CASE STUDIES IN CLASSICAL LOCATION RECORDING
USING IMPROVISED TECHNIQUES

A Thesis
Presented to
The Honors Tutorial College
Ohio University

In Partial Fulfillment
of the Requirements for Graduation
from the Honors Tutorial College
with the degree of
Bachelor of Science of Communication
in Media Arts & Studies

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April 2015
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1. ABSTRACT

This thesis discusses classical location recording from an improvisational standpoint. Improvised techniques in audio production are methods for recording, studio design, and communicating with clients that are not standard within classic audio literature such as textbooks, scholarly articles, or online educational resources. This thesis is comprehensive to every aspect of recording classical music on location, such as mobile studios, equipment design, pre-production, recording and mixing. Everything is explored at length to demystify the entire process through the improvisational scope. Additionally, many of the ideas presented are advanced, requiring tools and calculations that are often beyond the standard repertoire of novice and amateur classical engineers. These techniques are condensed and explained for their practical use, in hope that all engineers can implement this heightened fundamental understanding to design better recording systems. The conclusion of these materials aims to provide classical engineers with a new, unique, and grounded perspective with which to approach the recording process.
2. INTRODUCTION

Recording classical music on location is an art form that requires not only broad strokes to capture an orchestra and reproduce the music transparently, but also a precise and infallible attention to detail bringing subtleties that often go unnoticed to life. Just as a painter has a brush, the recording engineer has a studio. The broad strokes of recording are the canon of recording techniques that an engineer is capable of implementing, founded by experience and a wide background in audio literature. But with only this, the act of recording a symphony on a budget is utterly mundane. An engineer can pick from an assortment of techniques- practical techniques drawing from the equipment that he or she has, setting it up, and finding contentment with the result because it is the best that a recording on a budget can sound. There is then no room to accommodate little nuances within the music and the beauty of a piece will be lost. But, the broad strokes can be revolutionized if improvised techniques are used. Just as a painter with only a brush can enhance his or her art by smearing the paint with a finger, flicking paint off the brush, or any other improvised techniques. Classical engineers draw from the same inspiration to make their recordings great.

This thesis aims to demystify recording classical music on location for those who are novices in the craft, and explore improvised recording methods that break the budgetary barrier- a limiting factor for the majority of audio engineers. Within this thesis, numerous aspects of classical location recording will be explored. Although, many subjects will be abbreviated, as it would be futile to reiterate the lengthy details from which many ideas are drawn, especially because the grandeur of the source
materials is not easily rivaled. The aims of this thesis are to condense and clarify other recording texts to deliver practical explanation of the most important topics in audio. Principles will be explained for their functionality, without going into detail about their derivation. In these cases the source material has been provided in an effort not to diminish the importance of the original research in any way. Audio literature is not cryptic or difficult to find; rather it is readily available. Today’s audio engineer must first learn to find these resources above all else.

Neither the novice engineer nor the seasoned engineer should stray at this point. Keep in mind the perspective from which this is written, as I feel it is important to clarify. I am a young, classical recording engineer with only five years of practical experience; my passion lies in continuing to learn and refine my craft. I have spent these five years deeply immersed in audio literature, practice, solving problems, and ultimately trial and error. This thesis is comprised only of the most relevant information to learning within classical location recording, including articles that stuck out as being particularly useful, tools I have designed to overcome budgetary restrictions, and improvised techniques I developed to complete projects in the most efficient way with the best sounding results. This journey has been filtered for the most useful information. Ultimately, this thesis is from one learning engineer to another, so use it as crash course in becoming a better engineer.

Much of the material herein is still original information though, lest the reader get the impression that this is only a compilation of other works. Almost every successful audio engineer developed his or her own methods for recording, journeyed
through the tedious process of acquiring and building a functional recording studio (or a mobile recording studio in this case), and had a period of time before their recordings evolved from needing improvement to being great. This process requires a great amount of experimentation and exploration that cannot be read about online or taught in a college audio course. Short of having a great mentor, audio engineers are often left alone to figure things out themselves. Much of this thesis is the result of that process, such as instances in which drawing upon innovation was the only way to overcome budgetary barriers.

As one final introductory thought I encourage the reader to listen to the supplemental materials while reading this paper via this link. There are five recordings performed at the 2014 Lake George Music Festival, and they are in accordance with the case studies in this these. They are in both 16 bit/48KHz WAV, and 320kbps MP3 format for your convenience.
3. MOBILE PRODUCTION STUDIO DESIGN

The first musical recordings in the history of audio were performed on location. Most notable is the 1888 recording of Handel’s *Israel in Egypt*, which was recorded in London’s Crystal Palace and featured a choir of over 3,000 singers. It was recorded directly to a wax cylinder\(^1\) designed by Thomas Edison. This is significant because today there is still no brick-and-mortar\(^2\) studio in the world capable of reproducing this feat, since no studio can comfortably stage that many people. This is the heart of mobile recording, which is to go beyond what normal studios can do. This goes without saying: the mobility of studio equipment is second to nothing (an admittedly opinionated statement).

Aside from a strong background in classical recording skills and literature, a high performance, reliable mobile production studio is a core element in recording classical music. Today, few classical engineers have the luxury of working solely out of a brick-and-mortar studio. Only a small percentage of studios in the world have enough space to accommodate a full-size orchestra, and the amount of overhead required to operate such a large studio is too much when the majority of the recording in classical music is for chamber ensembles. For this reason it is imperative to maintain the option to record in a variety of spaces such as churches, venues, or rehearsal spaces. Additionally, not all studios are equipped with proper instruments for

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\(^1\) A revolutionary recording medium, in which sound was etched into wax on a spinning cylinder.

\(^2\) Brick-and-mortar studios are non-mobile studios, with all installed equipment.
specific pieces of music, such as a pipe organ or a genre specific style of grand piano. This means that a Yamaha piano may sound perfect for jazz, but almost intolerable for classical music in contrast to a Steinway. For these reasons, mobility is key in classical music production.

What qualities must a good mobile production studio possess? Ease of transport, reliability, price, and audio fidelity are important, but are not all-inclusive answers. Audio equipment can be good, reliable, or cheap. However, it can never be good, reliable and cheap. Take for example Millennia microphone preamps\(^3\). Most are transformerless\(^4\), totally transparent\(^5\), and will provide the cleanest preamplification on the market. Millenia preamps can also last decades without needing maintenance if treated correctly. However, they are extremely expensive even when purchased second-hand. Other preamplifiers on the market may be cheaper and still sound nice, but will deteriorate on the road because of their inferior build quality. This is true for all equipment, from microphones to monitoring\(^6\) devices. Other cost considerations are important, such as weight, size, and longevity. This is the focus for making purchasing decisions, and finding the correct tradeoffs in building a mobile production studio.

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\(^3\) A preamplifier is a device that amplifies the extremely weak signal of a microphone to a standard operating level compliant with standard equipment.

\(^4\) Having a circuit design that does not use a transformer. This usually allows cleaner and more precise audio.

\(^5\) The sonic quality of reproducing the sound source without altering the sound, so the output sounds exactly the same as the input, only louder.

\(^6\) Monitoring is the act of listening critically to audio.
3.1 Workflow vs. Sound Quality

As a discussion of recording classical music with a budget conscious focus, the worth of an item is essentially broken into two categories:

1.) Equipment that *directly* improves the quality of the recording.

2.) Equipment that *indirectly* improves the quality of the recording.

An engineer must ask him or herself an important question before making any investment. Will the piece of equipment actually enhance the level of audio fidelity, or will it improve the workflow of the studio, making his or her job easier? Investing in workflow improvements will make the studio more user intuitive and will save time during operation without having any direct impact on the audio.

For example, a microphone is an investment that can directly impact the sound of a recording, and hopefully improve it. A $2,000 Schoeps CMC 6 microphone and a $200 Røde NT5 microphone both have the exact same function despite their conflicting price tags; they are both cardioid small diaphragm condensers (SDC)\(^7\), both the same size and weight, and implementing one is the same as implementing the other. However, the Schoeps will perform far better when compared side by side with the Røde NT5 under identical circumstances. It will have a far greater signal to noise ratio (SNR)\(^8\), a much more transparent signal, and will last much longer among several other benefits. Purchasing a Schoeps CMC 6 over a Røde NT5 is an example of an investment which directly impact the recording sound quality for the better.

\(^{7}\) A condenser microphone with a capsule equal to or less than approximately 16mm.

\(^{8}\) The difference in level between the audio captured by the microphone, and the self-noise/operating noise that is present in all equipment (Eargle, 122).
In contrast to items that directly affect the sound that is being recorded, certain hardware accessories can indirectly impact an engineer’s quality of work. A more powerful computer does not directly make your recordings sound better. A better stereo bar does not alter the signal that the microphone is capturing at all. A proper, sturdy combination boom stand does not hold microphones in a way that makes them sound better. Rack cases that roll do not change the way any of the signal processing\(^9\) gear housed within it operate. But, all of these seemingly unimportant and mundane pieces of equipment are the unsung heroes of the classical location engineer.

To break this down I will explore the advantages of these seemingly indirectly related items. First, the computer. While owning a computer with a suitable Digital Audio Workstation (DAW)\(^{10}\) by itself is definitely a sonic advantage over simply a stereo recorder without any ability to mix, owning a faster computer does not inherently produce better sounding audio than a slower computer. But, the manner in which the faster computer works for the engineer can allow the engineer to do better work. More instances of highly CPU intensive plugins\(^{11}\) can be used, less time is spent

\(^9\) Signal Processing gear is equipment that manipulates sound in any way.

\(^{10}\) A Digital Audio Workstation is a software program for recording, editing, mixing, and mastering audio.

\(^{11}\) Plugins are audio processing programs within Digital Audio Workstations. In this context, some plugins require great amounts of a computer’s resources, and only a finite amount of them can be used on any particular system depending on the processing power of the computer.
waiting on the computer while more time can be used for creativity, and the overall experience of mixing in-the-box\textsuperscript{12} is enhanced.

With microphone mounting gear, such as stereo bars\textsuperscript{13} and microphone stands, the engineer can expedite stereo microphone setups much faster. An engineer can purchase and use an On Stage Stands stereo bar, but he or she will have an absolutely wretched time trying to assemble a pair of small diaphragm condensers into an ORTF\textsuperscript{14} position. Røde makes a rather useful stereo bar that includes vertical spacers and a variety of labeled measurements (such as ORTF), and creating any stereo system\textsuperscript{15} takes only seconds with it (Streicher, 551). As for microphone stands, buying a 4m stand and a boom attachment is great for classical recording, but they often require assembly and reassembly while on location, which is a time consuming process. On the contrary, a combination (or Combi) boom stand can be purchased from companies such as Impact or Manfrotto which has excellent strength, rigidity, and takes a fraction of the time to set up. Combination boom stands can be used as stands that only extend straight up but can also extend their last sections out of a clutch to become a boom stand. While it is twice as expensive it does not actually

\textsuperscript{12} Mixing in-the-box is mixing on a computer without any external signal processing equipment.

\textsuperscript{13} A stereo bar is a microphone mount for two microphones that can be positioned how the engineer needs.

\textsuperscript{14} ORTF is a stereo recording technique that places two microphones at a 110 degree angle from one another, and 17 centimeters apart.

\textsuperscript{15} A stereo system is a pair of microphones that are positioned according to an engineer’s calculations. A stereo system is usually calculated to record an ensemble with a high degree of accuracy in the stereo image.
sound better than any other stand and boom combination, the time saved can be well worth it.

Finally, even a good rack case\textsuperscript{16} is crucial to the engineer’s mobile studio. This is one of the most basic luxuries most engineers own. Weight can add up quickly; moreover a case with wheels is almost mandatory in spite of the extra cost. Additionally, a quality case will extend the longevity of the gear it holds. Simply put, at times extra money must be spent to make the process of recording bearable. After all, this sort of expense only impacts profitability temporarily, and has little long-term monetary effect.

A smart engineer must find the balance between audio fidelity and good support equipment for his or her own studio and budget. Most engineers want to achieve the highest level of sound fidelity in their studio before purchasing gear that will improve their workflow, but a level of compromise must be made. All microphones, regardless of their price and function demand proper support equipment and proper implementation. Mindfulness of this is imperative when making purchasing decisions in a low to mid budget studio.

\subsection*{3.2 Building a Studio}

No matter what a project requires, there are certain fundamentals in a studio that must be made for all steps of the recording process to be met. If an engineer wants

\textsuperscript{16}A rack case is a transportation unit that holds a varying number of rack mountable audio devices, depending on its size.
to record classical music on location in a serious fashion, he or she must have the following at the most basic level:

1. At least two microphones for stereophonic recording\(^{17}\)
2. A device to hold microphones, whether suspended or on the ground
3. At least two preamplifiers
4. An analog to digital converter\(^{18}\)
5. An editing and playback system to process the recordings
6. All the necessary cables to connect the devices

These are important, but paramount is the ability to expand into larger, better sounding, or more efficient systems. This is always the first thing I tell others when I am asked about buying equipment. The worst item anyone can possibly buy is that which cannot be integrated into a larger system in some way. This is true for gear that does not have optical (ADAT)\(^{19}\) connections, S/PDIF\(^{20}\) connections, or other digital ins and outs that are crucial to the interconnection of recording equipment. For example, a Focusrite Scarlet 2i2 limits the engineer’s recording capability to two

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\(^{17}\) Stereophonic recording is a method of sound reproduction that uses two simultaneous and independent playback sources to create localization, depth, and width.

