positive function. The Hilbert transform $\mathcal{H}[g(t)]$

of a signal $g(t)$ is defined as:

$$\mathcal{H}[g(t)] = g(t) * \frac{1}{\pi t} = \frac{1}{\pi}\int_{-\infty}^{\infty} \frac{g(\tau)}{t-\tau}\, d\tau. \tag{16}$$

Given the convolution operation denoted by $*$, for each t, the convolution formula can be represents the area under the product of $g(\tau)$ and $\frac{1}{\pi(-\tau)}$ shifted by the amount t.

3. *Moving Average filter*

A moving average filter is employed to smooth out the data. This filter averages every set of M points of data together. The intent is to flatten any bumps that may result in deviations from linear decay after integration. While this step is arguably redundant as the integration effectively 'averages' chunks of data on its own, it was still performed to ease visual analysis of the data. The frequency response of moving average filter given by:

$$H(f) = \frac{\sin(\pi f M)}{M\sin(\pi f)}. \tag{17}$$

M was set to 350 as $s/100$ where s is the sampling rate of data taking was typically used in literature. [10] RT60 times were calculated with and without this filter for preliminary trials with results differing by less than $0.01s$.

4. *Schroeder Integration*

The curve can be further smoothed for calculations by using the Schroeder Integration method of envelope (also known as inverse time integration). In this step the sound pressure can also be converted to decibels. The Schroeder Integration is given by:

$$L(t) = 10\log_{10}\left[\frac{\int_t^{\infty} h^2(\tau)d\tau}{\int_0^{\infty} h^2(\tau)d\tau}\right] \tag{18}$$

where $h(\tau)$ is the smoothed envelope of the original signal after applying the Hilbert transform and a moving average filter. After obtaining the signal $L(t)$, we can perform a linear fit to determine the RT60 time.

A. Results

RT60 times were calculated by fitting a line to the decaying portion of the inverse time integrated data. An example of this is seen in Fig. 9. Here the decay time is calculated over 30 dB of decay. The linear fits starts from -5 dB as the first 5 dB encompass what is known as the early decay time (EDT) which is typically a much sharper drop than the following decay.

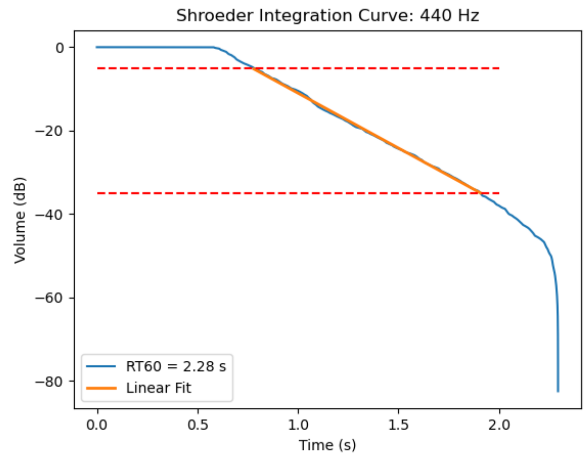

FIG. 9. The decay of the 440 - 880 Hz band of an impulse response in the studio. Here the red lines show the -5 dB and -35 dB levels between which the analysis takes place.

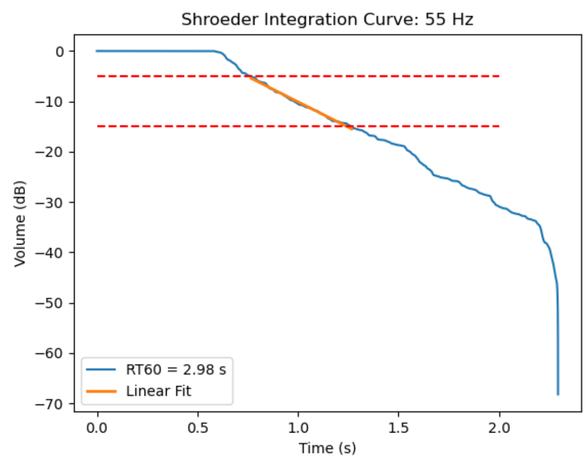

FIG. 10. The decay of the 55-110 Hz band of an impulse response in the studio. Here the red lines show the -5 dB and -15 dB levels between which the analysis takes place.

