Planning Acoustics and Recording Equipment for a Live Stream Studio

An Interactive Qualifying Project Report
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Table of Contents

Abstract iii

Executive Summary iv

1. Background 1

- **1.1 History of Live Streaming** 1
- **1.2 Room Acoustics** 2
- **1.3 Recording Equipment** 9

2. Methodology 13

- **2.1 Acoustic Treatment Requirements and Calculations** 13
- **2.2 Acoustic Treatment Plan** 17
- **2.3 Audio Equipment Product List** 20
- **2.4 Cables** 26
- **2.5 Software** 28

3. Room Setup 30

- **3.1 Wiring Guide** 30
- **3.2 Acoustic Calibration** 31

4. Conclusion 32

References 33

Appendix A: Python Code 35

Appendix B: Room Renders 37

List of Figures

Figure A: Acoustic Designs for the Studio and Control Rooms **vii**

Table A: Cost Breakdown of Audio Equipment **ix**

Figure B: Audio Equipment Wiring Guide **xi**

Figure 1.1: First Three One-Dimensional Standing Wave Modes **5**

Figure 1.2: Standing Waves in a Studio Setting **6**

Figure 1.3: Three Types of Room Modes **6**

Figure 1.4: Example of a Commercial QRD **9**

Figure 1.5: Diagrams of Dynamic and Condenser Microphones **10**

Figure 2.1: LEDE Control Room Conceptual Diagram **15**

Figure 2.2: Room Modes Plot **16**

Figure 2.3: Acoustic Designs for the Studio and Control Rooms **17**

Table 2.1: Cost Breakdown of Acoustic Treatment **19**

Table 2.2: Cost Breakdown of Audio Equipment **21**

Figure 2.4: SM-58 and SM-57 Microphones **23**

Figure 2.5: Focusrite Scarlett 18i20 **25**

Table 2.3: Cost Breakdown of Audio Cables **28**

Figure 3.1: Audio Equipment Wiring Guide **31**

Abstract

This paper is meant to serve as a guide to the audio-related aspects of converting Worcester Polytechnic Institute's Riley Hall basement rooms into a live streaming studio. The recording room was designed using mathematical analysis of reverberation time and resonant frequencies. The second room, meant for audio mixing and stream editing, was based around the live-end dead-end (LEDE) design philosophy. Lists of necessary equipment and software are presented along with the cost-benefit reasoning behind each choice.

Executive Summary

The goal of this project was to plan the audio-related aspects of a recording studio. This included acoustic treatment of the room as well as the equipment used for audio recording. The studio itself was adapted from two adjacent rooms located in the basement of Worcester Polytechnic Institute's Riley Hall. One room was converted into a recording studio, while the other served as the control room with computers and technicians. A detailed cost analysis was provided for each aspect of the plan. The budget for the entire studio was approximately $75,000. This paper aimed to provide a plan that takes up only a small fraction of this budget, leaving plenty of room for non-acoustic equipment such as high-end streaming computers and cameras.

Historically, live streaming has existed in some form or another since the late 19th century, when musical performances could be transmitted over telephone lines. Modern internet infrastructure makes live streaming and viewing available to billions of people worldwide. As technology continues to advance, new industries are developing around the practice of live broadcasting. In addition to facilitating live internet performances of all sorts, having a studio space at WPI will allow students to gain valuable experience working with modern streaming technology.

This paper began by explaining the basic concepts required to plan a room with high-quality acoustics. *Absorption*, the first acoustic property discussed, determines how reverberant a room sounds, and can be estimated using the Sabine equation. A room with too much reverberation will sound spacious, but this is undesirable when recording speech. Meanwhile, a room that is too absorptive may sound "dry" or "dead". It is important to balance high frequency absorption with that of lower frequencies. Different types of absorptive elements, such as

acoustic panels and porous bass traps, can be incorporated into the room design to create the desired frequency response.

Diffusion is another important property that determines how evenly the sound is distributed about the room. A highly diffuse room is also less susceptible to comb-filter effects, which occur when a sound bounces off of opposing walls and interferes with itself. As a rule of thumb, concave surfaces are often better at diffusing sound than concave ones, which tend to focus sound. Specialized structures called *quadratic residue diffusers* use a mathematical sequence to optimize diffusion at a wide range of frequencies.

Next, there are *room modes*, which are specific frequencies at which standing waves develop between the walls of the room. These can cause very noticeable timbral defects, especially in a space as small as the one in Riley Hall. Resonating absorbers are acoustic devices that absorb only a narrow frequency band, and can be used to selectively attenuate room modes.

Finally, there are *external sounds*, which originate from outside the studio and can show up in a recording. These originate from a wide range of sources, and each source requires a different treatment method.

After these basic concepts were introduced, the paper moved on to the methodology section. As already mentioned, the budget goal was that the acoustic costs should only take up a small portion of the total $75,000 budget. The methodology section detailed the process behind planning the acoustic design and provided a full equipment list for the rooms.

According to research, the ideal reverberation time for both speech and music in a room with 1,500 cubic feet is approximately 0.45 ms. Using the Sabine equation, the amount of commercial acoustic paneling required to bring the reverberation time down to this value was calculated to be about 160 square feet. Porous bass traps installed in the corners of the room would provide absorption at lower frequencies. Adequate diffusion was achieved using wooden quadratic residue diffusers. The “room mode equation” was used to determine what frequencies need to be reduced by resonating absorbers, although the design and construction of these absorbers were left for future research. Finally, a brief description of external sound reduction discussed the use of caulk and soundproof doors to acoustically isolate the studio room. Further exploration of soundproofing options was also left for future research. One wall was left mostly empty for a television to allow communication with technicians in the control room while the doors are closed.

The control room was designed around the *live-end dead-end* (LEDE) philosophy. As this room is only used for audio playback, the acoustics only needed to be optimal in a single listening zone. Absorptive panels were laid only on the side of the room with the loudspeakers to prevent reflected sound from causing comb-filter interference. The other end of the room is left reflective (or “live”) to avoid absorbing too much sound.

The final acoustic designs for both rooms are presented here in Figure A.

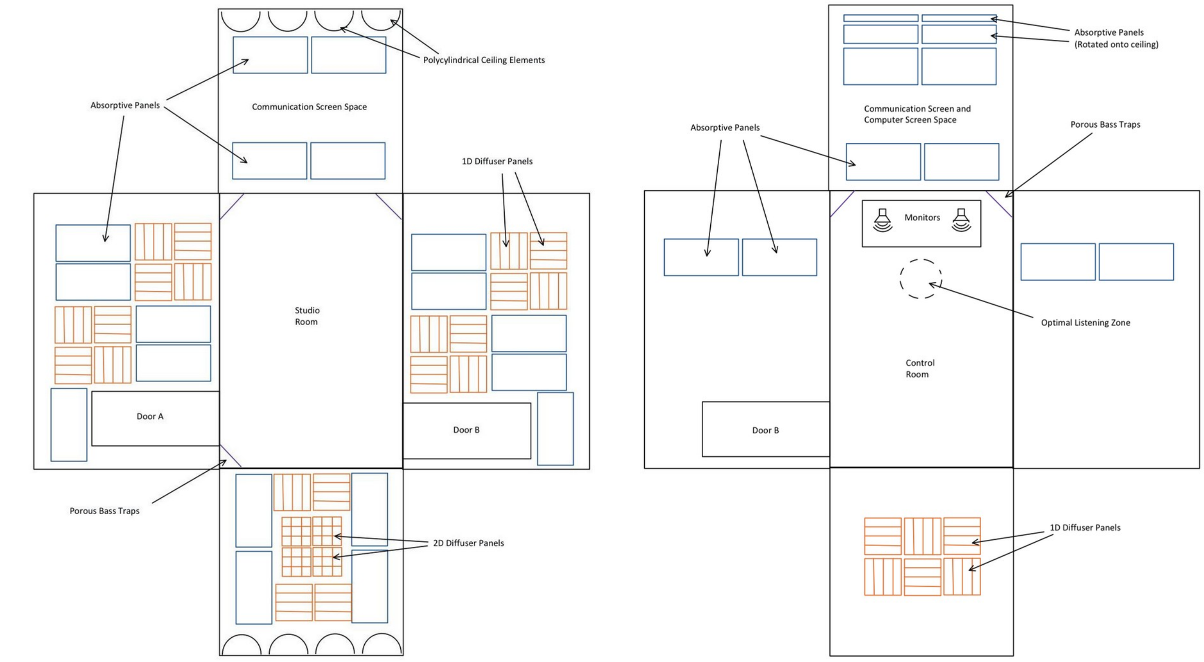

Figure A: Acoustic Designs for Studio and Control Rooms

By comparing different acoustic products across a range of vendors, the cost of such a design was estimated to be close to **$5,320** (excluding the costs of resonating absorbers, external soundproofing, and installation).