\(^{18}\) An analog to digital converter converts electroacoustic energy produced by microphones and other analog audio equipment to digital data that can be processed by a computer

\(^{19}\) Alesis Digital Audio Tape (ADAT) is a digital connection format that can carry up to eight independent channels of uncompressed digital audio.

\(^{20}\) Sony/Philips Digital Interface Format (S/PDIF) is a digital connection format that can carry two channels of uncompressed digital audio.
channels of audio, and cannot send its own two channels of audio into any other unit. It is a dead end piece of gear, and while it may suit the engineer’s needs initially, it probably will require outright replacement in time.

Today, for any good classical recording, a pair of microphones is mandatory for stereophonic recording. The engineer will use this pair create a stereo system that will accurately capture the space and depth of the music and the venue it is played in. Stereo recording is important because the goal is usually to accurately reproduce the performance so it can be listened to for years to come. In order to accurately reproduce audio, engineers must use at least stereo recording to reproduce the human hearing system, which is binaural. Generally, a great sounding and cost effective solution for classical music are condenser microphones. Within the category of condenser microphones, generally there are two subsets that further affect the cost efficiency. Microphones that sound great but are not flexible, and microphones that are flexible but do not sound as good.

For example, two similarly priced microphones are the Neumann TLM103 and the AKG 414 ULS. Both are highly regarded microphones in the $1,100 range. The Neumann is simply a cardioid microphone, but truly sounds great for its price. The

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21 Binaural literally means “having two ears,” meaning humans can localize sounds that are to the left and right of them.

22 A polar pattern in which a microphone only mainly responds to sounds that are in front of it.
AKG has up to nine polar patterns\textsuperscript{23}, a two position pad\textsuperscript{24}, and a high pass filter\textsuperscript{25}, and while it still sounds nice, it does not quite hold up to the Neumann’s superior sound quality. For the same money, these microphones have a stunningly different type of value. One can purchase an AKG 214, which actually contains the same capsule as the AKG 414 ULS but has a fixed polar pattern of cardioid. It will sound identical to the 414 ULS while it is in cardioid for only $400, which is a staggering price difference. Different audio engineers will vehemently debate which of these is a better pick, as there is no clear answer for which is better (Williams, 155).

For holding these microphones, one can naturally purchase microphone stands. Traditional lighting stands, however, are invariably a cheaper and higher quality option. Impact makes a 13ft, air cushioned, sturdy, lightweight stand for only $50. These are absolutely fantastic for classical recording, and only require a 3/8” to 5/8” threaded adapter to fit microphone attachments. A similar product from a dedicated microphone stand manufacturer such as K&M or Latch Lake would be at least five times costlier. Finally, to use two microphones in a stereo array, it is far less obtrusive to purchase a stereo bar instead of using two microphone stands.

For preamplifiers and analog to digital conversion, most starting engineers will purchase an interface that will do all of this together. This is the area where it becomes

\textsuperscript{23} A polar pattern is a representation of the directions that a microphone responds to sound on a horizontal plane.

\textsuperscript{24} A pad on a microphone lowers the audio input to its internal circuitry, protecting its components and lowering its overall output.

\textsuperscript{25} A high pass filter (HPF) allows all sound that is higher in pitch than a set frequency to pass, ultimately attenuating low pitched sounds.
slightly more complicated, because of the previously mentioned importance of expandability. Instead of something like the Focusrite Scarlett 2i2, consider something slightly more expensive such as the Focusrite Pro 40. I use this as an example because it was my own first interface, and even though I have upgraded to a MOTU 896mk3 as my primary interface, the Pro 40 easily integrates into the 896mk3 to act as an ADAT extension. This means the MOTU interface receives the digital output of the Focusrite interface and transmits that information to the computer along with its own information. Additionally, the MOTU still has another bank of ADAT connections free, offering further expandability still. At the risk of sounding like a sales pitch, this is an extremely important factor when choosing an interface.

Furthermore, there are many other qualities in recording interfaces. First, is the level in quality of the preamplification. High quality interfaces will have preamplifiers that have low levels of self-noise, low levels of harmonic distortion, and generally sound pleasing to the ear. A good preamp is the best compliment possible to a good microphone. Good interfaces will have more switches for -10dB or -20dB pads, high pass filters, or even polarity inversion as well. These features are not always required, but are great to have. Additionally, be aware of the physical format and build quality of an interface before purchasing it. Rack-mountable designs will physically integrate into other systems much more easily, and a metal chassis is always a testament to a unit’s durability. Even paying attention to whether or not the interface has inputs on the front or back is important to how an engineer will implement the interface.
This is an excellent time to mention the necessity of owning a good power conditioner. Power conditioners perform three main functions. First, they provide several outlets within a racked road case for your convenience. Secondly, they are a reliable means of surge protection in the event of an electrical accident. Finally they effectively “clean up” the AC power through intelligent filtration (Janssen 1). This can actually increase the sonic performance of the equipment it powers. Through side-by-side comparisons I have conducted through my own equipment, the difference is extremely subtle. However, I have found that it is more noticeable when a system goes from being conditioned to not conditioned rather than vice versa. For this reason, I regard my power conditioner, a Monster PowerCenter Pro 3500, as something I cannot live without.

An editing and playback system is the final must-have item for a classical recording mobile studio. This involves having a computer running the digital audio workstation of your choice, a pair of headphones, and a pair of studio monitors. Any type of computer will work for recording classical music as long as it has a sufficient amount of power and is portable enough to bring on location. Laptops are often ideal for most engineers, although a small desktop computer such as a 2014 or earlier Mac Mini with two internal hard drives and a small USB powered monitor is my own preference. For DAW’s, most classical engineers find that programs such as Pro Tools, Adobe Audition, soundBlade, or Pyramix are suitable for classical recording.

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26 Apple has since discontinued the production of Mac Minis with two internal hard drives. Having two internal hard drives eliminates the need for an external hard drive, since it is always preferred to record audio to a separate hard drive than the computer’s system drive.
for various reasons. Stray from using Logic Pro, Cubase, Ableton Live and other like programs for classical music. These programs do not provide adequate precision editing tools or metering\textsuperscript{27} options that are needed for classical production.

Both headphones and studio monitors are mandatory for recording classical music. Headphones are required because setting up studio monitors at a venue is rarely an option and the engineer needs a way to monitor the audio that is being recorded. Monitors are a necessary investment as headphones cannot reproduce an accurate stereo image for the engineer (which will be explained). Closed back\textsuperscript{28}, circumaural\textsuperscript{29} headphones are the most-common type of headphones for this type of recording, as opposed to open back\textsuperscript{30} or supra-aural\textsuperscript{31} headphones. Sennheiser HD280’s, Shure MDR-7506’s, Audio Technica ATH-M50’s, and Beyerdynamic 770’s are all great, relatively inexpensive headphones for this purpose.

Monitors are a tricky subject to discuss. For mixing in a small to medium sized room, 6” monitors are arguably the largest monitors possible without creating more

\textsuperscript{27} Metering is a term for visualizing audio on a graphic user interface (GUI).

\textsuperscript{28} Closed back headphones physically seal the engineer’s ears from outside noise, having no holes or ports to let sound through.

\textsuperscript{29} Circumaural headphones completely cover the ears.

\textsuperscript{30} Open back headphones allow sound to flow through the cups of the headphones, and provide little isolation from outside noise.

\textsuperscript{31} Supra-aural headphones rest on top of the ears.
problems than they solve to begin with. Additionally, ported monitors will drastically decrease the low-end accuracy of a mix compared to sealed monitors.

According to Mike Senior, buying ported monitors will create excess low-end energy in the mixing room, that will result in extremely inaccurate portrayals of the sound reproduction. Monitors with ports and large speaker cones create an excess of nodes and antinodes in the room, where certain frequencies build up or cancel out (see figure 3.2a). This means that a 100Hz sine wave could be 70dB at one point in the room, and be 85dB a few feet further back from the monitors (Senior 5). These are also known as standing waves. Smaller monitors are not an outright fix to this issue, but are a big help.

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32 Ported monitors have a carefully placed hole in their construction that allows sound produced by the rear of the speaker to escape the enclosure.
33 Sealed monitors do not have a hole in their design, allowing only sound from the front of the speaker to transmit.
34 A speaker cone is the actual speaker element that is mounted in a monitor enclosure.
Another downside of purchasing studio monitors is that they are usually more expensive than engineers are willing to spend in their early years. A decent pair of monitors will cost at least $500, and buying monitors any cheaper are often not even worth buying, as quality almost always correlates with price. Cheaper monitors will give an extremely skewed picture of the actual audio they are reproducing, and leave the engineer sonically blind to their mix. They often hype the audio, accentuating bass and treble frequencies to fool amateur engineers into buying them, but they will do nothing to give them an honest reproduction of the audio they are processing. Names
such as M-Audio, KRK, and PreSonus are all brands to avoid, while Yamaha, Tannoy, Mackie, and Dynaudio are reputable companies that make relatively inexpensive monitors of a respectable quality.

Lastly, the engineer must make or purchase all of the cables to interconnect every device, and make or purchase the necessary cases to transport the equipment. For cables, analog cables are relatively easy and cost effective to solder together yourself, while digital cables are almost always preferable to purchase from a retailer. The best practice while doing this is to only use cables that are just long enough to connect between their two destinations and remain out of the way. One can find extremely short IEC cables, ADAT cables, S/PDIF cables for low prices online, which are perfect for the interconnection of parts inside of a rack case. Additionally, the connectors and bulk wire to make XLR and TRS cables can be purchased to save money, and ensure high quality cables. Parts-express.com and sweetwater.com are great resources for these items.

For classical location engineering, long XLR cables and cable snakes are essential. I use several 50ft and 100ft XLR cables regularly, and have two cable snakes that are absolutely invaluable to me. Cable snakes are sets of cables that fan out to several male XLR connectors on one side, and often have a stage box of matching

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35 IEC cables are power cables for electronic equipment that operates at 120, or 240 volts.

36 Balanced XLR cables are standard microphone cables.

37 TRS cables are balanced ¼” cables, usually used for the interconnection of professional audio equipment.
female XLR connectors on the other side. One of my snakes is 100ft long and has sixteen XLR inputs and four TRS inputs, which is perfect for making long cable runs from stages to the backs of auditoriums, or even side rooms. The other snake is 50ft long, which is perfect for smaller venues, and requires less time to wrap and unwrap. These are items I could not survive without, since setting up on stage or even just off stage is never an option. To put this further into perspective, I would sooner part with several of my microphones before parting with either cable snake!

As for purchasing a road case to transport equipment in, there are a few options. First, a used road case is hardly any different than a brand new one and is a smart way to save money. An aesthetically pristine road case, free of scratches and dents will not sound or protect its equipment any better. Furthermore one can make transportation equipment his or herself. For example, every engineer has experienced the nuisance that every microphone, or pair of microphones comes with its own separate case, making carrying fifteen microphones into a venue can take several trips. For that reason, I built a wooden case loaded with Pick N Pluck foam\textsuperscript{38} that holds every one of my microphones and microphone accessories such as clips, stereo bars, shockmounts, and adapters. This saves me several trips in and out of the venue, saving time for actually setting up the equipment (see figure 3.2b).

\textsuperscript{38} Pick N Pluck foam comes in 2” thick sections, and 0.5”x0.5” squares can be pulled out to customize the shape of the foam.
Otherwise, Pelican cases are useful for the transportation of cables and other hardware. SKB and Gator make great rackmount cases and offer great warranties and replacement plans. For carrying microphone stands from venue to venue, one can purchase purpose-built canvas carrying cases that hold seven or eight stands comfortably. Additionally, a collapsible cart is extremely useful even though many cases have wheels. Carts save trips, and make loading in and out much less tiring leaving the engineer more time and energy for the process of recording itself.

3.3 Final Notes

Building a mobile recording studio is a stressful process. I often found myself putting together several Microsoft Excel spreadsheets comparing features, options, and prices, but when it came to actually ordering the equipment it still felt like a leap of
faith. It is nearly impossible to hear every piece of gear before buying it, and while online reviews are sometimes the only resource to back up the purchasing decision, they are rarely reliable. I find that there are three important reminders that stand out when putting together a mobile recording studio. They are the importance of the aesthetic, the importance of setting up quickly, and the fact that studios grow and constantly change in their interoperability. It is fair to say that no successful engineer has ever invested $10,000 in a recording studio and been so immensely happy with the result that they never changed anything. I usually change or adapt my equipment and workflow at least once a month. They are not unlike routine updates, and they affect everything from the sound quality of the recording to the aesthetic of how clients perceive the structure of my work.

That aesthetic of the production is important. It includes everything from how the engineer presents his or herself, to how the equipment looks during load in and load out. First and foremost, all equipment must look neat and tidy. It must be properly organized, easy to access, and look logical in one way or another. For example, it is not enough to transport all cables and hardware in a pelican case, it must be organized within. Use dividers, Pick N Pluck foam, or anything else to make the equipment look presentable. Organization is critical in the eyes of the client.

This is part of the reason I built a wooden case for all of my microphones. Yes, it is necessary to transport and safely house all of my microphones at once, but it achieves the aesthetic I want. The case is well built with dado joints, dowels, and a good wood finish, so it is as attractive as it is strong and functional. It is the
presentation of the microphones that make the client want to recommend me to their friends.

To set up quickly at any venue, simplicity must be taken advantage of. In essence, how can the engineer design their mobile studio in a way that takes the fewest amount of steps possible to set up? Many engineers can set their primary road case down (containing some preamplifiers, an interface, and other important gear), plug a single power cable into an outlet, plug a single data cable into their laptop, and be ready to record. This takes less than a minute, which is hugely advantageous over engineers who must stack several cases and interconnect a dozen cables to get online. For this reason it is often smart to use cases that are big enough to hold as much gear as necessary to the process of recording without becoming too heavy. That way, all cables can be left in place during transportation, and fewer cases need to be opened and set up as well, saving additional time.