For other frequency bands there is not 30 dB of signal to analyze so the linear fit only encompasses 10 or 20 dB. An example of this can be seen in Fig. 10.

The fluctuations in the data from the linear fit arise from differing paths that the sound can take within the room. [6] In order to account for this, data from 5 microphone boards are averaged. The errors of the RT60 time can be found from the standard error of the fit, but this value is typically several order of magnitudes smaller than the variation between devices and trials. Because of this, the error is simply found from deviation between multiple trials.

Using this method, the RT60 time at different frequency bands can be calculated by varying the range of frequencies in the band pass filter. The RT60 time in the studio across various frequency bands can be seen in Fig. 11.

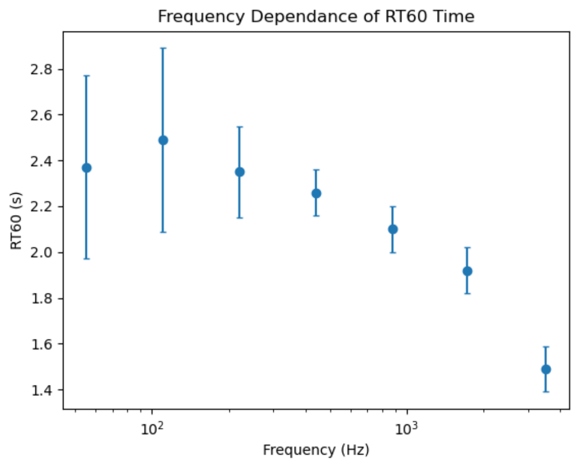

FIG. 11. The frequency dependence across octave ranges from 55 Hz to 7040 Hz. The locations of the points correspond to the lower end of their frequency band, and the frequency is in a logarithmic scale for ease of viewing. 440 - 1720 Hz were measured over 30 dB of decay, while other frequency bands were measured over 10 dB of decay.

It would be useful to compare this method to the investigation of acoustic pressure in Sec. III, but we were unable to create an impulse response quiet enough for use in the acoustic panel chamber.

V. EFFECT OF ACOUSTIC PANELING

The analysis of impulse responses using the Schroeder integration technique can be used to investigate the effectiveness of acoustic paneling at reducing reverberation time. This was done by measuring the RT60 time as discussed in Sec. IV with a varying amount of acoustic panels covering the walls of the studio. Panels were added to the back and right wall as shown in Fig. 12, with an even distribution across the two walls. This is not the exact location that panels would be placed in, but the assumption is made that it does not matter where they are covering the surface, only which material they are covering.

Data were taken at 6 different amounts of paneling, with RT60 times averaged between 5 microphone boards and 2 trials. Results from this test for the 440 - 880 Hz octave can be seen in Fig. 13. Results for other frequency bands can be seen in Appendix B. While there appears to be a general decrease in RT60 time with added panels, the results are inconclusive. It may be possible to further refine these data by taking many more trials or increasing the total number of panels used.

FIG. 12. The placement of 61 panels along the back and right wall of the studio. 25 were placed against the back concrete wall, 36 were covering the right concrete brick wall. Two of the microphone boards are located in the red circle in the middle of the room (near the sound source), one is in the red circle at the far end, and the other two are each in a corner outside of the picture denoted by the arrows.

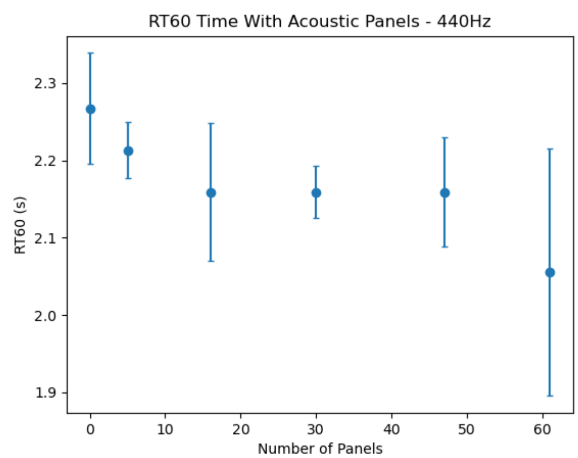

FIG. 13. The effect of acoustic paneling on decay time for 440 - 880 Hz. The RT60 time was extrapolated from 30 dB of decay.