For recording equipment, the room required microphones, speakers, an audio interface, and other accessories such as stands and shock mounts. The full product list is provided below in Table A. The room itself was quite small; It would likely only be able to accommodate a maximum of six people. Depending on whether you are recording a loud instrument or a soft voice, different kinds of microphones are better suited for the task. A full selection of six dynamic and six condenser microphones should reasonably cover all possible use cases. Two of the dynamic microphones were better suited for speech, and the condenser microphones had a variety of diaphragm sizes for different uses. An additional measurement microphone, the EMM-6, was included for calibrating room acoustics. Providing accurate audio playback for the audio mixing technician in the control room required two studio monitors and a subwoofer. At Professor Bianchi's suggestion, the list included the JBL 503P MkII monitors and the LSR310S subwoofer. Generic stands and mounts for the microphones and speakers were included in the cost analysis as well. Finally, an audio interface was necessary to connect all the microphones and speakers to the mixing computer. For this purpose, the Focusrite Scarlett 18i20 has enough ports to support an eight-microphone setup.

There would likely be two computers in the control room, one dedicated to audio mixing and the other just for streaming. There needed to be some way to send the mixed audio from the audio computer to the streaming computer. The difficult part was finding a way to do this without losing audio quality in the process. The solution this paper suggested is to have a smaller, secondary audio interface connected to the streaming computer. The output of the audio interface from the mixing computer can then be plugged directly into the input of the secondary interface via two TRS-to-TRS cables. For brand consistency, we suggested the Focusrite Scarlett 2i2.

Table A: Cost Breakdown of Audio Equipment

Item	Product Name	Vendor	Count	Unit Price	Total Price
Dynamic Microphone 1	SM57	Shure	4	$100	$400
Dynamic Microphone 2	SM58	Shure	2	$100	$200
Small Diaphragm Condenser Microphone	VMS ML-2	Slate Digital	2	$150	$300
Medium Diaphragm Condenser Microphone	AT2020	Audio Technica	2	$100	$200
Large Diaphragm Condenser Microphone	AT2035	Audio Technica	2	$150	$300
Measurement Microphone	EMM-6	Dayton Audio	1	$80	$80
Shock Mounts	GFW-MIC-4248	Gator Frameworks	2	$25	$50
Tall Mic Stand	MS7701B	On-Stage	6	$40	$240
Short Mic Stand	GWF-MIC-0821	Gator Frameworks	2	$50	$100
Desk Mic Stand	DS7200B	On-Stage	5	$20	$100
Main Audio Interface	Scarlett 18i20	Focusrite	1	$550	$550
Secondary Audio Interface	Scarlett 2i2	Focusrite	1	$180	$180
Studio Monitors	JBL 305P MkII + LSR310S Subwoofer	JBL	1	$570	$570
Studio Monitor Stands	GFWSPKSTMNDSK	Gator Frameworks	1	$100	$100
				Total:	**$3,370**

As shown in the table, the cost of all the equipment came out to **$3,370**. The additional cost for the cables was estimated at **$370**.

The software requirements were fairly straightforward. There needed to be at least one digital audio workstation (DAW) installed on the audio computer to perform the mixing and mastering. This paper suggested Ableton Live, as it is specifically designed to work in a live setting with minimal interruptions. Easy-to-access tools and automation would be convenient when time is limited. A standard Ableton Live license costs **$450**. Ableton is not without its faults, and a second DAW would be useful for times when live audio is not a priority. Pro Tools is the industry standard among professionals, which made it the ideal choice. A studio license for Pro Tools costs **$100 per year** with the education discount.

Another software called Room EQ Wizard (REW) should be installed on the audio computer in order to measure the frequency response of the environment. REW is completely free, and audio technicians can use its frequency response measurements to determine where absorptive paneling and loudspeakers should be placed. This paper did not provide any information on how to use REW. For in-depth tutorials, see their website.

Drivers for the audio interfaces will also need to be installed on the computers. These will either come free with the purchase of the interfaces, or they will be found on the company's website.

In total, the cost of materials for the acoustic treatment, audio equipment, audio cables, and software came out to approximately **$9,610.**

The final section of the paper served as a short guide for installing all the audio equipment. Figure B is a cable diagram showing how everything would be wired together. Note that a hole must be drilled between the two rooms to run the snake cable through. To keep the hole soundproof, it should be sealed up with caulk afterward.

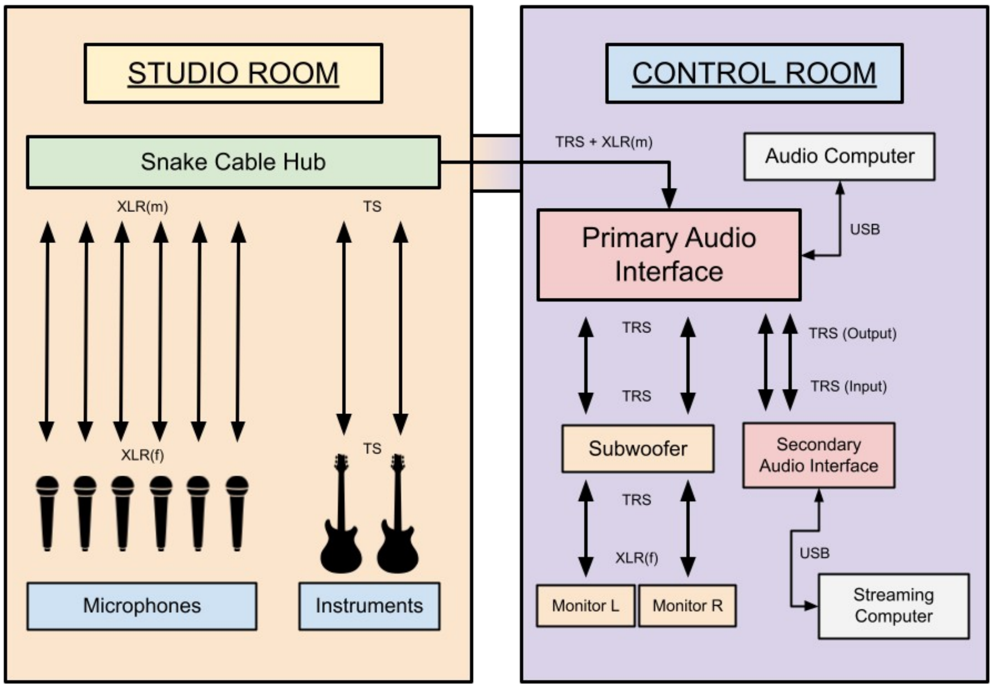

Figure B: Audio Equipment Wiring Guide

1. Background

Before designing a live streaming space we need to introduce the relevant concepts. These include acoustical properties and recording equipment to ensure the sound in the room will be high in quality both in-person and over the internet.

1.1 History of Live Streaming

Although the concept of a “live stream” may seem hip and modern, live streaming existed as early as 1890, when the French Théâtrophone service used telephone lines to broadcast live performances of theater and opera to its paying subscribers. The service continued broadcasting music and intermittent news updates until 1932, when the mainstream adoption of the radio and phonograph rendered it obsolete. A similar service called Muzak operated in America from 1934 to 1981, which streamed music into elevators and workplaces. Another service, the Telephone Music Service, allowed patrons of bars in Pittsburgh to send music requests to an operator through special jukebox telephones from 1929 to 1997. The revenue from the jukeboxes was shared between the music service and the tavern owners. (Wikipedia Contributors, 2022)

Throughout the later 20th century, the invention of computer networks allowed users to communicate across the globe. Powerful home computers, advanced internet infrastructure, and novel data compression techniques eventually supported the bandwidths required for live transmission and playback of video and audio. The first band to ever perform live over the internet was Severe Tire Damage on June 24, 1993. Their performance in Palo Alto, California was watchable as far away as Australia. Although the quality of the live stream was poor by today's standards (only 152x76 pixels), this was an impressive feat at the time. (Wikipedia Contributors, 2022)

Over the next decade, the internet became faster and increasingly commercialized. Live streamed internet broadcasts became increasingly common. RealAudio Player was one of the first media player software capable of streaming over the internet, and it was used by ESPN to stream the first live internet broadcast of an MLB game in 1995. In 1999 Bill Clinton participated in the first presidential webcast. (Wikipedia Contributors, 2022)

In the modern day, live streaming has become ubiquitous. Anyone with a smartphone and a decent internet connection can easily start a live stream in a matter of seconds. Platforms such as Youtube, Twitch, Facebook, and Vimeo provide convenient platforms for users seeking to distribute their content to hundreds, thousands, or even millions of viewers. For some people, live streaming has become a full-time career. Revenue can come from a number of sources including advertisers, merchandise, affiliate programs, and donations directly from viewers (Bybyk, 2022).

From musical performances to esports, live streaming is an industry on the rise. Having such a studio on the Worcester Polytechnic Institute campus will give students the opportunity to gain experience working in a professional live streaming setting. The space will also allow other members of the community to create high-quality audiovisual recordings to share their performances and/or ideas with a wider audience over the internet.

1.2 Room Acoustics

The first step in planning a live streaming studio was making sure the sound quality is the best it can be. A poorly planned room can ruin the quality of even the best musical performances.