Finally, studios change and grow; a well known reality in the audio industry. Extremely few audio engineers have the money to invest in a high quality studio at the start of their careers, or even want to invest such a large amount of money all at once. Even once a studio is effectively completed with the highest quality gear available, newer and better technology arises annually. So when initially investing in a mobile production studio, be realistic. As previously mentioned, purchase equipment that is able to expand in one way or another, or is reasonably flexible in what it can accomplish. These decisions are all up to the individual engineer, so always purchase what is best for current purposes, but have the future in perspective at some level. It
takes serious planning and research, but this is the reality of building a mobile recording studio.
4. DESIGNING THE NECESSARY TOOLS

In designing a classical mobile production studio, there are certain items that can directly and drastically improve the quality of your recordings but are simply too costly or are not readily available on the audio market. Microphone mounting equipment is a great example, and Decca Tree\textsuperscript{39} arrays are one of them. Recording in Decca Tree can have a truly heightened sense of space over a stereo pair, but mechanically it can be a complicated setup. Only two companies make these commercially (Grace Designs and AEA), and they are upwards of $700 and still have their limitations. Another item is an orchestra angle\textsuperscript{40} viewer, which is an angle measurement viewer to help you configure a stereo recording system designed by Michael Williams and discussed in his paper *The Stereophonic Zoom* (Williams 16). It is called The Williams Crocodile, and nobody except for Michael Williams makes them. It can be indispensable in the field, and leave you bulletproof when there is no margin for error in live recording scenarios. Finally, there is a myriad of other microphone mounting equipment that is not available for purchase as a full package. If one needs a truly unique microphone support rig for a specific venue, they have no choice but to design it themselves.

\textsuperscript{39} A microphone technique named for Decca Records that is commonly used in orchestral recording. It usually features three omnidirectional microphones spaced left, forward, and right 70-100cm from a center point.

\textsuperscript{40} The orchestra angle is the angular width of an ensemble from a given point. The crocodile angle viewer is built to measure this.
Overall, the takeaway of this section is that anything can be done when left up to the imagination. Beyond the Decca Tree array and the Crocodile angle viewer, I have designed several modular microphone mounting solutions that can be integrated into almost any venue. Each design originated as an idea, as a problem needed solving or a more economic solution was necessary. In this case, classical location engineers are ever in need of better ways to mount microphones. It is relatively easy to buy the parts to install a single microphone system in a venue, but it is a whole new challenge to create a modular system that can be integrated quickly into any venue. The various systems described in the third subsection are all designed with aesthetics in mind, as they are all alternative methods to place microphones before a stage that do not require placing a microphone stand in front of the ensemble. This is an extremely important idea for classical location engineers, and the improvisational focus of this thesis is perhaps one of the only ways to achieve such a goal.

To note, always test designs at home before implementing them in a live recording scenario. These designs have been perfected for my own practical use, and any engineer will need at least one attempt to practice it on their own to get it right. For anyone using any unique design, these are new tools that require practice to use. There will always be kinks to work out in the workflow, and a system of getting any design set up on location must always be worked out and able to be replicated time and time again.

Lastly, with one more note on the aesthetic concerns, while these are absolutely Do-It-Yourself (DIY) constructions, they do not have to fall victim to the
negative connotation that improvised designs look ugly. Too many DIY designs that rotate through online audio forums use PVC or copper pipe, and look absolutely hideous as a result. They are often dangerous as well, using heavy and ill-fitting parts that threaten to crash to the floor at any moment. As a measure of professionalism, these designs are as attractive as they are functional and practical. Carbon fiber is a perfect material for designing microphone suspension systems. Additionally, the parts are almost entirely black, which is perfect for live use. The next three subsections are dedicated to the use of DIY and improvisational mentality, while being unquestionably professional and usable.

4.1 Carbon Fiber Decca Tree Array

Decca Tree recording is a staple of orchestral recording techniques. It was first developed by Decca Records personnel in the early 1950’s, and since then has become a widely used stereo technique for those who have the necessary equipment. It is usually described as an AB stereo system⁴¹ with a center fill⁴² extending forward. It has no fixed measurements, but each microphone is usually placed 70-100cm from the center point, as shown in figure 4.1a from Los Senderos Studio’s website.

⁴¹ A stereo system in which two omnidirectional microphones are placed equidistant from the sound source, spaced at a specified distance from one another. Cardioid microphones may also be used, but must both be pointed at the sound source.

⁴² A center fill is used to fill the center gap that some stereo recordings lack. Stereo systems can mistakenly be calculated to be too wide sounding, leaving the left and right sounds feeling disconnected. A properly used center fill can help correct this issue.
This allows the recording engineer two primary advantages. The first, is that the separation between sections in an orchestra is especially good. For use with a standard stringed orchestra, violin I’s and II’s are easily perceived on the left of the stereo image, the violas and cellos are easily heard toward the center right of the stereo image, and the basses are heard further on the right. This would be a properly localized stereo recording for that ensemble. Secondarily, the engineer has flexibility in mixing. He or she can raise or lower the volume of the center microphone to fill the center of the stereo image, a problem that many engineers cannot fix when a two microphone stereo system is not calculated correctly.
In no way am I the first person to build a Decca Tree array at home and use it in the field. I have seen various prototypes and homemade arrays that have been used in the field, through online audio forums and social media, and quick search on forums such as Tape Op, or Gearslutz will reveal dozens of results. However, there is a certain standard that so few custom Decca Tree arrays have met. A unit made from PVC or copper tubing simply cannot be put on a stage during a professional live performance. Aesthetically it is distracting, and looking at plumbing on or near the stage is utterly displeasing. They could possibly be used in a studio setting without an audience, but it still does not appear to be professional to the client, and they may consider going to another engineer for future recording. Why should they pay over $100 per hour for an engineer whose equipment appears to be made out of basement scraps? The standard for a good Decca Tree array as high, but luckily, the following section describes how I created mine (see figure 4.1b).

Fig. 4.1b. My custom carbon fiber Decca Tree array recording a thirteen-piece orchestra.
First, consider other professionally manufactured Decca Tree arrays. Grace Design, a professional audio manufacturer specializing in extremely high fidelity recording equipment has an array available through Sweetwater.com at $675. Audio Engineering Associates (AEA), which is actually owned by Wes Dooley who authored a paper explaining Decca Tree recording, has one available for over $850. Furthermore, purchasing more accessories for these models is extremely expensive, at more than $100 for something as simple as a single modular microphone mount. Neither of these models collapse smaller than one meter, and both are heavy and bulky compared to my carbon fiber model.

Carbon fiber is a great material for making microphone mounting hardware. It is durable enough to be used repeatedly in the field without wearing out or breaking. It is much lighter than any other material that would be considered practical for suspending microphones or other equipment. It is black and does not reflect light which is a must for placing equipment on stage for recording live performances. One meter of 0.75” carbon fiber tubing is strong enough to hold a light microphone horizontally on its end while being fixed at the other end with an immeasurable deflection. A similar length of aluminum tubing would bend further and risk folding over, and steel tubing would be far too heavy for conventional microphone stands. Therefore, it is ideal to use carbon fiber tubes for Decca Tree array construction. The only considerable weakness of carbon fiber is that it is still not quite as strong as steel. Holding a considerably heavy microphone, such as a Neumann M150 would probably

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43 The downward bending of a horizontal structure once suspended and weighted.
risk creating too much deflection compromising the integrity of the material. Also, one must use caution in using metal clamps on carbon fiber, since over-tightening can crack the material. Using rubber tipped clamps, or covering the metal in a few layers of gaffer’s tape is preferable over metal tipped clamps.

A good source for purchasing carbon fiber tubing is DragonPlate™. Not only do they carry a wide variety of carbon fiber products, but they also sell the necessary accessories that are crucial to this Decca Tree design. Other parts for holding the microphones to the frame of the design can be found by Impact Lighting, and one last part is actually by an outdoors equipment supplier called SHEENROAD. Finally, a few nuts and bolts from a local hardware store are required. There are two possible parts lists for this project, depending on the budget and functionality. Table 4.1a below is a parts list for an array that can extend microphones out from the center point up to 1.3 meters out, costing about $435. Table 4.1b below is a parts list for an array that can extend microphones out from the center point up to 80-85cm from the center point, costing about $250.

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<td>0.75” Female Clevis Connector</td>
<td>Dragon Plate</td>
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</tr>
<tr>
<td>Bolts for connectors</td>
<td>Black Anodized bolts and an allen wrench</td>
<td>Hardware store</td>
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<tr>
<td>Microphone Clamps</td>
<td>Bike Bicycle Handlebar Mount 1/4 screw</td>
<td>SHEENROAD</td>
<td>3</td>
</tr>
<tr>
<td>Thread adapter for clamps</td>
<td>Impact Female 1/4&quot;-20 to Male 3/8&quot;</td>
<td>Impact</td>
<td>3</td>
</tr>
<tr>
<td>Stand clamp for Array</td>
<td>Impact Atom Clamp w/ Ratchet Handle</td>
<td>Impact</td>
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Table 4.1a. A parts list for a carbon fiber Decca Tree array with a reach of up to 1.3 meters. See living parts list with links.
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<td>Bolts for connectors</td>
<td>Black Anodized bolts and allen wrench</td>
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<td>3</td>
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<td>Thread adapter for clamps</td>
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<td>3</td>
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<tr>
<td>Stand clamp for Array</td>
<td>Impact Atom Clamp w/ Ratchet Handle</td>
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Table 4.1b. A parts list for a carbon fiber Decca Tree array with a reach of 75-80cm. See living parts list with links.

To note, some of the items on these parts lists can be swapped out. For example, if different microphone clamps are desired, any others could be used as long as it can safely attach to a 0.75” rod. Take care to stick with rubber or plastic lined parts for these items to not compromise or damage the carbon fiber.

Once all of these items have been acquired depending on which size Decca Tree is desired, it is a matter of assembling everything together. Because of the modular nature of the parts, there is no cutting, bending, or machining required. Only light sanding and applying epoxy to bond parts together is necessary, using store bought epoxy in combination with the microfiber beads that are included with the Dragonplate™ parts. Follow the instructions for the epoxy, and assemble the parts.

This Decca Tree array design is above all meant to be small, modular, and fast to set up, because those are the first and foremost requirements for location recording engineers. Given that workflow is crucially important, the time required to set this array up was a crucial element in the design. Because of this, it only takes five minutes to set up the Decca Tree, from choosing the placement of the Decca Tree to raising it on a stand with microphones placed and cables neatly run. Some practice is required to
maintain a quick set up time, but that is the nature of all Decca Tree arrays and classical recording techniques in general.

The product is something an engineer can use for a wide variety of recordings. In addition to its popular use, it can be used on trios, piano quintets, voice with single instrument accompaniment, or anything else the engineer wants to use it on. Thankfully this design is relatively simple, so in the event that the application of it for a certain ensemble is not quite a perfect fit, it is certainly possible to tear down and replace with something else with minimal downtime. With this design, Decca Tree recording can actually be an entry level technique. It is often associated with high budget projects, as three omnidirectional\(^{44}\) microphones and a commercial Decca Tree array can be extremely expensive. This design is a challenge to that notion, and hopefully beginning engineers can experience this technique earlier in their careers.

4.2 Crocodile

The Williams Crocodile is a tool originally designed by Michael Williams as a means to identify an orchestra angle from any point to calculate a stereo recording angle (SRA)\(^ {45}\) while on location. His paper, “The Stereophonic Zoom,” is a monumental piece of audio literature that discusses the use and practicality of having a variable standard for stereo systems in recording. What this means is that for any

\(^{44}\) Omnidirectional is a polar pattern in which the microphone responds to sounds from all angles.
\(^{45}\) The stereo recording angle (SRA) of a stereo system is the width of the stereo image it will capture. It is usually designed by engineers to match the orchestra angle.
given scenario, a stereo microphone system can be calculated with little to no room for error. This is especially useful for recording on location. Most on-location recording sessions do not have a live monitoring system. Classical engineers use headphones to monitor on location, which is not at all accurate for evaluating a stereo image. A stereo image may sound usable on headphones, but correctly calibrated studio monitors may reveal a level of angular distortion in the stereo image. It is extremely difficult to detect this with only headphones. As well, many recording engineers do not always have the opportunity to set up their microphones before the ensemble’s rehearsal. They sometimes cannot set up, evaluate the stereo image, and then make corrections. For these scenarios, being able to accurately calculate a stereo system is a requirement.

Again, the Crocodile identifies the orchestra angle of any given ensemble from any given points. What this means, is that if one stands in front of an ensemble, he or she can measure exactly how wide the ensemble is from that point accurately within a few degrees. From there, several smart phone applications are available to calculate a suitable stereo system, such as Neumann’s Recording Angle Calculator, available for free. Once completed, the recording engineer is relatively safe in having a good stereo image. See figure 4.2a for a diagram of suitable stereo systems and their matching SRAs. Knowing this is a big advantage over engineers who do not do this, and adds a

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Angular distortion is the non-linear artificial widening or narrowing of a stereo image. For example, a quintet should sound evenly distributed across the speakers. But, angular distortion would either cause the three center players to sound bunched in the center of the stereo image, or the second and fourth player to sound too far left and right while the center player is too isolated in the center of the image.
considerable peace of mind. A practical use of this process will be discussed at length in the case studies.