VI. CONCLUSION

This work outlines two experimental techniques in determining the RT60 time of a performance hall. The first method involves creating a sound with a single frequency and recording the reverberation after the sound is cut off. The second is the industry standard outlined by Schroeder.

While fitting directly to the raw data when using the acoustic foam box worked quite well, this method falls apart when presented with anomalies or artifacts in the raw data. These could include the presence of higher-order frequencies, noise from outside the room, or internal noise from the microphone PCBs. These challenges make direct fits to the raw data impractical, as it is difficult to obtain a reliable RT60 time. Consequently, no tests using acoustic foam paneling were performed with this method.

Generally speaking, RT60 times were easier to determine using Schroeder's method. However, for many frequencies, it was difficult to determine a clear trend between RT60 time and the number of sound absorbing panels. Despite this, there are some frequencies where a trend is visible.

In future trials using impulse measurements, it will be useful to make sure the air pressure in all the balloons is consistent. This will ensure that the impulse sounds of the balloon pops are consistent across trials. It may also be interesting to try different impulses. In practice, starter pistols and dedicated clapping devices are often used.

It would also be necessary to try creating the impulse in different locations around the room. In this work, balloons were only popped in the middle of the room. This could be useful in determining areas with problematic acoustic responses. Knowing this will be helpful in identifying areas to focus on when improving the sound quality of the room.

Moreover, a greater number of acoustic absorption panels will be needed to see an appreciable difference in RT60 times across all frequencies. The most used in this work was 61. Having a greater number of panels could also allow for more flexibility in testing panel orientations and placements throughout the room.

ACKNOWLEDGMENTS

We would like to thank Professor George Gollin and Professor Yuk Tung Liu for their contributions, support, and guidance throughout the duration of this project. We would also like to thank Professor Rick Scholwin for giving us access to the dance center and for providing acoustic foam for our tests. We would also like to thank Ivan Velkovsky for their help in writing analog input software.