This section describes the basic principles of acoustic treatment and design, which was later applied to the design of the live streaming studio at WPI. For simplicity, assume all of the information in this section is cited from Everest and Pohlmann (2021), except for the parts that are specifically cited from elsewhere. We assume the reader already has a basic concept of sounds as waves with frequencies and amplitudes.

Reverberation is a familiar and immediately noticeable acoustic property of any room. A reverberant room may add a bright, spacious quality to music, but it will also blur notes and syllables together, making speech difficult to understand. Inversely, if a room is not reverberant enough it may sound dry or "dead". Reverberation can be quantified using what is known as the *RT_{60} value* (also called the "reverberation time"), defined as the amount of time for sound energy in a room to decay by 60 dB. A room covered in acoustically *reflective* surfaces such as hardwood tables or ceramic tiling will have a *higher* RT_{60} and will therefore sound more spacious compared to a room with a lot of *absorptive* surfaces such as carpeting and drapery.

To further complicate things, different frequencies of sound decay at different rates – some surfaces are excellent at absorbing lower frequencies while others only absorb higher frequencies. All surfaces can be characterized by an *absorption coefficient* (ɑ), which ranges from 0 to 1 and describes how much sound energy the surface absorbs per unit of area. An open window, for example, will have an absorption coefficient of 1.0 because no sound is reflected back. Retailers of acoustic paneling usually provide data sheets listing their material's absorption coefficients at a few standard frequency bands. Some brands even advertise coefficients greater than 1.0, but this is due to a technicality in the way the values are measured. The coefficients should be rounded down to 1.0 before making calculations.

The Sabine Equation, devised in the 1890s by Wallace Clement Sabine, is a widely used equation for estimating the value of RT_{60} given the absorptivity of a room at a particular frequency:

$$RT_{60} = \frac{0.049\,V}{A} \tag{1.1}$$

Where V is the volume of the room in cubic feet, and A is the total absorption of the room measured in units of sabins. The total absorption A can be calculated by adding together the surface areas of everything in the room multiplied by their absorption coefficients. 1 square foot of material with $\alpha=1.0$ will contribute 1 sabin of absorption. For reference, a typical college student standing in a room will add about 5.0 sabins for frequencies above 1000 Hz.

Recording spaces will often have absorptive paneling mounted on the walls to absorb middle and high-range frequencies. Spacing absorbing panels apart from each other and mounting them slightly away from the wall can make them more effective by increasing their exposed surface area. For peak efficiency, the absorbers should be placed ¼ wavelength away from the wall, which is where air particles will have the greatest velocity (though a 2-inch gap is usually wide enough). Lower frequencies are reduced using special absorbers called "porous bass traps", which often look like wedges placed along the corners of the room, which is where low frequency energy tends to gather (Sweetwater, 2022). Additionally, mounting the panels in a way that makes them easy to rearrange and remove, such as on wooden racks, can allow users of the room to tailor the absorptivity of the room to their liking, creating a more versatile recording space.

The next part of acoustic design to consider is *standing waves*. When a sound wave hits one or more walls, it is reflected in a new direction. The reflected sound is then "stacked" on top of the original wave. If the two waves are out of phase, they cancel out. However, if they are in phase they combine into a more powerful wave in a process called *constructive interference*. Certain frequencies of sound in a room will reflect and combine constructively multiple times, forming “standing waves” that appear to oscillate in place. Figure 1.1 shows the first three standing wave modes in one dimension. The dotted lines are included to illustrate how the waves oscillate in time. Note how certain points in this figure oscillate at a maximum amplitude (called “antinodes”), while others do not oscillate at all (called “nodes”). In the context of acoustics, a person standing at an antinode would be able to hear the standing wave frequency much better than another person standing at a node.

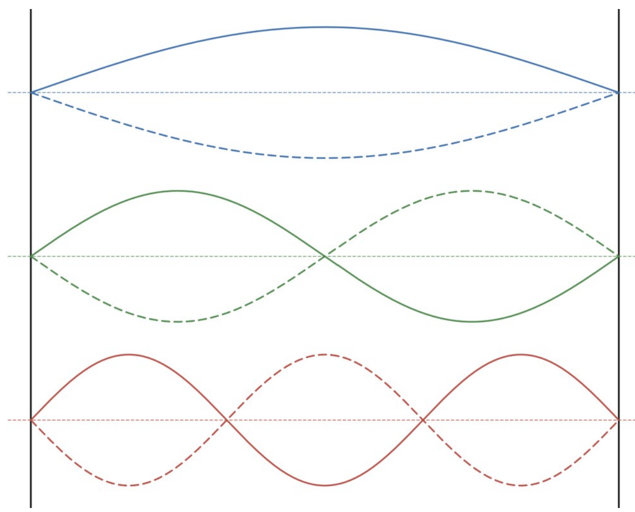

Figure 1.1: First Three One-Dimensional Standing Wave Modes

While Figure 1.1 depicts one-dimensional waves, standing waves also exist in three-dimensional spaces, and they are a huge problem in acoustics. Figure 1.2 shows several overlapping standing waves in a small mixing studio. The colored lines represent the intensity of the frequencies at each location in the room. The listener will hear different frequencies more clearly depending on where they are seated.

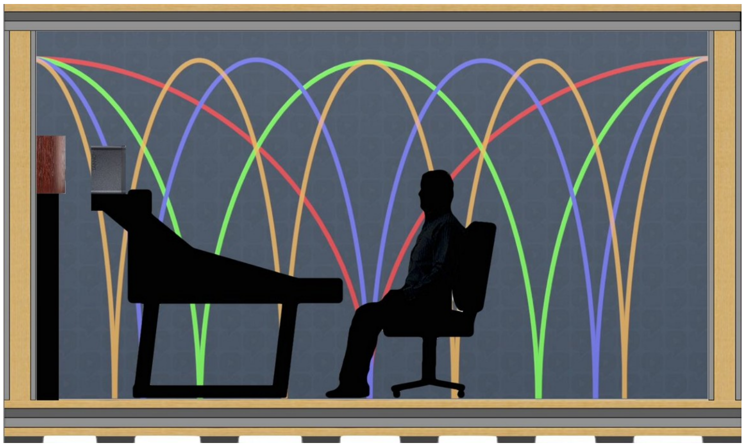

Figure 1.2: Standing Waves in a Studio Setting. The lines represent the intensities of the standing wave frequencies at each location. Image Source: Albano (n.d.)

The specific frequencies at which standing waves occur are called *room modes*, and they are uniquely determined by the geometry of the room. For example, a large auditorium will have modes at lower frequencies than a small recording studio. In a six-sided box-shaped room there are three kinds of modes: axial, tangential, and oblique. Axial modes are the strongest modes and result from sound reflecting off of a pair of opposite-facing walls. Tangential modes are the result of sound reflecting off of two pairs of opposite walls and are typically 3 dB weaker than axial modes. Oblique modes are the weakest (6 dB below axial modes) and result from reflections on all three pairs of walls. These three types are illustrated in Figure 1.3.

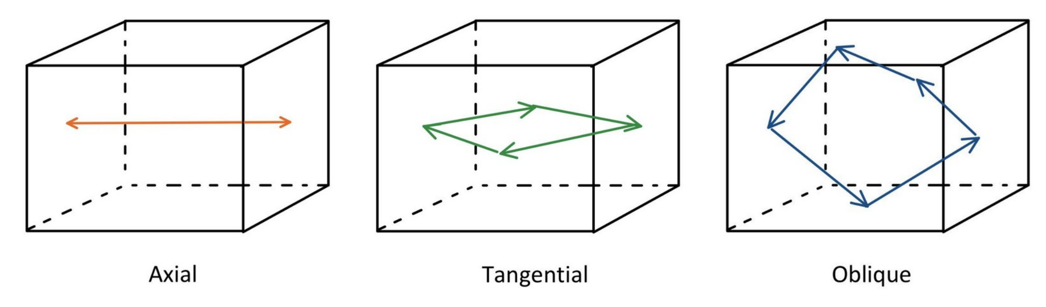

Figure 1.3: Three Types of Room Modes

The mode frequencies for a box-shaped room can be easily calculated using the following formula:

$$Frequency = \frac{c}{2}\sqrt{\frac{p^2}{L^2} + \frac{q^2}{W^2} + \frac{r^2}{H^2}} \tag{1.2}$$

Where c is the speed of sound (1,130 ft/sec), L W H are the room's dimensions, and p q r are all integers greater than or equal to 0, representing the mode number. A room can have thousands of modes, but a computer program can very quickly compute all possible room modes within a given frequency band. It is also worth mentioning that designing a room to have a non-cubic shape will not remove standing waves. Instead, it will shift them around and make them more complicated to predict.