Fig. 4.2a. An SRA diagram for cardioid microphones. The curved, diagonal lines represent the stereo recording angle of stereo systems along those points. The numbers in boxes along these lines are values of angular distortion (to be discussed later). The lower this number, the more accurate the stereo image. Williams, Michael. "The Stereophonic Zoom: A Practical Approach to Determining the Characteristics of a Spaced Pair of Directional Microphones." AES E-Library. The Audio Engineering Society, 1 Mar. 1984. Web. 13 Nov. 2014.

What it is, is a half circle viewer that one puts up to the eye, and looks through. It has the same fisheye eyehole that would be installed in a door peephole, which enables a larger field of view to see a wider angle. It has teeth, and each tooth represents an angle marker at every ten degrees. Every thirty degrees, the marker is either larger or longer for the convenience of the engineer. One points the center tooth the center of the ensemble, and looks at the degree markers on the edges of the ensemble. For example, if the edges of the ensemble are on the 45-47° mark, the
orchestra angle is approximately 90-94°. For this ORTF would be a great match. Go one step further, and purchase a simple digital rule/protractor tool. What this is, is two rulers joined at one end, where the you can create an angle with the two and read the exact angle on a screen. They are easily available and inexpensive online. With these tools in hand and some audio literature to back it up, any engineer can make great stereo images on a tight budget.

My take on this design was to simplify it, make it buildable with relatively weak woodworking skills, but still fully functional. There are two main ways to design it. The simple way is to keep it flat. The more complex design has an open mouth, with its top and bottom having an angled opening. This is harder to build, since it requires a drill press and a considerable amount of precision and sanding. This one in particular was actually built by Michael Williams himself. Here are the two different designs (figures 4.2b and 4.2c) and while the wide mouthed design is more complex, it is obviously more aesthetically pleasing.

Fig. 4.2b. A Crocodile angle viewer built by Michael Williams.
The functionality of both designs is almost entirely the same. The difference between the wide mouthed design and the flat design, is the wide mouthed design lets in a little more light, making it slightly easier to use in dark rooms. However, the flat design is substantially easier to build, and is recommended for anyone whose woodworking skills are anything less than professional. I wholeheartedly recommend trying to build one of these for anyone who is in their first few years of classical recording. Being able to use one while referencing Michael Williams’ *The Stereophonic Zoom* is a truly fantastic learning tool.

### 4.3 Unobtrusive Microphone Mounting Systems

There are a dozen ways to mount a pair of microphones in front of an ensemble. For example, imagine a string quartet with a soprano soloist. The easiest and most obvious way is to place a tall stand in front, and mount a pair of microphones on a stereo bar. For studio situations without an audience, this is
certainly okay and could sound beautiful. However, this is not ideal for live situations, and actually should be the last resort. A microphone stand in front of the ensemble should be avoided at all costs. The following information is a series of examples of solutions I have made to mount microphones in less visually obtrusive ways.

Many engineers hang microphones from wires or lines. This is probably most common for the permanent installation of microphones since anchors can be set into ceilings and walls with great strength. When using this as a location recording technique, several items are required. The engineer needs at least two anchor points (ideally three), strong fishing line, and something with which to attach the fishing line and a microphone mount. For the anchor points, woodworking style trigger clamps are great as long as the engineer stays under a few pounds for their complete setup. They are available in various sizes to be able to clamp to arches, pillars, bars, poles, and almost anything else found in standard venues. They can clamp with enough pressure to adequately support more than twenty times the weight of a microphone array for extended periods of time. They also come in black, which is rather important.

For the fishing line, I use 100lb black, braided Dyneema fishing line which is strong and does not stretch as much as standard fishing line. It can then be double, or quadruple run for added strength and rigidity, and is easy to tie reliable, conventional knots into. Other engineers use thin nylon rope, or even steel wires. Finally, for the piece I use to attach the microphones, carbon fiber is again perfect. From DragonPlate, one can buy a single 2’ section of 0.75” tubing, two female ends, and two eye hooks. Only light sanding and a five minute epoxy is required to assemble the parts, and any
engineer can suddenly have the perfect hanging spacebar. I find it perfect for attaching not only a stereo bar, but additional microphones such as a spaced omnidirectional pair for different sounding options in mixing. See table 4.3a for a comprehensive parts list and figure 4.3a for the finished product.

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</tbody>
</table>

Table 4.3a. A parts list for a hanging microphone mount. Multiple attachment clamps can be purchased for additional flexibility. These parts come to less than $75. See living parts list with links.

Fig. 4.3a. A custom built hanging bar for suspending microphones. This particular configuration features three microphone attachments to hold two pairs of microphones.
The biggest advantage of hanging microphones is that the stereo arrays can be far above anybody’s line of sight. The downside is that it takes extreme caution, care, and attention to detail to get right. Since time is everything, I find it takes at least fifteen to twenty minutes to properly set this up. It can take over thirty minutes if the engineer is inexperienced or has not developed a proper workflow. That amount of time wasted during set up is unacceptable. So if this is a method an engineer chooses to pursue, he or she should put a considerable amount of time toward practicing the suspension of microphones. An intelligent workflow, meaning a memorized set of steps that is easily replicated over and over, must be developed. Otherwise, the engineer will lose professional credibility while blundering about the stage before a performance, or even risk dropping their microphones from great heights.

A quicker and easier method is to create an attachment for microphones directly onto the aforementioned trigger clamps. With this method, an engineer can simply clamp a microphone array to anything a trigger clamp can safely attach to. One can also use an Impact or Manfrotto super clamp to attach to a truss, or other overhead bars if there are any available, which is fantastic for stages with fly rails. There is slightly less flexibility in placement with this, but there are workarounds.

First, if a trigger clamp is the desired tool to go about this, my recommended technique requires a bit of drilling. In my own designs I use the part of shock mounts that screws onto the microphone stand and holds the structure of the shock mount. With that, it is just a matter of getting the right nuts and bolts from a hardware store, then drilling through the bar of the trigger clamp to create the necessary attachment.
Once assembled, this can be implemented almost anywhere, and only coupling adapters and extensions are needed. From the Decca Tree design in the previous section and the carbon fiber hanging bar, two foot carbon fiber tube sections can be attached for additional downward extension. I have purchased a cheap, three section articulating arm from Impact that allows for over a meter of forward to backward extension. See figure 4.3b for the result.

Fig 4.3b. A modular hanging bar for microphones.
5. PRE-PRODUCTION

Pre-production is a relatively tricky concept to define from a classical location recording standpoint. Pre-production usually begins far later for location engineers than for other fields in audio. It technically begins from the moment a location classical engineer is booked to record, but almost never truly begins from that point (depending on how last minute one might be engaged). Realistically, if one were earning a living in this field, he or she would be in pre-production for dozens of concerts at a time, justifying the fact that classical location engineers start pre-production later than most other audio engineers.

Pre-production for recording classical music on location is different from any other genre or style of recording. For example, pre-production for an experimental rock album can be a daunting task. Demo tracks often must be recorded, the equipment can be auditioned, and producers might take the band out to a cabin in the woods to meditate over the compositions until they are ready to actually show up at the studio. For recording classical music in the studio, great engineers often test out different microphone systems, seating arrangements, and even pianos to ensure when they start actually recording they will get exactly what they want. At the other end of the spectrum, I would venture to say that pre-production for live classical recording, depending on how you define it, is almost always already in some state of completion.

At the risk of this sounding like a complaint, recording classical music on location affords few luxuries. In a live concert setting, sometimes it is possible to set up and record the dress rehearsal before the performance, but obviously this does not
always happen. Engineers often must set up only hours before the concert, before hearing any of the music that will be performed. Additionally, putting on a concert is often a last-minute process for stage managers and administrative personnel. While concerts are most often booked several months in advanced, seating arrangements and even decisions on where an orchestra might set up in a venue are constantly made only hours before the performance. Considering all of this, it is no wonder location classical engineers must make so many decisions “on the fly.”

Take a tracking sheet, for example, to narrow the scope. While I personally arrive at a venue with an idea of how I might record the ensembles performing in the concert, I know that my plans will almost always change. That does not mean that the original plan was either better or worse than the final recording system but one can never implement a cookie-cutter system before seeing the ensemble. A pew in a church might be just too close to the orchestra to place a microphone stand, the orchestra may be squeezed too far left and right for an effective use of a Decca Tree array, or a stage may be too small for any stands in general. Whatever system an engineer may have in their head, written down on paper, or set up in a DAW session will almost always change. Conversely, engineers in a studio recording setting often make a tracking sheet based on the instruments and musicians present. This is a wise tactic to use and allows them to explore several options before the band arrives. Luckily, in brick-and-mortar studio setting they can usually stick closely to that set up.

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A set of notes comprising of channel numbers, microphones, placements, and any other relevant information for recording.
This just is not something classical location engineers can plan very well for, so they should avoid wasting energy on a relatively lost cause.

Now consider the studio. Many location classical engineers are mostly packed up and ready to go at all times. As discussed in the previous mobile production studio design section, the recording studio is designed with mobility at its heart. It is also designed to be flexible enough to record a piano trio in the afternoon and then pack up to record a symphony in the evening. The engineer has the knowledge, experience, and literature behind them to be able to read a recording situation and develop a plan of recording on the spot. The mobile studio and the engineer are a package deal, and they move as one.

Classical location engineers are usually in some state between being packed up and unpacked. Personally, I have designed my studio to not need much packing. Other engineers have their own methods, but mine only consists of shutting down two power conditioners, unplugging the XLR cables going to my monitors, unplugging the FireWire 800 cable going to my computer, then latching up all my road cases. I have one case for my cables, adapters, mounting hardware, and tools. I have one case that holds my core gear (interfaces, preamps, and hard drive). My microphones are all in one box. My stands are always in two stand bags, as I always pack them back up after using them to record at home. I designed my studio to be able to pack up and go with the most ease, and it shows since my time between mixing with my gear at home to being packed up and driving away is often as low as fifteen minutes.
This is essentially all pre-production in classical location recording has to offer. Do not let that dismiss the process of pre-production as unimportant though. This is the last thing an engineer does before leaving for the recording session, meaning it is the last chance the engineer has to remember what he or she has forgotten, or anything of that nature. We have all forgotten something before a recording, and some engineers will forget many more things during their time in this profession. Driving off to pick up something such as a microphone clip from home or from the local music store appears unprofessional to the client, and wastes valuable time. My greatest wish in this section is to urge every engineer to dedicate a short moment of their time to pre-production before they leave. Packing up equipment is a routine task, during which many people start multitasking. Anything from contemplating an email that has yet to be written, to checking social media notifications can distract long enough to neglect noticing that a microphone was out of its proper case. Give yourself that crucial, often clairvoyant moment after packing to ensure you really have everything you need. It could be superstition or it could be entirely mad but it is good practice.

5.1 Advancing the Show

Advancing the show is an all-inclusive term for coordinating details before a show. This is the practice of communicating with whomever is organizing a concert, festival, or recording session to work out everything from hotel arrangements to microphone placement. This is a critical stage in the process of recording classical
music. These are obviously going to be some of the first interactions that the engineer has with their client, and first impressions are always important. Furthermore, most musical groups, orchestras, choirs, and symphonies are already used to operating, performing, and generally existing on their own without a recording engineer. The engineer is the outsider, and can even be seen as an invasive species to a stable environment to some. Clarity in communication, keeping technical information to a reasonable level, and politeness are going to be the most important skills the engineer has at this point.

With clarity of communication in mind, one can think of several ways to talk about the details of an event with a stage manager. While having a phone conversation is direct and fast, email is usually a better form of communication for this sort of planning as having details in writing is extremely important. Never decide on an arrival time, what sort of microphone setup will be implemented, or anything else of even minor importance without the ability to recall the decision in text. As an engineer, it is good to always be right, better to be able to prove you are right, but best to not have to prove you are right. A quick keyword search on a smart phone’s email archive can clear up any confusion in seconds, preventing any arguments on the behalf of poor communication. There are always exceptions to the preference of email, as with anything. For some, a routine recording project in accordance with a pre-scheduled event would fit this exception, or even a long-time client that the engineer can trust could warrant a simple phone call to work out some details. It is always at the
engineer’s discretion to decide what would work best, but email is always a smart starting point.

Technical information when communicating with classical music administrators is a tricky subject to broach. On one hand, information that is crucial to the production of a concert must be shared. There should be no surprises for the stage manager on the day of a concert. It is important for their own planning to know if the recording engineer will be suspending microphones over the musicians, placing a stand behind the conductor, or roughly how many spot microphones they plan on using. On the other hand, a music festival director might be sweating over how sixty musicians playing in forty different ensembles are going to rehearse before their concerts throughout the week, and they do not need to be bothered by the engineer’s explanation of Decca Tree measurements. The best tactic in this stage is to gauge the level of technical knowledge the person in charge of the communicating with the engineer has, or how much they care about the technical details as fast as possible. Perhaps a brief phone conversation would be ideal to do this, or “feel it out” in an early email. Writing in an overly technical fashion will go over their head, but simplifying too much can easily be misinterpreted as demeaning and rude.

There is a good compromise for this situation that works more often than not, which I use quite often (while not causing problems if it fails). It is essentially a recipe to figure out the client’s level of audio acuity. Write something relatively short, and easy to understand. Use relatively common, but unnecessarily vague terms, and do not be afraid to over-simplify just yet. Then, follow it immediately with a simple,
friendly apology, reminding the client that their understanding for audio is not being assumed to be any less than anyone else’s. For example, write a statement like: “I was thinking that a spaced pair of omnidirectional microphones, one placed near the principle violinist and one placed near the principle cellist, about two meters in front of the orchestra might yield a nice, open sound. Then, a stereo pair of more directional microphones in front of the woodwinds, and finally a spot on the bass section would help fill in more detail and fullness. And, my biggest apologies if I am simplifying this too much!” Your client’s response to this will reveal everything you need to know. Yes, it is manipulative to an extent, but as long as you are polite in the rest of the email it is harmless and usually works.