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Appendix A: Schematics

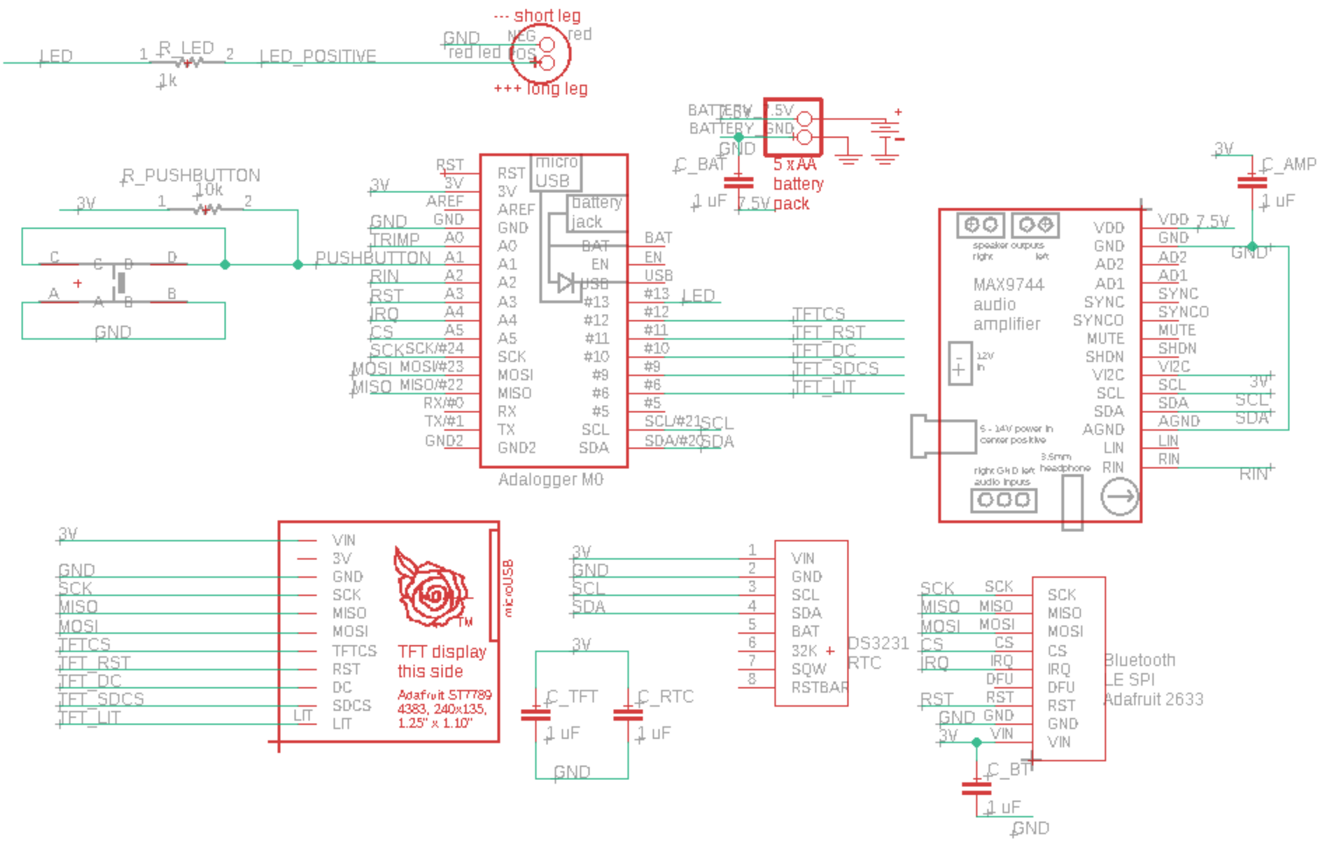

FIG. 14. Schematic of the speaker board.

Appendix B: Acoustic Paneling Results

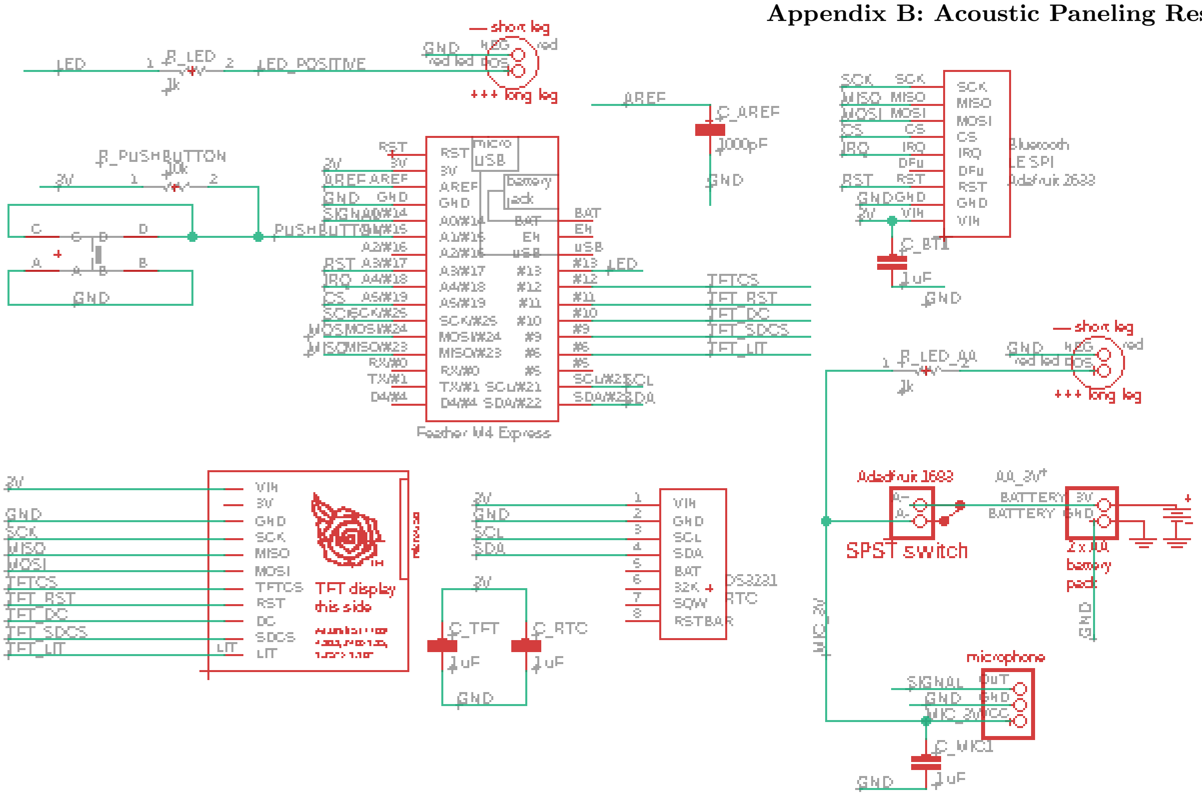

FIG. 15. Schematic of the microphone board.