At high frequencies, there are so many modes that they blend together and become unnoticeable to a listener. However, at lower frequencies, room modes are spaced out and can create audible defects in the sound. To reduce the severity of this problem, specialized absorbers called *resonating bass traps* can be tuned to selectively absorb specific frequencies (Sweetwater, 2022). Resonating bass traps have been used by humans for a very long time; Medieval churches in Scandinavia have been found with hollow pots embedded in the walls that are thought to have served as early *Helmholtz resonators*. A Helmholtz resonator (HR) is a simple resonating bass trap that consists of an air cavity with one or more openings. The sizes of the openings and the volume of the cavity determine the frequency that the resonator will absorb, similar to how a glass bottle will produce a characteristic sound when air is blown across the top. A fibrous material stuffed inside the cavity creates air resistance that converts the vibrational sound energy into heat energy, dampening the sound. Another type of resonating absorber is the *perforated panel absorber* (PPA). These are panels covered in small holes that are mounted onto a wall. The air between the panel and the wall form what are essentially many small Helmholtz resonators. Fibrous material can sometimes be inserted between the panel and the wall to broaden the absorption frequency range. PPAs are very easy to construct, but they usually do not go much lower than 90 Hz (Foley, 2013). The type of a resonator can be whatever is most convenient in terms of construction or aesthetics, and different types of resonators will be better suited for different applications.

The next acoustic property worth considering is *diffusion*. Diffusion is the process of scattering reflected sound so that it is evenly spread around the room. In an ideal, perfectly diffuse sound field, reverberation time and sound intensity would be the same no matter where a listener is positioned. Additionally, a diffuse sound field will have a perfectly exponential decay, free from irregularities and comb-filter effects, which occur when sound interferes with its own reflection. Although this “perfect diffusion” is not possible in the real world, incorporating diffusive elements into a room’s design can still enhance its acoustics. This is done using acoustically reflective surfaces with geometries that scatter sound evenly in all directions. Something as simple as a wooden cylinder would make an excellent diffuser, but more sophisticated solutions also exist. Manfred R. Shroeder developed a special kind of diffusive element called a quadratic residue diffuser (QRD). These diffusers have grooves carved into them at various depths based on a mathematical sequence to optimize scattering across a wide range of frequencies. An example of a one-dimensional QRD that diffuses sound in the horizontal direction is shown in Figure 1.3. The same type of sequences can be used in two dimensions to scatter sound vertically as well as horizontally. A well-designed studio room should have diffusive elements on all pairs of opposing walls to reduce comb-filtering.

Figure 1.4: Example of a Commercial QRD. The diffuser pictured here will only scatter sound in the horizontal direction. Image Source: Acoustic Fields (n.d.).

Finally, acoustic treatment must account for *external* sources of noise. External noise might originate from ventilation systems, a busy road, or even the footsteps of people walking around upstairs. If someone sneezes in another room, it should not be audible in a recording. Realistically, external noise can never be prevented entirely, but with proper planning and treatment it can be significantly reduced. A brief exploration of potential soundproofing options is included in the Methodology section, but this is a complex topic that is best left for more thorough analysis.

1.3 Recording Equipment

Microphones are devices necessary to record and stream audio. Although many cameras come with built-in microphones, these cannot be repositioned without moving the entire camera as well. For top-quality audio and recording flexibility, a studio needs several dedicated microphones. There are so many kinds of microphones that choosing the right ones can be a daunting task. Thankfully, most modern microphones fall into one of two categories: *dynamic* and *condenser* microphones. Dynamic microphones operate by using sound energy to vibrate a conductive coil. As the coil vibrates, it passes near magnets that produce a detectable electric signal. Dynamic microphones tend to be very durable and can register louder and lower-frequency sounds than condenser microphones, making them well-suited for instruments like drums. Condenser microphones contain a diaphragm made out of conductive plates. When sound hits the diaphragm, the capacitance between the plates changes in a way that can be measured by an electric circuit and converted into a signal. Unlike diaphragm mics, condensers must be supplied with 48 volts of "phantom power" in order to function. Fortunately, almost all modern audio interfaces support this. The diaphragms of condenser mics come in various sizes, where smaller diaphragms are better for higher frequencies and larger diaphragms are better for lower frequencies. These microphones are more fragile than dynamic mics, but they can also register quieter sounds, making them well-suited for recording speech (Wreglesworth, 2022). Figure 1.4 shows simplified diagrams of both a dynamic and condenser microphone to illustrate how they function.

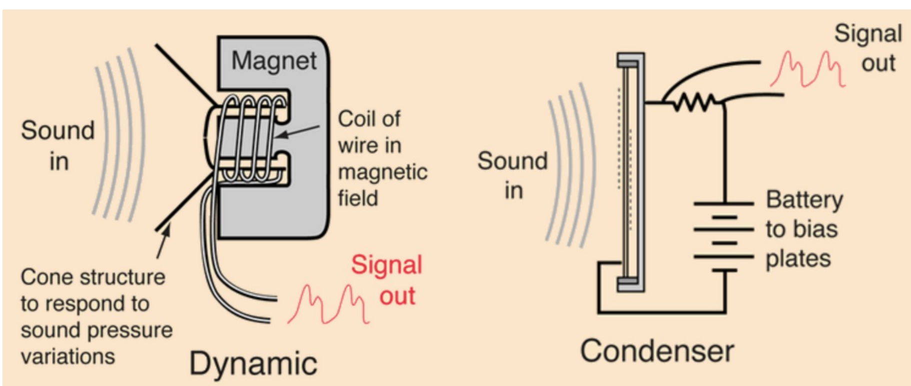

Figure 1.5: Diagrams of Dynamic and Condenser Microphones. Image Source: Nave (n.d.).

There are other properties that affect how a microphone sounds. Some microphones are designed to record sound coming from all directions ("omnidirectional"), while others are made to only listen in a narrow direction ("cardioid") (Wreglesworth, 2022). Some microphones can even toggle between multiple polar patterns with an onboard switch. Different microphones have their own strengths and weaknesses, so it would be useful to have several kinds of microphones in a studio for an audio technician to select from.

It may at times be necessary to connect many microphones into one computer simultaneously. An *audio interface* is a piece of hardware that allows many microphones and speakers to plug into a single computer at once. The device “interfaces” between the digital signals used by the computer and the analog signals used by the microphones and speakers in a way that preserves as much sound quality as possible (E-Home Recording Studio, 2022). Interfaces usually plug into a computer through a single USB port, but there are also products that use Thunderbolt or Ethernet connections. Knobs on some interfaces allow audio technicians to easily adjust the signal gain of each input and output, and preamplifiers built into an interface can boost incoming microphone signals before processing, resulting in a cleaner sound. Audio interfaces also supply the “phantom power” that condenser microphones need to function. There are many different audio interfaces available, so it is important to know the number and type of connections that are needed. Smaller, less-expensive interfaces might only provide two analog inputs and outputs, while the larger rack-mounted interfaces can support 20 or more. (Sweetwater, n.d.)

An audio interface should not be confused with an audio *mixer*, which combines many inputs into a single stereo input track. If someone wanted to isolate or remove an instrument after recording using an audio mixer, they would be unable to do so. Audio interfaces do not combine signals. Each input goes into its own track in whatever software is used by the computer, allowing for much more flexibility in a studio environment. Some mixers do support multichannel recording as well, but a regular interface is the simpler and more cost-effective option for our purposes (Sweetwater, 2020).

Finally, a studio needs at least one pair of *studio monitors*. Similar to home speakers, studio monitors are used for audio playback. The difference is that home speakers will often amplify certain frequency ranges to enhance the listening experience, which is not favorable in a studio environment. Studio monitors are designed to render sounds as accurately as possible so that any imperfections or imbalances in the sound will not go unnoticed (Neumann, n.d.). Some modern studio monitors also provide additional features such as the ability to adjust the speaker’s frequency response to better suit its acoustic environment.

2. Methodology

Now that the relevant concepts have been introduced, we discuss how they can be applied to design the room and purchase the best equipment. The approximate budget for the entire live stream space was $75,000, which included non-acoustic equipment such as computers and cameras. The room acoustics take up as little of this budget as possible without sacrificing too much in quality.

After everything was considered, the total cost of materials for the acoustic treatment, audio equipment, audio cables, and mixing software came out to approximately $9,610. This excluded some additional costs for acoustic treatment, such as soundproof doors, resonating absorbers, and installation costs.

2.1 Acoustic Treatment Requirements and Calculations

In order to properly plan the acoustic treatment of a room, we needed to know the exact dimensions of the rooms. Each room was 10x15x10 feet, and they shared a long wall. We refer to the room in which recording takes place as the "studio room", and the other room, which contains the streaming and mixing computers, as the "control room".