Furthermore, a note on politeness while advancing the show. Consider first, who has what at stake. If the engineer is communicating with the stage manager, the stage manager has the artistic direction of the conductor at the top of their priorities. Many conductors are full artistic directors, and if they express this to their stage manager once, extending even one microphone over an orchestra during a live Mozart performance might come across as gut wrenching. This would obviously be problematic to the recording process, especially with limited equipment. To repeat, the engineer is often the outsider, and this sort of situation is often a losing battle. The best thing to do is remain calm and do not argue, but just make sure every person involved is on the same page. Above all else, demystify the process before any decisions are made.
As an anecdote on the subject of situations that turn problematic while advancing the show, I recently completed recording with a symphony orchestra in Ohio, where this was the exactly problem at hand. After securing the job with the executive director for the orchestra, I was then given the contact information for the stage manager, who would be the point person for communication. After emailing her with a rough idea for the recording involving a Decca Tree array between the conductor and the front line of violins and violas, she immediately called the number in my email signature and vehemently went on about how problematic this would be. Without going into further detail about the conversation, I chose to tell her I was busy, and would have to email her from the car while on the road. This was only a half truth, but I felt it was important to have the rest of the communications in writing, and the situation could be dramatically improved if it proceeded slightly slower. With politeness, clarity, and tactical wording, the situation improved drastically. I was able to convince her to speak with the conductor herself about the proposed ideas, and I was given full control over the microphone setup from then on. The recording then was a fantastic experience, and my interactions with the stage manager are now fully, and genuinely friendly.

Finally, in communicating with classical musicians, be sure to have a decent understanding of music etymology. Obviously, pronouncing the composer Chopin “chop-in” instead of “cho-pan” within fifty meters of a classical musician is a sure way to destroy your credibility for years to come. A quick Google search can help clear any confusion before having to make any query to a musician or concert
administrator, so there is no excuse for a blatantly wrong mispronunciation in today’s music industry. Additionally, being able to identify a Scherzo movement from a Prelude without having heard a particular piece before takes some considerable practice, but is a good skill to have. Not many recording engineers have equal classical music education backgrounds to the men and women they record, but being able to have a working vocabulary is useful. Engineers must be able to speak the same language as classical musicians at this stage of recording.

5.2 Listening Before the Show

This is a good time to take note of one last thing location classical engineers can do to prepare themselves for a recording. Listen deeply. There is a certain kind of pressure involved in the process of recording. When a larger ensemble, such as a symphony is involved, one often feels the eyes of all eighty musicians at the back of his or her head. Some see the engineer as an intruder to the creative process and are skeptical as discussed previously. Some are hopeful, simply excited to hear a recording of themselves and their excellent peers. Many younger engineers can find this daunting. Instead, I find this competitive. If I can get my hands on a program prior to the concert, I spend the last few days before the concert incessantly listening to different recordings of the pieces that will be played. The competition is not against the musicians or myself, but against the often faceless engineers of those recordings. It is ultimately the desire to produce something that will stand amongst, or outshine professionally recognized recordings that can drive a young engineer to success.
This is the almost ethereal process of memorizing the timbre, the ambience, the stereo representation, and even the fade ins and outs of previous recordings of pieces that the engineer is about to record. It is a healthy obsession. Listening to several different renditions of recordings helps create a sum of several different engineers’ ideas and aims. This may still be incorrect, or at least different than any particular engineer’s own vision, but it leaves a distinct imprint that cannot be disturbed by the engineer’s first perceptions of an ensemble through their headphones on location. This imprint is crucial. When most engineers sets up their microphones, and first monitor their work through headphones on location, their psychoacoustic analysis of what they are hearing is often unjust and corrupted by their own perception. There is nothing to draw upon to determine if what they are hearing is correct, or if they must make any changes in their microphone techniques, because there is no concrete standard to how a recording should sound.

This obsession to have a perfect, memorized concept of how the recording should sound must be delicately constructed. After all, a misconstrued perception of classical recording can lead to a recording that is less than perfect. Even a balance between low and high frequencies that is off by two or three decibels\(^48\) can sound too bright or too dull. Listen to something that is in the same vein as the music that will be recorded meticulously, and memorize that balance. It is difficult, not unlike possessing perfect pitch, and it will never be perfect. But the more in-tune with the ideal perception of a recorded piece the engineer is, the better the engineer can make

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\(^{48}\) The Decibel is a logarithmic measurement of sound. In this case, decibel refers to volume (Bartlett, 50).
judgments while recording on location. After all, no engineer can claim to make great recordings if they do not know exactly what the music should sound like.
6. NOTES ON RECORDING, MIXING, AND DELIVERY

There are many concepts, practices, and habits I have formed that are worth mentioning before moving into the case studies. They have been derived from years of reading literature, insight from mentors and more experienced friends in the industry, and ultimately trial and error. Some are important reminders to bear in mind each time one approaches a recording, and others are routine practices that often get forgotten. Either way, they are a culmination of the skills I have developed and the reader should keep in mind that reiterating them again and again in the case studies would grow tiresome. The case studies section will skip these details, to spare the reader the habitual, often neurotic way I set out to record live classical music.

Understand your ears, and your hearing. While it is relatively common knowledge that auditory fatigue\(^4^9\) and hearing damage are usually results of long exposure times instead of simply loud noises, there is much more at play.\(^5^0\) Because of this, avoid mixing for hours on end the afternoon before a recording session, and keep music quiet during long drives to the venue. I find listening to podcasts and live comedy to be a good substitute to music that is easier on the ears. Additionally,

\(^{4^9}\) Auditory fatigue (also known as hearing fatigue or temporary hearing loss) is a short-term symptom of hearing damage. After listening to loud noises over a period of time, engineers usually find their ability to discern sonic detail to be less keen.

\(^{5^0}\) While hearing loss and fatigue are the result of listening to loud noises, the exposure time is just as relevant to the volume of the noises. Sounds in excess of 80dB may not cause temporary hearing loss if a person is exposed for less than a minute, but will increasingly cause temporary hearing loss the longer the person is exposed (Hirsh).
caffeine, certain medications, and other ototoxic chemicals\textsuperscript{51} can alter hearing perception (Mujica-Mota). Even the common cold may greatly affect hearing by temporarily congesting the inner ear. This is just one more reason to stay healthy, and be careful of what substances you put in your body (Cruz, 35)!

Neatness counts: The way the engineer runs their cables, sets up microphone stands, sets up their computer station, or even readies their cases to load out are all reflections of his or herself. Excess cable should always be coiled neatly under the microphone stand or by your input section (probably by the stand, giving yourself a few service loops to adjust mic positioning this way). Cable that are run across walkways should then always be taped down with gaffer’s tape\textsuperscript{52}, and important stage points should be marked intelligently with spike tape\textsuperscript{53}. Stands should be staged off to the side of your workspace when not in use, neatly in a line. This train of thought could go on and on. Neatness requires foresight, but engineers are smart and should be able to grasp it. Ultimately it saves time, and makes the engineer look much better.

Do not settle for just having gaffer’s tape, have several kinds of gaffer’s tape. Personally, I have a bungee cord and carabiner that I use as a belt clip to carry tape when I need it, and I keep 2” black tape, 1” black tape, and at least three different colors of ½” spike tape. In the absence of having a marker, using three different colors

\textsuperscript{51} Ototoxic chemicals are harmful to the ear.

\textsuperscript{52} Gaffer’s tape is a reliable tape for all production work that is matte in color and leaves no residue.

\textsuperscript{53} Spike tape is similar to gaffer’s tape in nature, but is usually thinner and made in bright, often neon colors to be more visible.
of spike tape on three different cables can help keep Decca Tree channels straight. Colorful spike tape is also perfect for helping legs of a microphone stand remain visible in low light to prevent tripping. Even a strip of orange tape over an already taped down cable run can help prevent someone from stumbling. There are endless reasons for having different kinds of tape, so take advantage of it.

Measure the distance between your main microphones and your spot microphones. There are several opinions that exist between different recording professionals, but compensating for delay between your microphones will yield a more realistic sounding recording. This is called delay compensation.\(^{54}\) Since sound takes time to travel between two points, this means that microphones placed at different distances from the sound source can muddy the final sound because it negates the way sound naturally travels. For two microphones, measure the distance between them, and divide that distance by the speed of sound. The output of that equation is the time that it takes for sound to travel that distance, which is also the delay time you must set on the microphone that is closer to the sound source.

Sandbags are not optional. Professionals of all kinds use sandbags to stabilize stands holding important objects, and audio engineers are elusively the pinnacle of this practice, or in some cases, this malpractice. Grips on movie sets put big steel lights on tall stands with sandbags at the base, construction workers put sandbags on temporary signs so they do not get blown over by the wind, but audio engineers put irreplaceable vintage microphones on flimsy, wobbly aluminum stands. Worse yet, we often place

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\(^{54}\) Delay compensation is the act of correcting time of arrival differences between microphones, since sound travels relatively slow.
these inadequate stands in front of famous, delicate, even elderly classical musicians. This is not a risk that can be taken, especially when sandbags are not a major expense.

When placing a microphone near, or pointing toward a musician, be friendly and say “hello.” It is extremely important that the musician is comfortable and does not feel that the microphone is intruding on their space, so do everything to keep them happy, within reason. I find it best to briefly introduce myself, explain that I will be placing a microphone within their vicinity, and ask them to let me know if there is anything I can do to ensure that it will not be obtrusive. Any lack of politeness in this process is sure to cause anxiety for the performer, or even anger. For example, I have never seen a musician become so angry so quickly as when I witnessed a sound reinforcement engineer shove a microphone inside a french horn player’s bell as he was playing it. This is unacceptable, and the musician had every right to be upset.

Understand standard protocol during a live performance. This means that there is a chain of command, and that certain tasks must be completed by certain people. There are people that the engineer is expected to communicate with, and there are people that it would be unwise to communicate with. It is not unlike working at a theater or on a film set. For example, be sure to talk to the stage manager before going directly to the conductor with a question. As important as the recording engineer is to the concert, there are boundaries. Seasoned engineers can read any room, and determine how to best handle these situations.

Check where any sort of emergency exit routes may lead, and do not set up in them. This means if there are fewer than four exits out of the venue, recording
engineers cannot set up in front of, or in any of those hallways outside of the main performance room. I would not mention this if it had not been a serious nuisance several times in the past. Be careful of where you set up, or else a building supervisor will inevitably ask you to move- a task most disappointing after set up is complete.

    When mixing, cut out distractions. This advice is especially aimed toward younger engineers that often fall victim to being overly anchored to their smart phone. Silence the phone, turn off notifications on the computer, lock the door to the mixing room, and even give the dog something to chew on. Mixing is inarguably a creative endeavor, even if the project is classical music that may require no equalization, compression, or anything further than balancing track volumes. Only once the engineer totally immerses himself or herself in the mix, can they make the best and wisest decisions about the music.

    Understand compression\textsuperscript{55} and equalization\textsuperscript{56} on a theoretical level before attempting to mix classical music. Arguably, there is room for error in this for other genres, such as rock, hip-hop or electronic music, because dynamics processing\textsuperscript{57} is much easier to hear in these scenarios. Conversely, I believe it is very difficult to hear how compression and limiting affects classical music until it is compressed so hard that major problems arise. It is also difficult to hear the negative effects that

\textsuperscript{55} Audio compression is the process of restricting the dynamic range of a sound.

\textsuperscript{56} Equalization is boosting or attenuating certain frequencies of a sound to change its timbre.

\textsuperscript{57} Dynamics processing is anything that affects dynamics in audio. This is compression, equalization, de-essing, gating, etc.
equalization has on classical music, such as coloration from phase shifting.\textsuperscript{58} For novice engineers, I recommend becoming as familiar as possible with these processes before mixing classical music.

Be careful of colloquialisms when communicating with the client. If the client requests for the recording to sound more warm, ask nicely for something more specific. “Warm” can actually mean distorted, lacking high-end, or any number of other sound qualities because it is too imaginative of a term. Audio engineers have been thrust into an industry full of colloquialisms so there is no escaping them all together, but one can always muster the vocabulary to express ideas in a more accurate manner with a little thought.

\textsuperscript{58} Phase shift is the fundamental way that most equalizers work. This is the shifting of the time alignment of sound waves, then adding them back to the unaltered sound waves, which causes certain frequencies to drop out. This does not actually create a linear change in what the equalizer is affecting (Winer).
7. CASE STUDIES

The following case studies are in accordance to the supplemental recordings with this thesis (see appendix A, or download here). They relate my personal experience working in classical music as a young recording engineer and while I believe there is always room for improvement and thoughts over my own process, I am proud of my work. The techniques, though processes, and stylistic choices I made are the culmination the last several years of my education and exploration in classical production. In no way do I see these four case studies and the methods I used to record them a complete representation of how classical music should be recorded but perhaps they could serve as inspiration for using improvised techniques for those just starting in this industry as I am.

All of these recordings are from one single week of recording the Lake George Music Festival from August 14th to the 21st of 2014. This is a classical music festival that highlights young musicians, and is run by relatively young music entrepreneurs. The vast majority of the musicians are recent Ivy League graduates, doctoral students, and other talented young musicians interspersed with seasoned principle musicians from well regarded symphonies across the eastern United States. This is the fourth year the festival has run, and it has quickly proven itself as a rapidly expanding festival in the classical music community of New York and its surrounding states. My role in the festival was to be the sole recording and production person. I ended up with about ten hours of recorded material from the festival, some of which has found its
way to NPR, and to a CD sold in the New York area. It turned out to be one of the best experiences I have had in my career so far.

7.1 Case Study One

*Copland, Appalachian Spring Suite – Variations on Shaker Hymn (Simple Gifts)*

As someone who unabashedly picks favorites in all of my interests, recording this piece was definitely one of my favorite moments in recording thus far. Interestingly, the program notes for this piece point out that Aaron Copland’s inspiration for writing the piece had nothing to do with Spring in Appalachia where I currently reside, though the believability of the music’s imagery for those who do not know this fact is uncanny. The excerpt included is only the seventh of eight movements, and I recommend listening through all eight uninterrupted if time should allow for the reader.

This piece was originally written as a ballet for a thirteen piece orchestra. It includes a double string quartet, double bass, clarinet, flute, bassoon, and a piano. The double string quartet is arranged in a semi circle around the conductor with the woodwinds centered behind them, and the bass was positioned far right, and the piano was at half stick on the left. Musically it is relatively dynamic, and is known for moving the melody repeatedly to different instruments while using unique harmonies.

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59 A double string quartet is an ensemble where every instrument of a standard string quartet is doubled, meaning four violins, two violas, and two cellos.

60 A position of the piano lid that is halfway between being completely open and closed. This quiets the piano considerably, while preserving its openess and timbre.
Additionally, Copland only wrote the bass to play sparingly throughout the twenty-three minute piece, which creates a moving frequency range that extends to sub 150Hz notes only during emphasized passages in the piece. It is tastefully done, and I used this to a creative effect in recording.

The room that the piece was recorded in was the Sacred Heart Catholic Church in Lake George, New York. It is a new church that was designed with acoustics as a priority. With the ensemble performing on the floor between the altar and the pews, the side walls were extremely far, and the back wall (the wall behind the audience) was still far but had a ten degree angle either way to prevent direct reflections. The ceiling then was angled upward toward the back of the church, and also at least fifteen meters high. Due to the layout of the musicians, my initial reaction was that a relatively tight Decca Tree configuration would be a good match. This is for several reasons. First, I chose Decca Tree because omnidirectional microphones would be good to pick up the natural reverberance of the church. Of course this would also bring out audience noise, but it was a willing sacrifice for the sake of the music.

Secondly, the way stereo image was an important factor. A stereo system, such as one configured with the principles in Michael William’s *The Stereophonic Zoom* would still not be perfect. No stereo array would accurately represent the chamber orchestra if placed in front of the conductor, and any stereo array placed behind the conductor would have left the woodwinds in the back row sounding too distant. Decca Tree allowed for an accurate representation of the stereo field, while providing a great
direct to ambient sound ratio.\textsuperscript{61} I also knew I would have a significant amount of control over the strength of the woodwinds in mixing with Decca Tree. Finally, it only required me to place one stand, ensuring minimal aesthetic interruption of the performance as the ceilings were far too high to attach anything, and side walls were over 25 meters away making suspending the microphone array impossible. The Decca Tree array was placed on the single stand between the conductor and the double string quartet in the front line, just shy of three meters up.

Additionally, I used cardioid spot microphones\textsuperscript{62} on the piano and on the bass. The piano spot was placed on a short stand at the edge of the lid, between the piano body and the lid, pointed at the midrange of the piano. A stereo microphone system on the piano would have been problematic for this effect, since the orientation of the chamber orchestra would not have allowed the piano to be anywhere but left in the stereo image. The bass spot was placed fifty to sixty centimeters from the instrument, pointing between the strings and the f-hole, and slightly above where the bassist was bowing the instrument. While experimenting with placements during the ensembles rehearsal, this seemed to be the most natural sound for both instruments, while not having a high aesthetic impact on the performance (Woszczyk, 80).

Finally, I placed an ambient pair of microphones in modified ORTF, with a slightly smaller angle than 110 degrees, far away from the ensemble. I placed it two

\textsuperscript{61} The direct to ambient sound ratio is the amount of sound reaching the microphones directly from the instruments versus the amount of sound coming from other directions after the instrument’s sound develops into the space.

\textsuperscript{62} A spot microphone is a microphone that only supplements a main system of microphones by being much closer to the instrument. They usually do not provide the main sound for the recording.
two meters in front of the back wall that had the wall protrusion facing left and right to further prevent direct reflections. The ORTF array had nothing to do with the orchestra angle as explained in the Crocodile chapter, as any accurate stereo system according to the real, very narrow orchestra angle would have to be extremely wide to compensate, because wider stereo systems capture smaller center images. ORTF simply was a go-to technique, and modifying it to be slightly more narrow gave me slightly more direct sound, but a wide image that I could use subtly in the mix. Here is a full tracking sheet (see table 7.1a), and an overhead diagram (see figure 7.1a) just for reference.

<table>
<thead>
<tr>
<th>Channel</th>
<th>Placement</th>
<th>Microphone</th>
<th>Preamp</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Bass Spot</td>
<td>Rode NT5 with MJE Capsule</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>2</td>
<td>Piano Spot</td>
<td>Rode NT5 with MJE Capsule</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>3</td>
<td>Decca Tree Left</td>
<td>Modded Oktava MK-012</td>
<td>Grace M101</td>
</tr>
<tr>
<td>4</td>
<td>Decca Tree Right</td>
<td>Modded Oktava MK-012</td>
<td>Grace M101</td>
</tr>
<tr>
<td>5</td>
<td>Decca Tree Center</td>
<td>Oktava MK-012</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>7</td>
<td>Far ORTF Left</td>
<td>AKG 414 ULS</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>8</td>
<td>Far ORTF Right</td>
<td>AKG 414 ULS</td>
<td>MOTU Interface</td>
</tr>
</tbody>
</table>

Table 7.1a. A tracking sheet for case study 6.1.
Two notes must be mentioned in accordance to the diagrams. First, this is not an ideal Decca Tree setup. Decca Tree is originally designed to have three identical microphones, and three identical preamps. Note that the microphones are close to matching, but are slightly off, and the center channel is using a different preamp. However, in an improvised, budgeted setting, this setup still functions and gives the desired stereo image even though it does not traditionally fit the description. Secondly, the keen eye will notice the absence of channel six in this table. This is intentional, but only because Pro Tools only allows stereo tracks to be started on an odd numbered track. Place no importance on this decision, other than the desire to have a faster workflow within the software. This will be consistent throughout the rest of these case studies.

Also note that this placement of a Decca Tree array is slightly unique. Traditionally, Decca Tree systems are described as being placed a few feet behind the
conductor, rather than just in front. I think of this placement as more modern, both in how this disobeys the tradition and how it affects the sound sonically. Instead of being more reverberant as the audience might experience the music, it adds far more direct sound and separates each instrument much better. It can detrimentally affect the stereo image, so much care must be taken to ensure that the ensemble is a good match for this configuration, as was the case here.

Due to the distance between the Decca Tree array and the spot microphones, measurements were taken and time delays were calculated on the spot to perform delay compensation. The piano was roughly 380cm from the center point of the Decca Tree, and the bass was roughly 360cm from the center point of the Decca tree. Since Pro Tools 11 does not have a stock delay plugin that operates in milliseconds, calculations were done to convert centimeters to samples. This means dividing 380cm from the speed of sound at approximately 34,300cm/second, then multiplying that times 48,000 samples/second, which was the sample rate of the session. See figure 7.1b for the calculation.

\[
\left( \frac{380}{34300} \right) \times 48000 \approx 531.77843
\]

Fig. 7.1b. A delay compensation calculation for case study 7.1

A delay of 532 samples is then placed on the piano track. Furthermore, a delay according to the 360cm distance on the bass was added, which is 504 samples. This
calculates to a difference of only 28 samples, but in this case is still important to maintain correct time of arrivals in mixing.

Mixing this piece required a combination of modern and traditional standards. From a traditional standpoint, no automation or compression was done on the main three Decca Tree tracks. This is to preserve the dynamics that the performers played, and to prevent any audible change in the noise floor. As per the standard, the left and right channels were panned hard left and right, while the center channel was kept center. The center channel was used in the mix just enough to have a solid representation of the center image, while not compromising the stereo feel of the piece. For equalization, a 1.5dB boost between 2kHz and 8kHz was used to enhance the high end of the recording.

The piano spot was mixed in only enough to add a slight amount of closeness to its sound. It was panned to the left, matching the localization of it in the Decca Tree array. A high pass filter was used on it, around 120Hz at 12dB/octave, to remove any perceived muddiness from the mix, as well as preventing unwanted low end energy from being too far from the center image.

The processing on the bass spot microphone is considerably less traditional. In Pro Tools, it first panned right to match the localization in the Decca Tree array, then bussed to a mono aux send that remained center. The return of the aux send was the main submix\textsuperscript{63} with the rest of the tracks, making this an example of parallel summing.

\textsuperscript{63} The submix of a mix is the track that all other tracks are sent to. This process is also known as summing.
compression. Then, multiband compression was used to restrict the dynamic range of everything below 100Hz. It was compressed to have up to 6dB of reduction on the loudest parts. A low pass filter around 800Hz was then used to remove most of the string and bowing noise from the track. Finally, the aux send was only mixed in enough to have a noticeable presence when the bass was playing. This processing on the track creates a thick, consistent low end in the center image whenever the bass is playing without protruding out of the mix unnecessarily. Furthermore, the low pass filter still protects the stereo image, since the articulation of the bass is still panned right to where the bass is supposed to be.

This is not a mixing technique that many seasoned engineers will recommend. It is counterintuitive to preserving the dynamics of the performance, and admittedly is not the most accurate way to preserve the music from the exactly how it sounded from the performance. However, it does create a more full sound that can be considered an improvement over many traditional techniques used to fill out the low end in classical recording. This is an example of picking out subtleties in the music, and enhancing the sound so that every listener can enjoy a heightened atmosphere of the performance.

Finally, the ambient ORTF pair was mixed in just enough to enhance the reverb trails slightly. Such a distant pair can raise the noise floor to an unacceptable level.

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64 Parallel compression is combining an uncompressed track with a compressed version of the same audio.

65 Multiband compression is having independent control of the compression of different frequency ranges.

66 Dynamic range is the difference between the quietest and loudest parts of any sound.
level, as well as bring out unwanted audience noise, so it was used only subtly. This way, it does not appear to add any more space to the recording, but only make the space that is already there slightly more distinct. However honestly, muting this track from the final mix would not have made much of a difference!

The overall mix is true to the musician’s performance of the piece, but enhanced slightly to create a slightly larger than life sound. Conversely, a stereo pair in front of the ensemble can sometimes create a more realistic representation of the music, such as being a depiction of what it might sound like to stand before the ensemble during the performance. However, in a field where the most impressive recording catches more peoples’ attention, this was the route I chose. During the variation of Simple Gifts that is the loudest and most powerful, this recording system enhances that power and delivers a sound that can move the listener emotionally. And, to sum up several of the previous sections about planning for recording, this complete system was totally devised during the rehearsal of the piece. Thinking on your feet and creativity combined can do very cool things.

### 7.2 Case Study Two

*Ibert, 3 Pièces Brèves - Allegro*

When this recording was featured on American Public Media’s Performance Today, Fred Child used these words to describe it. “The music is so full of character and color, and the title is as bland as can be.” It is written for a wind quintet, and for this performance it featured a flute, oboe, horn, bassoon, and clarinet from left to right.
Other descriptions of the music include testaments to its liveliness, while noting how clearly the timbre of every melody can come through. Admittedly, this knowledge did not come off hand during the festival, and some Google research was required to learn more about this piece. Before online research was possible through smart phones, classical engineers probably could never have learned all of this prior to the concert. But this is a huge advantage for engineers today, because these details are so attainable, and the engineer can make better recording decisions as a result. My findings in that quick research were the primary reason I chose to use only a stereo pair to record this piece.

Why does lively music, and clarity on the musician’s part mean a stereo pair would be sufficient? Simply, my opinion on this matter is that if the ensemble can project well and distinguish its unique sounds on its own, overdoing the production can seriously draw away from what they are accomplishing. Lively, to me, means that the musicians are dynamic in their musical expression, but quite simply are just loud enough. They are loud enough to overcome audience and room noise to make a cleaner sounding recording, which is great. When this is not the case, additional spot microphones might be necessary to get more direct sound. Then, clarity is the ability of the ensemble to make each part distinguishable from the next. More microphones means less phase coherence,67 and more responsibility from the recording engineer to balance the quintet (Woszczyk, 92). In this case, the ensemble did such a good job of

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67 Having more phase coherence means that the level of signal loss caused by of time of arrival differences, and angular differences between microphones is minimal.
accomplishing all of this on their own that a single stereo pair was enough to make the recording work.

Principles from *the Stereophonic Zoom* were implemented to set up the stereo pair. The process for this first involves determining a point at which the stereo pair will be placed. Then, the engineer must peer through the crocodile angle viewer, and determine the orchestra angle, from the musicians on the furthest left and furthest right. The orchestra angle in this instance was 102 degrees. While the math to calculate a stereo system in this scenario is quite difficult, there is a free smartphone application by Neumann called the Recording Angle Calculator that can easily do the work for you. Any engineer can simply move the microphones on the display around until an appropriate stereo system is achieved. A microphone angle of 90 degrees at the 20cm separation was found to be ideal on the application. See figure 7.2a, a screenshot from the application.
Note that the two vertical bars on the left and right of the display are referring to the ratio between time and level differences. These are important to keep in mind, because a stereo system with a relatively even ratio will usually have less angular distortion. The stereo pair was then placed on a combination boom stand, and reached over some of the steps to have a roughly thirty degree downward angle pointing toward the quintet. This is primarily to capture a better timbre from the ensemble, because being too low or too high can often sound too muddy or have strange stereo images (see figure 6.2b).

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68 Time and level differences are the two properties that stereophonic recording operates by. Time difference is the time it takes for sound to travel past the first microphone to the second, and level difference is the difference in amplitude between the two microphones. These differences combined create stereo perception.

69 Timbre is the character and texture of a sound source or recording.
Due to the layout of the venue, the recording turned out to be quite dry. The floors were thickly carpeted, and the ceiling was only moderately high. One could solve this by placing the microphone a meter or two further back, to capture more of the church’s ambience outside of the altar area, but there were several other variables at work. It was a hot day, some windows were open and there were four fans in the room. The comfort of the audience was not second to the recording quality this day, so the sacrifice had to be made.

Fig. 7.2b. An overhead diagram of the ensemble and microphones in case study 7.2.
Mixing this piece was relatively minimalistic. The Pro Tools session was as simple as a master fader, a stereo audio track, and a stereo auxiliary bus. I simply put the stereo audio track at a level that would put the loudest notes of the piece between -1.0dBfs and -0.5dBfs, without any compression or limiting. The musicians played beautifully, with expression, dynamics, and emotion. Any dynamics processing that would have negatively counteracted this. A high pass filter was then used, only at 45-50Hz or so, to prevent any rumble from air circulating through the room.

One unique, arguable improvised technique was used however. This was the act of automating the high pass filter between movements and during fades. This means that the frequency of the high pass filter changes according to what I want it to do. Sometimes fade-ins can be slightly tarnished due to a truck passing on the road adjacent the venue, or the noise floor is simply distracting on its own. What I do in this situation is automate the frequency of the high pass filter to coincide with the fade in, starting around 300Hz, and dropping to its resting point of the aforementioned 45-50Hz. by the time the fade in is completed. The same thing is then done for the fade out, or sometimes during a cadenza in the music. All this does, is help the noise floor remain unnoticed whenever it has the potential to become obtrusive.

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70 In audio consoles and Digital Audio Workstations, the master fader is the final metering point and volume control.

71 A stereo audio track is a track within a Digital Audio Workstation that can record two tracks of audio.

72 A stereo auxiliary bus is a track within a Digital Audio Workstation that does not carry audio clips, but serves to route and process audio.
Finally, a light reverb\textsuperscript{73} was added in mixing. The executive director of the music festival first inquired on this subject and after a quick demo of the dry recordings. Although it would not be truly representative of the space the music was performed in, it would enhance the music. Reverberation is after all a pleasing sound to most listeners. The room in this case was dry enough that early reflections to the microphones were not too powerful to make the reverb sound awkward or out of place.

7.3 Case Study Three

\textit{Ludwig, Virtuosity, Five Microconcertos for String Orchestra – Concerto for two Violins}

David Ludwig is a composer from Philadelphia, and is a resident composer for the Lake George Music Festival. His piece, \textit{Virtuosity}, is a modern and thrilling sounding classical piece written for a sixteen piece string orchestra. It is in five movements, in which the first four feature a particular instrument in a brief concerto while the final movement features the orchestra as a whole. One could certainly describe this is a modern sounding classical piece with many strong, dark, and emotional themes throughout. Only the first movement, the concerto for two violins, is included for this case study, which features one first violin and one second violin (which interestingly are on opposite sides of the orchestra). The concert took place again in the Sacred Heart Catholic Church of Lake George, which as previously described is a large, richly ambient room.

\textsuperscript{73} Reverb, short for reverberation, is the prolonging or resonance of a sound.
Since this was the second concert being performed in this location, a different microphone setup was chosen. At the time, I had reflected on the recording of Copland’s *Appalachian Spring Suite* and decided that while the Decca Tree array had a nice timbre and an excellent sense of space, the omnidirectional microphones captured a slight bit more audience noise than was acceptable. Therefore, a cardioid stereo system was chosen with the intention of rejecting as much extraneous noise as possible using the rear null points\(^\text{74}\) of the microphones.

The layout of the chamber orchestra was a relatively simple one. There were four first violins, four second violins, three violas, three cellos, and two basses. They were not arranged traditionally with high instruments on the left and low instruments on the right, but rather with the violins on the outsides, and basses in the center. They were relatively cramped on the stage area in the church, but this is natural for concerts not at traditional venues. See figure 6.3a for an overhead arrangement diagram.

\(^{74}\) A null point is an area of a microphone’s polar pattern in which the microphone theoretically should not respond.
For microphones, only two stereo pairs were used. This is not necessarily a choice for this specific piece, but a choice for the entire concert. Other pieces featured a piano trio, a piano quintet, and a piece for voice with piano accompaniment. For the whole concert a main stereo pair was used, with a supplemental stereo pair for the piano spot and the basses in this particular piece. The supplemental pair was only mixed in subtly for the rest of the concert, but was mixed differently for *Virtuosity*, which will be further discussed (see table 7.3b).

<table>
<thead>
<tr>
<th>Channel</th>
<th>Placement</th>
<th>Microphone</th>
<th>Preamp</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Bass Spot Left</td>
<td>Rode NT5 with MJE Capsule</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td></td>
<td>Bass Spot Right</td>
<td>Rode NT5 with MJE Capsule</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>---</td>
<td>----------------</td>
<td>---------------------------</td>
<td>---------------</td>
</tr>
<tr>
<td>2</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>Main Pair Left</td>
<td>Modded Oktava MK-012</td>
<td>Grace M101</td>
</tr>
<tr>
<td>4</td>
<td>Main Pair Right</td>
<td>Modded Oktava MK-012</td>
<td>Grace M101</td>
</tr>
</tbody>
</table>

Table 6.3a. A tracking sheet for case study 7.3

Since this piece was at the end of the concert, this required me to come on stage once between pieces to place the microphones where the basses would be, but this was approved by the stage manager and the executive director of the music festival. Spike take was placed during the rehearsal of the piece, marking where the front row of musicians, the basses, the piano, and the supplemental pair of microphones would be placed during the concert. This is a useful tactic when dealing with placement and arrangement changes during the concert, and helps out the stage manager very much so there is no confusion. This is also an additional testament that it is important to dress in black, and be presentable for live performances in the event that one must ever be on stage.

The concert went well, and in mixing not much was done to alter the sound of the recording. First, after the piece was edited with proper fades, delay compensation was done. A delay of 644 samples was added to the bass spot microphones, to reflect a 4.6 meter distance between the two pairs and prevent any time of arrival issues. Next my attention was turned to the main pair of microphones, and helping them to sound as natural as possible. A high pass filter at 40Hz was added, only to prevent any unwanted rumble, which is sometimes a result of the lack of shockmounts on the microphones. Additionally, this is important because excess energy in sub-bass
frequencies\textsuperscript{75} (typically caused by excess air circulation or accidentally kicking the microphone stand) can negatively impact higher frequencies by overloading the preamplifier circuit within a microphone. Even though these sounds are extremely low, they distort the entire microphones output, and this cannot be fixed in mixing. A high frequency shelf, boosting frequencies above 4kHz was added to add more brightness and air to the recording. No compression, or any other further processing was done.

The bass spot microphones, were treated in the same way the bass spot was processed in the Copland \textit{Appalachian Spring Suite} recording. The only difference is that the bass was routed through a stereo bus, instead of a mono bus. Once this was done, the microphones were subtly mixed in to add heft to the recording, but not so much as to make the basses stick out in the mix whatsoever. No reverb was then added, as a compliment to the Sacred Heart Catholic Church’s fantastic acoustics. This helps the recording remain perfectly realistic, and sound natural.

Overall, this is a substantially different sound than case study 6.1, which was recorded in the same room at the same positioning. In perfect conditions, such as the lack of an audience and the absence of an high volume air conditioning (HVAC) system, either microphone setup might work for either piece. However, I am pleased with the choice for each concert. The timbres of each stereo system compliment their music well, although this should always be left open for debate.

\textsuperscript{75} Sub-bass frequencies are typically sounds below 60Hz.
7.4 Case Study Four

*Tchaikovsky, Symphony No.4, Op.36 – Finale
Glazunov, Violin Concerto, Op.82 – Andante sostenuto*

On the final day of the Lake George Music Festival, almost every musician performing comes together to form the Lake George Music Festival Symphony Orchestra. It is the culmination of the hard work and musicianship throughout the festival. The program this year included Wagner’s Overture to the Flying Dutchman, Tchaikovsky’s 4th symphony, and Glazunov’s only violin concerto. These are all staples of classical repertoire, and as a classical engineer it is quite a treat to have the pleasure of recording these three pieces. Additionally, the soloist for Glazunov’s violin concerto was a young man named Stephen Waarts, a renowned violin player who had just turned eighteen at the time. As a young engineer myself, I found his outlook on the classical music industry quite easy to relate to, as bringing more youth into classical music is quite important. It was a fulfilling experience to record him.

A common occurrence for this kind of concert is that the audience attendance will be much higher. There will be several more bodies in the audience, all of whom will cough, sneeze, open candy wrappers, and walk around during the performance. Worse yet, many audience members feel the need to hold their coughs for after intense sections of music, right during the orchestra’s moment of silence, utterly ruining beautiful pauses. Thankfully, symphonies and orchestras are naturally much louder than smaller ensembles, which helps overpower audience noise. But while this is still useful, many musicians are depending on the engineer to do everything they can do produce a clean recording, so additional measures must be taken.
I recorded the dress rehearsal of this concert. This is a relatively common occurrence, but absolutely crucial in some cases. This enables the engineer to edit between rehearsals and the live performance, sometimes saving a piece from having its beauty tarnished by extraneous noises. It still has limitations, however. For a famous symphony piece, such as Tchaikovsky’s 4th, brass sections will often reserve themselves during the dress rehearsal to save the integrity of their embrochure for the performance. Luck can be with the engineer for a piece such as a violin concerto however, since string players do not always have the same physical limitations of other instruments. The Glazunov recording takes advantage of this since the violin cadenza is being drawn from the rehearsal, and the rest of the recording is from the live performance.

The recording of this concert was done with a Decca Tree array, and various spot microphones around the symphony orchestra. The reasoning being that the Decca Tree array can pick up the overall sound of the symphony orchestra, and the spot microphones can help manipulate the balance and bring out specific sessions when needed. See table 7.4a and figure 7.4a for a tracking sheet and overhead diagram of the recording session.

<table>
<thead>
<tr>
<th>Channel</th>
<th>Placement</th>
<th>Microphone</th>
<th>Preamp</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Bass Spot</td>
<td>AKG 414 ULS</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>2</td>
<td>Decca Tree Center</td>
<td>Oktava MK-012</td>
<td>MOTU Interface</td>
</tr>
<tr>
<td>3</td>
<td>Decca Tree Left</td>
<td>Modded Oktava MK-012</td>
<td>Grace M101</td>
</tr>
<tr>
<td>4</td>
<td>Decca Tree Right</td>
<td>Modded Oktava MK-012</td>
<td>Grace M101</td>
</tr>
</tbody>
</table>
Each microphone in the recording had a specific purpose. First, the Decca Tree array was placed in front of the conductor, and lowered down enough to capture as much direct sound from the orchestra as possible. This not only helps bring out the strings over the brass section, but also helps maintain a good signal to noise ratio when the stage is noisy with fans and electrical noise, usually from lighting system power supplies that are not properly isolated (as was the case here). The spot on the soloist was only there for the rehearsal, and removed and placed on the harp for the live
performance. This is to minimize aesthetic distractions, since having an obtrusive spot microphone on the headlining musician is extremely rude for the audience. The spot microphone on the harp for the live performance was then useful to help bring out the instrument slightly, since harps often get lost in the overall sound of symphony orchestras.

The spot microphone on the bass section was placed between the principle bassist and the second bassist, with the same purpose as the bass spot in case study 6.1, for the Copland piece. The spot microphone at stage right is to create a stereo counterbalance for the bass microphone, to ensure there is not too much energy on one side of the mix. Additionally, bringing up this microphone in the mix helps the strings remain clear in the presence of a powerful brass section, as was the case here. The spot on the timpani was only present to keep the timpani sounding localized to one point, as this instrument tends to rumble all around the stage and often sounds ominous or distracting as a result. There was a spot on the brass section for the performance, but admittedly it did not contribute to the recording as intended, and was quickly cut from the mix.

Before any mixing was done, delay compensation calculated based on measurements taken during the concert. The bass microphone, the timpani microphone, and the harp microphone were all measured to be four meters from the center point of the Decca Tree array. The stage right outrigger was three meters out. These calculate to 560 samples, and 420 samples of delay on the respective tracks.

An outrigger is a spot microphone typically placed further stage left or right of the main stereo array on an orchestra.
This is always a good first step in mixing orchestral music, since decision made with the intent of tightening up a mix could be solved with delay compensation.

As the main Decca Tree array would carry the recording, it was the first set of tracks concentrated on in the mix. A prominent task in treating these microphones was controlling the level of noise on the stage. This was ultimately caused by the power supply system for the stage lighting, which were not housed in a separate room. To deal with this, a sharp high pass filter was used at 45Hz, which was enough to slightly lower the level of noise while not negatively affecting the music. A relatively trustworthy rule of thumb for this process is to find the lowest notes the bass section, or tuba plays, and find the corresponding frequency. To negatively affect musical notes below 45Hz, a musician would have to play below an F1 according to a free mobile phone application called *Pitch Perfect* (see figure 6.4b), which was never done for this performance. To further noise reduction for these pieces, the high pass filter was then automated in accordance with the fades as described in case study 7.2.