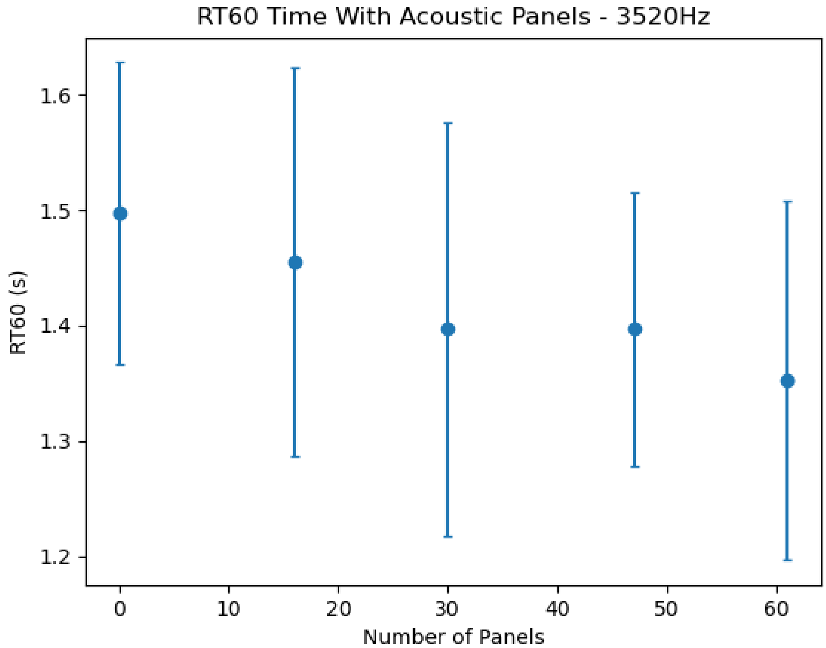

FIG. 16. The effect of acoustic paneling on decay time for 3520 - 7060 Hz. The RT60 time was extrapolated from 10 dB of decay.

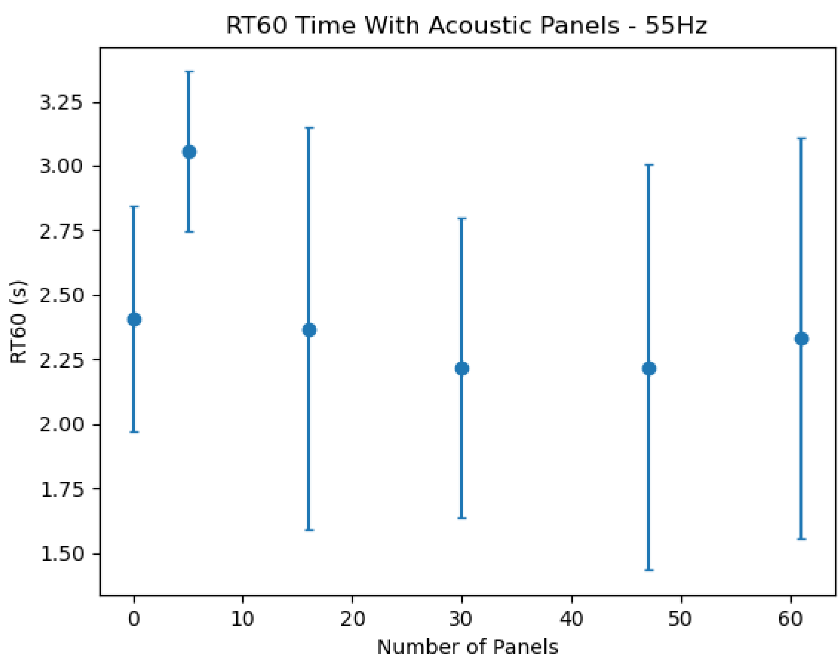

FIG. 17. The effect of acoustic paneling on decay time for 55 - 110 Hz. The RT60 time was extrapolated from 10 dB of decay.