We began by finding the amount of absorptive material the studio room would need for optimal reverberation time. Different use cases warrant different reverberation times – music tends to benefit from slightly more reverberation than speech does. Everest and Pohlmann (2021) suggest a music and speech "compromise region" for rooms with a volume of 1,500 ft^3 between approximately 0.3 and 0.6 seconds (p. 457). Based on this, 0.45 seconds was a reasonable target. The Sabine equation was used to solve for total absorption. Ignoring the absorption of the walls and floor, we found that we need 160 sabins of absorption. Commercial acoustic panels often have absorptive coefficients of at least 1.0 for high-frequency ranges, so that translates to roughly 160 ft^2 of acoustic paneling. The word “roughly” is used because certain factors will increase the absorption. For example, mounting the panels so that they are spaced a few inches from the wall increases their effective absorptive area. Also, this calculation did not account for any furniture, carpeting, or people in the room, which would increase the absorption further. This is why removable paneling is important. An audio technician can measure the room absorption using specialized software (discussed in the later sections) and then remove or add absorptive elements as they see fit. A removable floor rug may be preferable to a permanent carpet for the same reason. In the case of recorded audio, reverb could also be artificially introduced with software. The absorptive panels should be spread around so there is fairly even absorption around the room. Porous bass traps installed in the corners provide broad low-frequency absorption.

The control room followed a different design because it serves a different purpose. The goal of the control room was to optimize audio accuracy in just a single listening position. A common way of achieving this is the “live-end dead-end” (LEDE) method, which is suggested by Everest and Pohlmann (2021). For LEDE, all of the absorptive paneling is placed on the side of the room with the speakers and listener (called the “dead-end”). The opposite “live” end of the room is treated with reflective diffusers. The goal of the LEDE design is to prevent reflected sound from interfering with what the listener hears from the speakers without deadening the room too much. The diagram that Everest and Pohlmann (2021) use to illustrate this design is included below in Figure 2.1. There exist more complex designs that further improve the listening quality, such as creating a reflection-free zone using splayed walls, but those are left for future research.

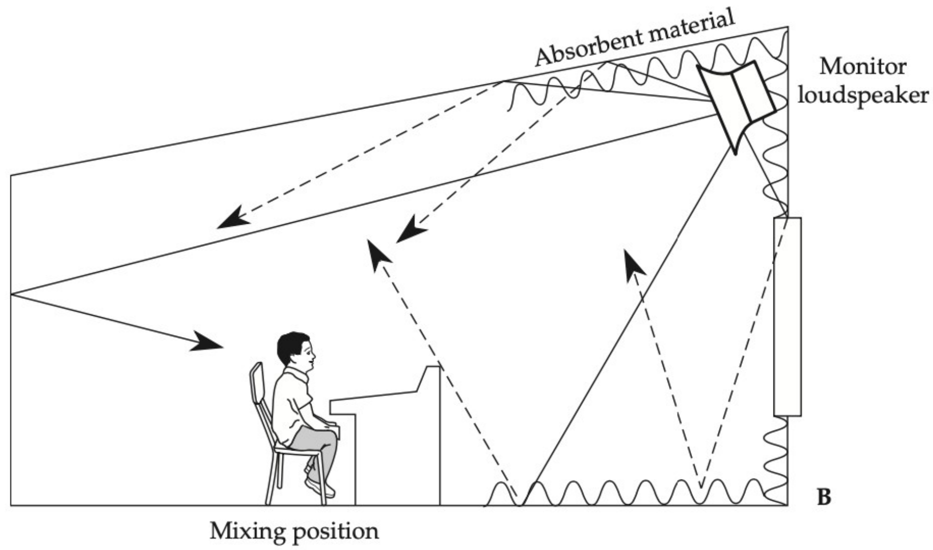

Figure 2.1: LEDE Control Room Conceptual Diagram. The side of the room with the loudspeaker is covered in absorptive material (the "dead" end), while the other is left reflective (the "live" end). Image Source: Everest and Pohlmann (2021).

Incorporating diffusion into the studio room was a simple process. For the case of small studio rooms, Everest and Pohlmann (2021) state that "In practice, it would be difficult to provide too much diffusion" (p. 458). Going off of this advice, all the remaining wall space was used for diffusive paneling. It was also important that all pairs of opposing walls had at least one side with diffusive elements to prevent comb-filter effects, which occur when sound reflections off a pair of walls interfere with each other. These diffusive elements could be one or two dimensional quadratic residue diffusers or even simply large half-cylinders made out of wood.

Another issue to tackle was room modes. With such a small space, room modes pose a great threat to acoustic quality, so resonating bass traps are necessary to selectively absorb those specific low-frequency tones. The goal was to simply match the resonator frequencies with the most problematic room modes. This section proposes the frequencies that should be controlled in order to optimize the frequency response of the room. To analyze and compare all the possible resonator designs would be too time-consuming, and unfortunately had to be left for future research.

Using Equation 1.2, a simple computer program was written to compute and plot the axial, tangential, and oblique modes of the rooms. The resulting plot is displayed in Figure 2.2. The complete Python code is provided in Appendix A.

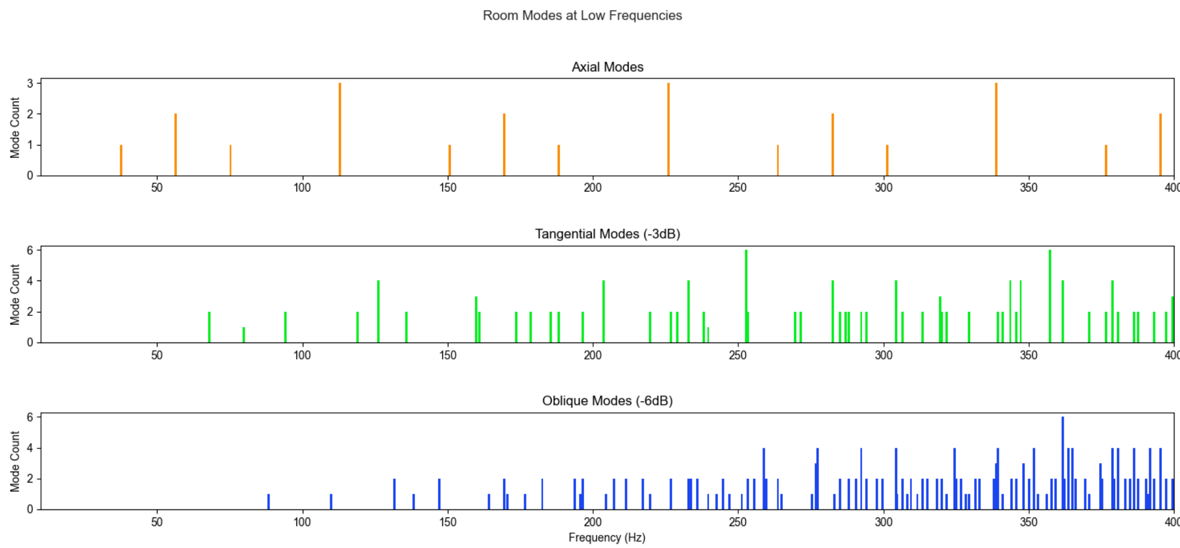

Figure 2.2: Room Modes Plot. Axial, tangential, and oblique room mode frequencies for a 10x15x10 ft room. Frequencies are rounded to one decimal place before counting degeneracy.

As stated in the Introduction, the modes should be evenly spaced with as few degenerate peaks as possible. It is worth mentioning that the room modes of the studio are *far from ideal*, and a room with these dimensions would not be a great candidate for a recording space. Anyway, axial modes should be dealt with first, as they are the most intense. The peaks in the first plot of Figure 2.2 occur at all multiples of 56.5 Hz, so these should be reduced first. The peaks at multiples of 113 Hz are threefold degenerate, so those frequencies may require more resonators to control. If space allows, there are also some tangential modes that could be controlled such as the peaks at 126 Hz and 252 Hz.

The studio room would also need some form of external soundproofing to reduce unwanted background noise. Everest and Pohlmann (2021) include several chapters detailing this very subject. A soundproofing measure one might expect for any recording studio would be soundproof doors designed to create airtight seals when closed. Noisy foot traffic can be reduced by installing floor padding upstairs, in neighboring rooms, and in hallways. Ventilation systems can be quieted by installing turbulence-reducing air ducts and/or isolating the HVAC unit from the rest of the building's structure using spring mounts. Other measures for noise reduction include altering the wall's material and support structure, but this is far outside the budget and scope of this paper.

2.2 Acoustic Treatment Plan

An example design for acoustic treatment is provided in Figure 2.3. For a fun 3D rendering of the design, see Appendix B.