![Figure 7.4b](image-url)

**Fig. 7.4b.** A screenshot of the mobile application *Pitch Perfect* showing a frequency chart in the sub-bass range.
Another decision made for the Decca Tree array was to add a slight saturation to the sound. Absolute clarity of preamps such as the Grace m101’s used is certainly ideal in some situations, but it had a level of lifelessness in this case. To counteract this, some extremely subtle tube distortion was used for everything above 500Hz and above using iZotope’s Alloy 2. Nothing was altered below 500Hz to prevent any accuracy loss in the low end. Additionally, since this can drastically add more high end to a recording even if used subtly, a high frequency shelf of -1dB at 500Hz was also added. This all ensures the same evenness of high to low end, while added a subtle warmth to the sound, at the risk of sounding too colloquial.

The bass spot was then added in with the exact same multiband parallel compression technique as described in case study 7.1. This is especially useful in an orchestra setting using a Decca Tree array, since the lack of proximity effect in omnidirectional microphones and the large distance between the array and the bass section can result in a thin sounding recording. In this case, if more microphones had been available, a second spot on the bass section (comprising of six bassists) would have been placed to create a more blended low end. Although this was not an option, having a great principle bassist such as Rick Robinson from the Detroit Symphony Orchestra certainly makes up for this loss.

The harp spot microphone, the timpani microphone, and the stage right outrigger were then all added in to taste, and they were panned to match where they

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77 Proximity effect is the phenomenon where bass frequencies are boosted when a microphone gets closer to a sound source. Omnidirectional microphones do not exhibit this characteristic, although the more directional microphones get, the more they exhibit this effect.
were naturally placed on the stage. Not much processing was done to these beyond slight high pass filters to help them sit unnoticed in the mix. A unique tool in the mix however is the stage right outrigger. Since the brass section in these recordings was relatively powerful, increasing the level of this microphone to bring out the first violins counteracts the brass section surprisingly well. This tactic was used to a pleasing benefit after the executive director of the music festival requested the string section to be brought out more.

Reverb was important for this mix. Since the concert took place in a high school auditorium, the hall was built to be economic, rather than reverberant. In addition, there were curtains lining the stage and ceiling, and nowhere for the sound to develop and reverberate. The sound directly from the recording was dry, and dull as a result. To make the recording sound more lively and interesting, a long reverb was added, around two to two and a half seconds. This was bussed off of every individual track,\footnote{This means that the audio from each individual track in Pro Tools was sent to an auxiliary buss containing a reverb plugin.} and levels were boosted and attenuated until it sounded as natural as possible. Finally, a limiter was placed on the sub mix for all the tracks. iZotope’s Ozone 5 was used for this, and it was only set strong enough to activate at the absolute loudest notes that the symphony orchestra played. It prevents the sub mix from clipping, enabling the recordings to be at an optimal listening level, and also makes the loudest notes slightly more full sounding, although there is never more than 1.5dB of attenuation.\footnote{Attenuation is the reduction in volume of a sound.}
As previously mentioned, the violin cadenza in Glazunov’s violin concerto was pulled from the rehearsal of the piece. This was not simply a drag-and-drop edit. First, there were multiple reasons that this was done. The most prevalent reason was the amount of audience noise during the cadenza. Several people coughed, several people moved around during it, and this is highly noticeable against a single violin.

Additionally, the amount of noise from the stage itself was also problematic. Some relatively heavy volume automation and cross fading was done to make this edit work (see figure 7.4c).

For scale, the Decca Tree array is being lowered by almost 5dB, and the soloist spot is being raised by more than 10dB after it is first faded in. Also, note that many spot microphones are being taken out of the mix, since they are not necessary to the

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80 Volume automation is programing the volume of a track to change over time automatically in a digital audio workstation.
mix at that point. Finally, equal power crossfades\(^8\) (as opposed to equal gain
crossfades)\(^2\) usually work better in these situations, preventing any dip in volume
during the transition. Great care must be taken during this, to ensure that everything is
working exactly as intended.

The reason that this automation is necessary is to first mask the edit. A
microphone cannot simply be added into the mix at full force, even with a fade. The
change in timbre of the recording would be too obvious. Secondly, the spot was a
cardioid mic, and the Decca Tree array is made up of omnidirectional microphones.
These two polar patterns do not mix without major problems. Cardioid, being the sum
of an omnidirectional and a bi-directional\(^3\) signal, becomes a subcardioid\(^4\) polar
pattern when combined with a second omnidirectional signal (Bartlett, 932). This can
work with microphones are coincident, but this was not the case here. When
omnidirectional microphones and cardioid microphones are not coincident, this creates
highly destructive polarity\(^5\) issues. In order to prevent this from being an issue, the

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\(^8\) Equal power crossfades are crossfades in which the volume is exponentially
lowered between each clip to transition between two audio clips.

\(^2\) Equal gain crossfades are crossfades in which the volume is linearly lowered
between each clip to transition between two audio clips.

\(^3\) Bi-directional is a polar pattern in which a microphone picks up sound in front and
back, and has null points on its sides.

\(^4\) Subcardioid is a closely related to cardioid in which a microphone picks up most of
the sound in the front, but still picks up some sound at the rear point.

\(^5\) Polarity is the positive/minus alternating current (AC) orientation of an electrical, or
sound signal (McGregor).
omnidirectional microphones in this scenario were automated to be significantly quieter than the cardioid spot.

This edit helps Stephen Waart’s excellent cadenza sound clear and free of excess noise in the recording, while the rest of the piece can still sound how it was intended to sound. Two completely different mixes are effectively spliced together, and no compromises are made for either which is important because distractions during the cadenza of this piece would be upsetting. Audiences may expect certain noises during the live performance of a piece and while this is a reality of live recording, the same members of that audience may be taken aback when someone’s cough overshadows an especially soft phrase in the music. As unfair as this is, the engineer must learn to live with that conundrum and overcome it in one way or another. In this particular scenario, recording the rehearsal of the piece was the perfect solution. This is an effective tactic most of the time, so take advantage of it when possible.
8. DELIVERY, REFLECTION, AND CONCLUSION

The majority of the mixing for the entire library of the festival’s recordings was done after the festival had ended. I did not have the opportunity to use accurate studio monitors during the mornings and afternoons before concerts, and mixing solely on headphones was not an option for reasons previously described. However, the first round of edits for every concert was done the morning after the concert. This was done to give the staff of the Lake George Music Festival something to listen to, and to give them early feedback of their own work. Additionally, this helped them decide early on which recordings would be submitted to the National Public Radio’s segment *Performance Today*, in which these recordings are often featured. Doing this for the staff of any music festival is usually not mandatory, as their sights are always focused on the next concert, but it is a kind gesture that never goes unnoticed.

Proudly, the CEO of the Lake George Music Festival changed the Recording and Production position’s job description in accordance with how I delivered the music to them after the festival. Previously, the contract stated that the music would be delivered back to them via redbook format\(^\text{86}\) compact discs. This is not up to the standard of how engineers should operate today, since redbook formatting is not actually a perfect copy of digital audio. However, using data discs is. In addition to the

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\(^\text{86}\) Redbook compact disks are formatted to a 16 bit bit depth, and a 44.1kHz sample rate format.
delivered compact discs, three data DVDs\textsuperscript{87} containing the 16 bit,\textsuperscript{88} 44.1kHz\textsuperscript{89} quality recording, as well as 320kbps quality mp3 copies were included. This way, 
distributing recordings to CD replication companies, or uploading audio to various 
online services is easy for the client. It is a simple gesture that only takes a small bit of 
time, but adds a much greater level of convenience for the client.

Improvised techniques in classical location recording are not exclusive to 
professionals or those who are already established in this industry as being among the 
best. Truly, they are anything that the engineer wants them to be. They always begin 
in an audio engineer’s canon of recording techniques as an idea, and become reality 
when the engineer has time to put them together and test them at home. Soft skill 
techniques never have a definitive point of reality either, which is interesting in that 
their origins must be unique to every individual recording engineer.

So, hardly any techniques are exclusive to any one engineer at all. Yes, Glyn 
Johns is certainly entitled to call the Glyn Johns drum recording technique his own, 
and the Office de Radiodiffusion Télévision Française is the clear originator of the 
ORTF stereo recording system. But short of these few examples, dozens of different 
methods to mount a Decca Tree array have probably been created, and hundreds of 
classical engineers have probably put together modular microphone suspension 
systems that can be implemented in unique locations. To that example, the offer of this 

\textsuperscript{87} Data DVDs are unformatted DVDs that can carry media of any kind as flash 
storage.

\textsuperscript{88} Bit depth is the amount of data per sample in digital audio files.

\textsuperscript{89} Sample rate is the rate per second at which digital audio is measured and recorded.
thesis is to describe a single reliable, effective, and easy way that any engineer can replicate, modify, or improve. Any technique described here is up to that same philosophy, so please be adventurous and make something new.

This is what being a classical location engineer is all about. Good classical engineers are crafty by nature. They should never let a design flaw hold them back permanently. At worst, an engineer must make do with an imperfect piece of gear once, then brainstorm how the design could be improved, or how to develop a workaround that would fix the situation. An engineer who is unwilling to put forth the imagination required to do this is either lazy and will not be successful in this business, or has a large enough budget to buy anything he or she would ever need!

After all, a classical engineer’s mobile recording studio is their paint brush. A painter who is unwilling to do anything but dab and brush paint onto their canvas may never create anything truly unique, and a recording engineer that is unwilling to get creative with their microphone techniques will never create a new recording that is above and beyond their last recording. Improvise, design, create, and be an entrepreneur. That is not just the future of recording technology; it is the present, and has always been the present for every generation of audio engineer.
BIBLIOGRAPHY


Please find the following supplemental recordings attached with this thesis.

They correspond with four case studies, and aid in this exploration of classical recording using improvised techniques. They include:

**Case Study One**

**Appalachian Spring** Suite (1943-44)                              **Aaron Copland** (1900-1990)
Variations on Shaker Hymn (Simple Gifts)                             (original version for 13 instruments)

**Case Study Two**

**3 Pièces Brèves** for wind quintet (1930)                                **Jacques Ibert** (1890-1962)
Allegro

**Case Study Three**

**Virtuosity**, five micro concertos for string orchestra (2013)  **David Ludwig** (b. 1974)
Con moto: *Concerto for two violins*

**Case Study Four**

**Symphony** No.4 in F Minor, Op.36 (1878)                              **Pyotr Ilyich Tchaikovsky** (1840-1893)
Finale. Allegro con fuoco

**Violin Concerto**, Op.82 (1904)                                      **Aleksandr Glazunov** (1865-1936)
Andante sostenuto
APPENDIX B: Musical Works and Credits

I would like to extend my deepest gratitude to the musicians and administration of the Lake George Music Festival who organized and performed the music used in this thesis. They are as follows:

Alexander Lombard, President & CEO
Barbora Kolářová, Artistic Director and General Manager, Vice President
Roger Kalia, Music Director and Conductor
Brendan Faegre, New Music Advisor
David Ludwig, Composer in Residence
Helen Tobias, Treasurer
Nancy Stafford, Secretary

Case Study 7.1

Appalachian Spring Suite (1943-44) Aaron Copland (1900-1990)
Variations on Shaker Hymn (Simple Gifts) (original version for 13 instruments)

Lydia Roth, flute – Christopher Pell, clarinet – Martin Gorden, Bassoon – Sam Fischer and Jennifer Choi, violin I – Xynie Niu and Allison Reisinger, violin II – Kimberly Sparr and Andrew Griffin, violas – Daniel Lelchuk and Hannah Collins, cellos – Kieran Hanlon, double bass – Maria Yefimova, piano – Roger Kalia, conductor

Recorded at the Sacred Heart Catholic Church in Lake George, NY
Case Study 7.2

3 Pièces Brèves for wind quintet (1930)  
Jacques Ibert (1890-1962)

Allegro

Mimi Stillman, flute – Samuel Nemec, oboe – JJ Koh, clarinet –  
Catherine Chen, bassoon – Phillip Browne, horn

Recorded at the St. James Episcopal Church in Lake George, NY  
Recorded 14 August 2014

Case Study 7.3

Virtuosity, five micro concertos for string orchestra (2013)  
David Ludwig (b. 1974)

Con moto: Concerto for two violins

Barbora Kolárová, Jacob Ashworth, Milena Kolárová, and Megan Prokes, violin I –  
Suliman Tekalli, Hyewon Kim, Minji Kwon, and Noco Kawamura, violin II – Colin  
Brookes, Lauren Nelson, and David Mason, violas – Natalie Helm, Nicholas Finch,  
and Arlen Hlusko, cellos – Matthew Weber and Kieran Hanlon, double basses

Recorded at the Sacred Heart Catholic Church in Lake George, NY  
Recorded 15 August 2014

Case Study 7.4

Symphony No.4 in F Minor, Op.36 (1878)  
Pyotr Ilyich Tchaikovsky (1840-1893)

Finale. Allegro con fuoco

The Lake George Festival Symphony Orchestra  
Roger Kalia, conductor

Recorded at the Lake George High School Auditorium  
Recorded 21 August 2014
Violin Concerto, Op. 82 (1904)  
ALEKSANDR GLAZUNOV (1865-1936)

Andante sostenuto

The Lake George Festival Symphony Orchestra  
Roger Kalia, conductor  
Stephen Waarts, violin

Recorded at the Lake George High School Auditorium  
Recorded 21 August 2014