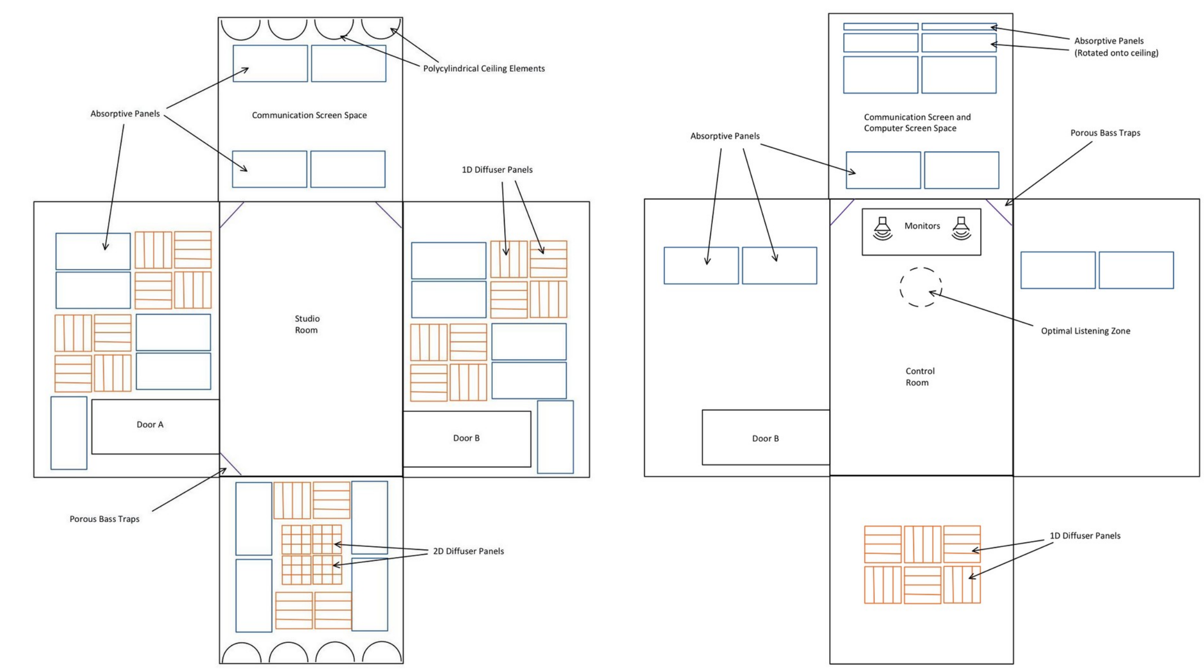

Figure 2.3: Acoustic Designs for the Studio and Control Rooms.

The plan included porous absorbing panels, porous bass traps, and two kinds of diffusing elements. Additional wall space was provided in each room for a television to facilitate communication between the two rooms when the doors are closed. The studio room plan contained the recommended 160 square feet of absorptive paneling divided into 2’x4’ panels of 2” thickness. The porous bass traps provide some high-frequency absorption due to their material, so they were counted towards the surface area as well. As mentioned in the Introduction, the wall panels should be mounted with at least a 2” gap from the wall to improve efficiency. The ceiling of the studio room was covered with poly cylindrical diffusive surfaces to prevent comb filtering with the floor. These ceiling cylinders were included as an example; differently shaped ceiling panels can provide the same effect. The control room followed the LEDE design with absorptive panels on one side and diffusing panels on the other. The resonating bass traps were not included in the diagram, as their exact dimensions and mounting style are unknown. In any case, they should be located near the room’s corners for maximum effectiveness.

From this plan, cost estimation was straightforward. We counted the number of acoustic elements and multiplied them by their unit prices. The cost of the design is summarized in Table 2.1, and the reasoning behind the vendors is described in the following section.

Table 2.1: Cost Breakdown of Acoustic Treatment

Element	Vendor	Count	Unit Price	Total Price
2ft x 4 ft Absorptive Panels (2" thick)	ATS Acoustics	30	$72	$2,160
1ft x 4ft Porous Bass Traps	Acoustics America	6	$90	$540
2ft x 2ft x 3" 1D Quadratic Residue Diffuser	BXI	26	$70	$1,820
17.5"x17.5"x5.5" 2D Quadratic Residue Diffuser	GIK Acoustics	4	$200	$800
			Total:	**$5,320**

The decision between product vendors was decided based on a balance of effectiveness and cost. For the absorptive panels and bass traps, the cost and performance were nearly identical between various vendors, so the decision was based on more arbitrary features such as fire-rating and color options. Cheap acoustic foam was not used because it tends to have poor mid-frequency absorption (which leads to an unbalanced frequency response), and it is usually not fire-rated. In other words, you get what you pay for. ATS Acoustics panels were chosen because of their option to print custom designs on the panels. With these, the studio could be adorned with a decorative pattern, a wall-spanning decal, the WPI logo, or something else.

For the diffuser panels, the decision of the vendor was based primarily on the lowest cost. The four 2D QRDs were chosen mainly for aesthetic reasons, and they can be substituted for 1D QRDs if the budget demands it. Similar to the cheap absorptive foam, there exist significantly cheaper diffusive tiles made out of plastic. While inexpensive, these panels tend to be quite small, which limits their ability to effectively diffuse mid and low frequency sounds (*Best Types of Sound Diffusers: Should You Buy or DIY Them?*, 2020). It is worth mentioning that the diffusive panels are made out of wood, so they could likely be constructed by WPI students for less cost than their retail prices. For this option, *Soundsplash* by HX Audio Labs is a convenient free software meant for simulating the effectiveness of QRD designs based on channel depths (Sound Splash, n.d.).

The ceiling elements and the resonating bass traps were not included in the price estimation as they would likely need to be custom-built for this room, and there was not enough time to compare vendors. As with the diffusers, these could also be constructed by WPI students out of wood or a similar material once their design is finalized.

Treatment for external sounds would also add to the budget. A soundproof barrier is only as effective as its weakest link. For most acoustic spaces the weakest parts are usually the doors (Everest and Pohlmann, 2021). Special airtight doors would need to be purchased to prevent sound from leaking through. According to Walker (2021), such soundproof doors can cost anywhere from $100 to $400 for a solid wood door, depending on the material and size. A tube of inexpensive soundproofing caulk should be used to seal any air holes in the room's walls, such as those used for microphone and video cables. Additional costs may include sound treatment for ventilation systems and floor padding to reduce the sounds of foot traffic in nearby rooms. Analysis of these techniques is left for future research.

2.3 Audio Equipment Product List

There are many required audio devices for a recording space, such as microphones, stands, an audio interface, and studio monitors. There are also some optional accessories such as pop filters and shock mounts. The cost breakdown of all the equipment is summarized in Table 2.2. The following paragraphs detail the reasoning behind each product. As with the room design, this list was just one of many possible studio designs, and higher-end alternative products can be substituted if the budget allows. The total cost of audio equipment (excluding cables) came out to $3,370.

Table 2.2: Cost Breakdown of Audio Equipment

Item	Product Name	Vendor	Count	Unit Price	Total Price
Dynamic Microphone 1	SM57	Shure	4	$100	$400
Dynamic Microphone 2	SM58	Shure	2	$100	$200
Small Diaphragm Condenser Microphone	VMS ML-2	Slate Digital	2	$150	$300
Medium Diaphragm Condenser Microphone	AT2020	Audio Technica	2	$100	$200
Large Diaphragm Condenser Microphone	AT2035	Audio Technica	2	$150	$300
Measurement Microphone	EMM-6	Dayton Audio	1	$80	$80
Shock Mounts	GFW-MIC-4248	Gator Frameworks	2	$25	$50
Tall Mic Stand	MS7701B	On-Stage	6	$40	$240
Short Mic Stand	GWF-MIC-0821	Gator Frameworks	2	$50	$100
Desk Mic Stand	DS7200B	On-Stage	5	$20	$100
Main Audio Interface	Scarlett 18i20	Focusrite	1	$550	$550
Secondary Audio Interface	Scarlett 2i2	Focusrite	1	$180	$180
Studio Monitors	JBL 305P MkII + LSR310S Subwoofer	JBL	1	$570	$570
Studio Monitor Stands	GFWSPKSTMNDSK	Gator Frameworks	1	$100	$100
				Total:	**$3,370**

There needed to be a selection of different microphones to choose from for different use cases – live streaming a noisy brass band will warrant different microphones than a quiet strings quartet. At the same time, there needed to be enough microphones of each type to allow multiple voices and/or instruments to be recorded at once. For the purposes of the WPI studio, two edge cases were considered: a small band of at most six instruments, and a podcast with five voices. Given the small area of the room, these two cases should reasonably encompass all possible situations. For a band with loud brass instruments, dynamic microphones are best suited for the job. For upper-range stringed instruments and human voices, small-diaphragm condensers would work best. For lower-range strings and voices, medium or large-diaphragm condensers are better. Therefore a good selection of microphones would include six dynamic microphones and six condenser microphones, with an even split of large, medium, and small diaphragms. There also needed to be one additional measurement microphone for fine-tuning the room‘s acoustic treatment.

Microphones can be expensive, so price was an important factor to consider when shopping. For dynamic microphones, the Shure SM-57 and SM-58 are two of the most widely recognized microphones available. Internally, the two microphones are almost identical. The key difference is that the SM-58 has a large built-in pop filter to reduce plosives (puffs of air produced when pronouncing certain syllables), making it more suitable for vocals (Henshall, n.d.). Figure 2.4 shows the visual difference between the two microphones. Due to their affordability and popularity, we included two SM-58s and four SM-57s.

Figure 2.4: SM-58 and SM-57 Microphones. The SM-58 (left) has a bulky mesh pop filter for enhancing vocals. Image Source: *Shure SM-57 / 58 Microphone Rental*. (n.d.).

The six condenser microphones were divided into three groups of diaphragm sizes: small, medium, and large. For variety, two of each kind were included in the list, each chosen for their affordability and good user reviews. For small diaphragms, the VMS ML-2 comes with a switch that allows it to pick up louder sounds without peaking. For medium and large diaphragms, we suggest the AT2020 and AT2035, respectively. Both are popular and affordable condenser microphones, and they appear visually nice on camera. The AT2035 also comes with its own shock mount. When combined with the SM57s and SM58s, this selection of twelve microphones should be sufficient for most use cases.

Finally, there is the measurement microphone. This is a special type of microphone used to measure the frequency response of the room. The website for Room EQ Wizard (REW), a free software for measuring frequency responses, suggested using the Dayton Audio EMM-6 measurement microphone, which costs $80 (Mulcahy, 2022). A later section will discuss how this microphone is used.

On top of all of this, we needed microphone stands and clips (the part that connects the microphone to the stand). Many microphones come with their own clips, and basic microphone clips tend to be quite cheap (less than $10 each), so they were excluded from the cost breakdown. Shock mounts are special clips that absorb low frequency vibrations transferred up from the floor. Gator Frameworks sells cheap shock mounts for $30, so two are included for the studio. To record a band of six standing players, we needed six tall microphone stands. The MS7701B is one such mic stand that can extend from three to five feet tall. There may also be times when a floor-level microphone mount is necessary, such as for a cello or a bass drum. For this, two GFW-MIC-0821 s are included, which extend one to two feet. For a five-person podcast, five desk-mounted stands, such as the DS7200B, would be useful. These three brands of microphone stands were chosen based on their affordable prices and positive buyer reviews. As always, other similar products could easily be substituted.

The audio interface used to connect all the microphones needed to have enough ports to support at least six microphones simultaneously. Many audio interfaces come with more than six inputs, but these are often “line” inputs, which are meant for electronic instruments and are not necessarily suitable for microphone signals (Neumann, n.d.). To be more specific, the interface must have six *microphone* inputs in the form of “XLR” connectors, which is the type of 3-pin input connector used by most microphones. We recommended the *Focusrite Scarlett 18i20 3rd Gen*, which costs about $550. It can be mounted on a tabletop or a server rack, and it has 8 XLR inputs for microphones. It plugs into the computer with a USB 2.0 connection, but also requires a source of external power from a wall socket. The reason this interface was recommended over other brands is that Focusrite is a widely-used brand of audio interfaces. Someone using the studio for the first time may already be familiar with the brand, and software support for common operating systems (namely Windows and macOS) is unlikely to be dropped in the near future. The 18i20 also comes with many extra line inputs and outputs if the need for those ever arises. Figure 2.5 shows the front and back sides of the 18i20. Note the 8 XLR connectors (two on the front, six on the back).

Figure 2.5: Focusrite Scarlett 18i20. Front and rear views show the number and types of available inputs. Image Source: Focusrite (n.d.).

Professor Bianchi suggested that there should be a computer dedicated solely to audio editing. The audio computer would combine the microphone inputs into a single stereo track, while the streaming computer would combine that stereo mix with the video from the cameras. The purpose of this would be to offload some processing from the streaming computer to reduce potential stuttering. This introduced the challenge of finding a way to send the stereo audio track from the audio computer to the streaming computer without losing sound quality. While a few software solutions existed, they introduced additional network latency and may be subject to lossy compression, which would reduce quality. The cleanest solution was to connect a smaller secondary audio interface to the streaming computer. The output of the main 18i20 interface would be connected directly to the input of the secondary one. For a visual diagram of this setup, see Figure 3.1 in the next section. This comes with the added benefit of being able to connect the studio monitors to the streaming computer instead of the audio computer when desired, as the secondary interface will also have its own output ports. To keep the brands consistent, a good secondary interface would be the Focusrite Scarlett 2i2. It is essentially a tiny version of the

18i20 with only two inputs and two outputs (for the left and right audio channels). The 2i2 is small enough that it does not require an external power supply.

The control room should have high-quality monitors for stereo playback. Professional-grade monitors are designed to playback audio as accurately as possible. Only two studio monitors are needed, as the LEDE control room design does not easily accommodate surround-sound systems. The control room would also benefit from a subwoofer for better playback at ultra-low frequencies. At Professor Bianchi's suggestion, we chose to use a pair of JBL 305P MkII 5-inch studio monitors. These are moderately high-end speakers that cover most of the audio spectrum (43Hz to 23kHz). The tweeter on the top of the speakers has a horn-shaped inset that is advertised to widen the optimal listening zone. Switches on the back adjust the high and low frequency output to better suit the room they are used in. The monitors would be purchased alongside the LSR310S 10-inch subwoofer.

The two monitors should also have stands/mounts to keep them acoustically isolated from the floor. This has the added benefit of raising the speakers off of the table to reduce unwanted reflected sound. Gator Frameworks sells a pair of desktop monitor stands that cost $100 in total.

Fortunately, a lot of these audio devices are modular. If the need arises in the future, more microphones or a larger audio interface may be incorporated without drastically altering the rest of the audio system.

2.4 Cables

Audio cables tie all the hardware together. Length, quality, and connector types were of key concern here. The microphones all use XLR-type connectors, but certain electric instruments require quarter-inch TRS or TS connectors as well. For convenience, a "snake cable" should run between the two rooms. A snake cable is a thick cable that bundles many smaller cables together into a manageable tube. Having one or two snake cable extenders permanently installed between the rooms would allow microphones/instruments in the studio room to be plugged into the audio interface in the control room. The Hosa Little Bro' 6x2 cable contains six XLR extenders and two TRS extenders and costs $76. This would support six microphones as well as two TRS or TS ported instruments. In the studio room, there needs to be extender cables to attach the microphones and instruments to the end of the snake. There is a huge variety of these cables available depending on how long and durable they need to be. For XLR extenders, the example used for cost analysis was the Pro Co EXM-25, which comes with a lifetime replacement warranty and reaches 25 feet. For the TS extenders, we have the Pro Co EG-20, which has the same lifetime warranty.

Next, there were the cables for attaching the monitors and subwoofer to the interface output. None of these cables needed to be very long since the monitors and interface would be right next to each other. Based on the ports available on the JBL monitors and subwoofer, we needed two TRS to TRS cables to connect the subwoofer and two TRS to female-XLR cables to connect the monitors. The monitors would be plugged into the subwoofer, which would be plugged into the audio interface. For TRS to TRS cables we went with the Hosa CSS-105, and for TRS to XLR we went with the Pro Co BPBQXF-5.

Finally, there needed to be two more TRS to TRS cables to connect the two audio interfaces together. The total cost of all the audio cables is listed below in Table 2.3.

Table 2.3: Cost Breakdown of Audio Cables

Cable Type	Product	Count	Unit Price	Total Price
Room-to-room Snake Cable	Hosa Little Bro’ 6x2 25ft	1	$76	$76
Studio Snake-to-microphone	Pro Co EXM-25	6	$30	$180
Studio Snake-to-instrument	Pro Co EG-20	2	$21	$42
Interface to Subwoofer and Interface to Interface	Hosa CSS-105	4	$8	$32
Subwoofer to Monitor	BPBQXF-5	2	$20	$40
			Total:	**$370**

2.5 Software

The last piece of the audio puzzle was the software. Digital Audio Workstations (DAWs) are software packages used to edit audio tracks. The audio mixing computer will need at least one DAW in order to do its job. DAWs allow users to edit, playback, mix, and export audio. There are many DAWs available depending on your needs, budget, and personal preferences. These days there are even fully-featured DAWs that are completely free, such as Audacity and Garageband. Unfortunately for our use case, most DAWs are not designed with live-mixing in mind. If the audio software cannot keep up with the rest of the live stream, listeners may end up with noticeable lagging or stuttering. This is why we decided on *Ableton Live*, as it is one of the only DAWs that is specifically advertised to work in a live setting, where time is limited and crashes are unacceptable. A standard license of Ableton Live costs $450. Most functions in Ableton can be accessed quickly, and built-in automation tools can be used to simplify commonly performed inputs (Sprankle, 2021). However, Ableton has its weak points. The interface is a little unconventional, and making precision edits can be tricky. For instances when live audio is not necessary, we decided to include *Pro Tools*. Pro Tools is the industry standard among mixing and mastering engineers (Sprankle, 2021). A license for Pro Tools Studio (with the education discount) costs $100 per year.

In addition to a DAW, the audio computer will need software for measuring the frequency response of the room. Room EQ Wizard is one such software, and it is completely free. More information on this is provided in the next section.

There are also drivers for the audio interfaces that must be installed. Drivers tell a computer how to communicate with external hardware, and they can be downloaded for free from the Focusrite website.

Adding together the total cost of materials for the acoustic treatment, audio equipment, audio cables, and mixing software, the price came out to approximately $9,610.

3. Room Setup

This section is meant to aid in setting up the room after the acoustic treatment has been completed.

3.1 Wiring Guide

The wiring for the audio equipment is fairly simple, and all the necessary wire types are detailed in section 2.4. Figure 3.1 is provided as a visual aid for hooking everything together. The cable connection types between devices are indicated in small letters. To pass the snake cable between the studio and control rooms, a hole must be drilled through the wall. After all the wiring is complete, the edges of the hole should be sealed with caulk to create a soundproof barrier. Figure 3.1 does not include external power connections.

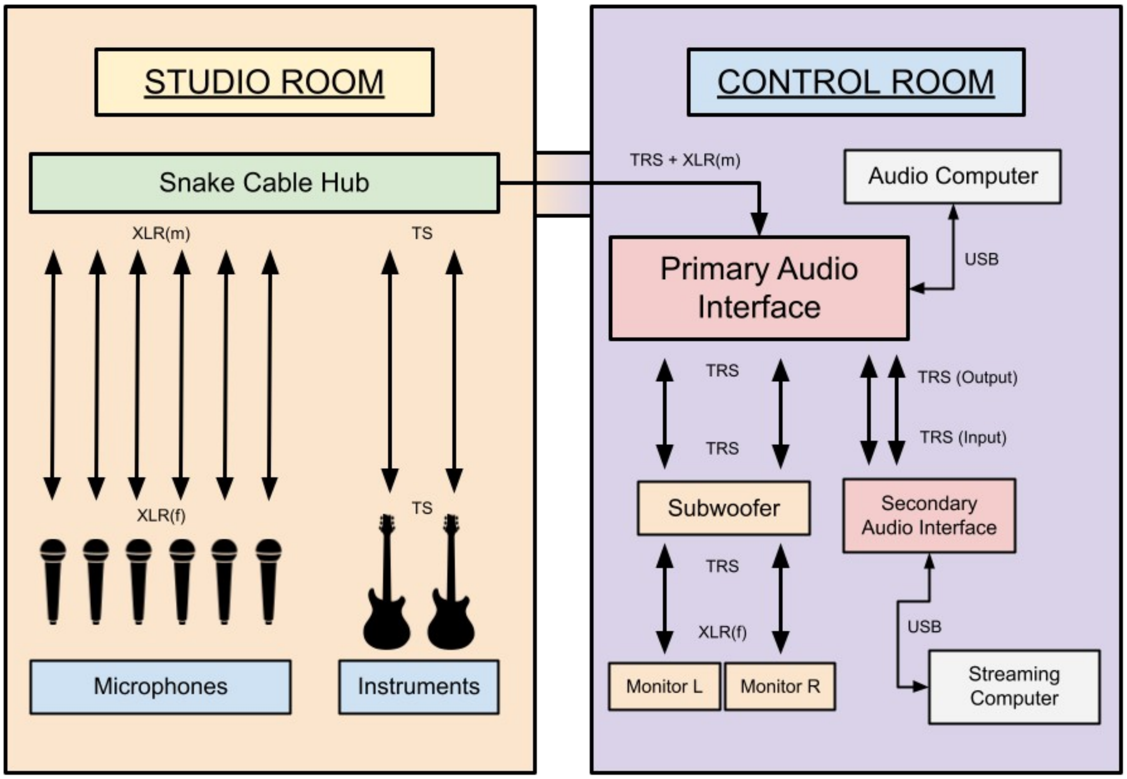

Figure 3.1: Audio Equipment Wiring Guide.

3.2 Acoustic Calibration

Once all the hardware is in place, you may begin adjusting the acoustics of the room using the free software called Room EQ Wizard (abbreviated as REW) (Mulcahy, 2022). The EMM-6 calibration microphone should come with a downloadable calibration file used by REW to remove bias from the input signal. REW will also need to generate a calibration file for the audio interface. To do this, the output of the interface must be plugged directly back into its input. Then run the calibration program built into REW, which will generate the file. After the hardware calibration is finished, the software is now ready to take measurements of the room's frequency response. These measurements are used to find the optimal listening position, speaker placement, and amount of absorptive paneling for both rooms. They can also reveal unexpected acoustic defects in the rooms that may require additional treatment. The specifics for most effectively using REW are outside the scope of this paper. More in-depth tutorials are available on the internet and on the REW website.

4 Conclusion

This has been a near-comprehensive guide on the audio-related aspects of converting the Riley Hall basement rooms into a fully-featured live streaming studio. The studio room has been designed using mathematical analysis of reverberation time and resonant frequencies, while the control room has been designed based on the live-end dead-end (LEDE) philosophy. Lists of necessary equipment and software have been presented along with the cost-benefit reasoning behind each choice.

Due to time limitations, some aspects of the design still require additional planning. The design and construction of resonating bass traps is one such aspect. Due to the multitude of possible resonator designs, this paper would not be able to explore the topic in enough detail. There was also external sound reduction, which requires a more detailed description of the pre-treated rooms than this paper has been working with. Sound reduction is a complex topic that includes everything from ventilation systems to structural support beam configurations, and therefore it warrants a dedicated investigation.

And remember: Aim for your dreams, but don't lose yourself along the way.

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Appendix A: Python Code

This is the Python code used to generate the mode frequency graph seen in Figure 2.1:

```
import matplotlib.pyplot as plt
import numpy as np
from collections import Counter
import math

#=======================CONSTANTS=======================#
C = 1130            #Speed of sound in feet per second
L,W,H = 10,15,10    #Room dimensions in feet

#Graphing and calculation parameters
X_MIN = 10          #Min frequency in Hz
X_MAX = 400         #Max frequency in Hz
MAX_MODE = 30       #The maximum number of loops when calculating modes. You probably
don't need to change this.
ROUNDING = 1        #The number of decimal places to round the frequencies to

#=======================CALCULATE MODES=======================#
a_freqs = []    #Axial mode frequencies
t_freqs = []    #Tangential mode frequencies
o_freqs = []    #Oblique mode frequencies

def get_frequency(p,q,r): #Gets frequency given mode numbers
   return C/2 * math.sqrt((p/L)**2 + (q/W)**2 + (r/H)**2)

for p in range(0,MAX_MODE):
   for q in range(0,MAX_MODE):
       for r in range(0,MAX_MODE):

           freq = round(get_frequency(p,q,r),ROUNDING) #Calculate frequency

           if freq<X_MIN:
               continue    #Ignore out-of-bounds frequencies
           elif freq>X_MAX:
               break       #Don't need to calculate any higher modes

           #Add mode to appropriate List
           if (p+q==0 or p+r==0 or q+r==0):#Mode is Axial
               a_freqs.append(freq)
           elif (p==0 or q==0 or r==0):    #Mode is Tangential
               t_freqs.append(freq)
           else:                           #Mode is Oblique
               o_freqs.append(freq)

#Counters keep track of how many times a particular frequency appears in the List
a_counter = Counter(a_freqs)
t_counter = Counter(t_freqs)
o_counter = Counter(o_freqs)

#=========================PLOTTING=========================#
fig,(ax1,ax2,ax3) = plt.subplots(3)

#Axial Mode Plot
x_vals = list(a_counter.keys())
y_vals = list(a_counter.values())
ax1.bar(x_vals,y_vals,color=(250/255, 142/255, 10/255))
```

```
ax1.set_xlim([X_MIN,X_MAX])
ax1.yaxis.set_ticks(np.arange(0, max(y_vals)+1, 1 if max(y_vals)<4 else 2))
ax1.set_title("Axial Modes")
ax1.set(ylabel="Mode Count")

#Tangential Mode Plot
x_vals = list(t_counter.keys())
y_vals = list(t_counter.values())
ax2.bar(x_vals,y_vals,color=(7/255, 237/255, 34/255))
ax2.set_xlim([X_MIN,X_MAX])
ax2.yaxis.set_ticks(np.arange(0, max(y_vals)+1, 1 if max(y_vals)<4 else 2))
ax2.set_title("Tangential Modes (-3dB)")
ax2.set(ylabel="Mode Count")

#Oblique Mode Plot
x_vals = list(o_counter.keys())
y_vals = list(o_counter.values())
ax3.bar(x_vals,y_vals,color=(26/255, 69/255, 240/255))
ax3.set_xlim([X_MIN,X_MAX])
ax3.yaxis.set_ticks(np.arange(0, max(y_vals)+1, 1 if max(y_vals)<4 else 2))
ax3.set_title("Oblique Modes (-6dB)")
ax3.set(xlabel="Frequency (Hz)", ylabel="Mode Count")

#Display Plots
fig.suptitle("Room Modes at Low Frequencies")
plt.tight_layout() #Formatting fix
plt.show()
```

Appendix B: Room Renders

Displayed are 3D renders of the example room design shown in Figure 2.2. Room dimensions and wall-mounted panels are all to correct scale. The size of the doors, television screens, table, and studio monitors are all approximate. Resonating bass traps are not shown, as their design remains unknown. Colors are chosen to distinguish between materials and are not representative of the final aesthetic design. The room was modeled in Blender and rendered with Cycles. All textures are attribution-free.

Top-Down View of Both Rooms:

Studio Room:

Control Room: