OHIO UNIVERSITY SCHOOL OF MUSIC

DEPARTMENT OF RECORDING ENGINEERING

Recording Classical Music

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for the position of

STUDIO ENGINEER

January 2022

Acknowledgements and Copyright

Contents

Introduction

What is this guide? And who is it for?

Hello! If you are reading this, it is most likely because you work for the Ohio University School of Music. If you don't, you were most likely recommended this text by someone else, in which case – wow, I'm famous.

I hope I have kept this guide brief while still providing the necessary information to begin creating high quality recordings. Recording is a huge topic, and many of the topics in this guide could be books in themselves. This guide is by no means comprehensive. It is my hope that by the end of this guide you are both equipped to handle the basics of the position to make beautiful recordings, as well as primed on how to effectively search for and seek out the answers to any problems you may encounter in the process. In other words, I hope to build familiarity more than anything – half of becoming technically literate is building vocabulary and a sense of what exists. At times this may feel pedantic or obvious, but I am in some ways here to affirm the obvious.

For that reason, although this guide is written for the Ohio University School of Music, I will not dwell on specifics. I have written out the current process of recording and streaming, but this is subject to change. Technologies shift and equipment is replaced as money becomes available. Most likely what I write here in the winter of 2021/22 is going to change in the next couple of years. What does not change however, is physics. Rooms will continue to be rooms, sound continues to travel at the same speed, and even some technologies like microphones haven't undergone a radical shift in decades. I will of course include a section of specifics so that the current setup is documented, but this is meant to be updated – hopefully by you, dear reader.

I am writing with the assumption that you have next to no experience with recording, classical music, or its technologies. It's not that I don't trust you or think you're smart, but I want this document to be accessible to anyone. For those of you with some experience, it may feel like I'm explaining that water is wet at times, but do your best to read through it anyway on the off chance that you might learn something. If nothing else, it might serve as a reminder to go back to the basics and remember what it is that is actually going on when you record something. I know I am constantly having to return to stuff I learned years ago, and each time I do I find that it comes to me in a different light and brings with it a deeper understanding. So please, don't feel talked down to – I'm trying to explain this so even your technologically illiterate

grandparent might be able to successfully make a decent recording by the end.

I have organized this guide in an order that I hope will make sense to you. I don't want to overwhelm you as a reader with too many technical details without practical application, but I also am writing this as a companion to physically being in the studio with a more experienced technician. There is also a need to explain some terms before I can really tell you what to do, so bear with the second chapter. It's by far the most technical, but if you can get through it even vaguely understanding it it'll provide you with a foundation for everything else in the book.

And finally, I am hoping to instill a sense of curiosity and wonder about the world of classical music recording. Classical music is in itself relatively obscure within our larger culture, and the process of recording it is even further removed from any popular understanding. It is not rock music recording – artful as recording rock is, the techniques and philosophies differ greatly. If you're coming from a studio background, perhaps as part of the MDIA music production track, you might find yourself initially frustrated, or even bored. I promise that with an attention to detail and an idea of what is possible, this will pass. I hope that this guide can serve you in this way.

All the best,

Sam

Chapter 1: What is Recording Classical Music

Fundamentals

With the introduction out of the way (which you should read, by the way), lets dive into what exactly it is we are doing here. In this position, you are tasked with recording and live-streaming the concerts and recitals that the School of Music puts on. These should be clean, clear, and functional recordings in some ways – parents and students alike have an interest in being able to hear the performance! Nothing is worse than going back to listen to a performance that you worked months for, only to find out that the audio is garbled, muddy, or otherwise unlistenable. We are here to serve the music.

This can sound daunting. Many people are put off by the fact that they think they are going to mess something up, or that the technology will be too confusing. Let me put this out there right away: *you are going to mess up.* That is ok. The stream will crash, you will lose a performance, your recording will be lost forever in a mess of ones and zeroes. Take a deep breath, figure out what went wrong, and do it better next time. We are in many ways totally at the mercy of the technology. People put a lot of trust in the technologies we use, but most of those people have never dealt with those technologies intimately. Every technologically savvy person I know – people with 30, 40+ years of experience – deeply distrusts technologies. That doesn't mean that it is a bad thing, but rather that understanding a technology is knowing its limitations and being mindful that it will behave unexpectedly.

The best way to proceed in all these situations is to not panic. Professionalism isn't just about impressing your boss, it's a way of working that will allow you to handle potentially stressful situations. I know this all sounds really obvious and pedantic, but I'm serious – acting calmly in these situations takes a lot of practice to fully commit to. It sounds easy on paper, but I say this now so you can really internalize it. Prioritize the calm, and the rest will come easy.

Let it breathe

While we're on the topic of fundamentally important things that you should think about always, I want to throw out a saying that a friend introduced to me years ago: *let it breathe.* Classical music is all about the performance in a space. For those of you with rock music studio experience, this might be familiar but may not seem essential. In rock and pop settings, instruments are often recorded with the microphones

quite close to the source, not necessarily allowing for the reverberations and flavors of the space to flourish. This has its uses, and there are certainly classical recording engineers who will experiment with close techniques (myself included), but overall I want you to remember to *let it breathe.* Allow the room to speak for itself – it's a lot more interesting than you think. Don't worry if none of that makes sense yet, with some experience you will see what I mean. *More on this to come*

Use your ears

This sounds so obvious, but your ears are your tools. Protect them, avoid listening at super high volumes. Learn to trust your ears, but also know your limits – your ears can quickly become fatigued. Generally though, if it sounds good, it is good. There are no hard and fast rules here.

What is our role here?

I also want to touch briefly on our role as a recording engineer. Without being too preachy, nobody likes the whiney, jaded engineer where every request is like pulling teeth. It is easy to fall in that trap, in part because so much of the culture of this job is steeped in that kind of mentality. Anyone who has experience with their local grumpy IT guy knows exactly the type.

There will be times it will be extremely frustrating to work with people. Artists can be whiney, needy, overly picky, uncommunicative, and straight up rude. You will probably feel under-appreciated. If you're in it for an ego boost, find another profession. But for every unpleasant experience, there are dozens of amazing opportunities to make something great with an artist. Being supportive, communicative, and open-minded are your best defenses against feeling jaded or picked on. Be open to trying new things with an artist, even if that means a little more work on your end. You will find yourself much more invested and involved, as well as happier.

Large Chapter 2: Intro to the Technologies for the Uninitiated

Or, dumb questions you think you (and might actually) know the answers to

What is sound?

Brief refresher from your high school physics class: sound is nothing more than vibrations. Say you have a speaker in a room – it produces sound by vibrating its cones back and forth. This is absorbed and transmitted by the air around it, essentially setting off a chain reaction.

This is slightly more complicated though, because *really* what sound is is a pattern of areas of high and low compression. Think of air like spongey dominoes. In the speaker, as the cone moves forward, it physically bumps into whatever air molecules are sitting in front of it. These get smacked forward and hit the molecules in front of them, transferring their energy and returning to a more docile state. This wave of energy is continually shunted forward at the speed of 343 m/s (roughly). In the world of music, though, we are not talking about just one wave of energy, but many. When it comes to musical sounds, there are a couple of considerations.

The first consideration is what is called frequency. This is how quickly the waves are coming, or essentially, how close together they are. A faster moving speaker cone will produce a faster succession of energy waves, which we would refer to as a higher frequency. Fortunately for us, higher frequencies produce higher-pitched tones – easy to remember. Conversely, the slower these waves move, the lower the frequency, and the lower the tone. Frequency is expressed in Hertz, abbreviated Hz, which measures how many times per second the wave is vibrating. Most western classical music is tuned to 440 Hz (the note A) as a reference point. You will probably hear about this. Some weird people tune to 432 because they think its more "in sync" with the "fundamental frequency of the universe," which is frankly a load of bollocks. Oh well.

The other consideration is amplitude. This is how much energy is packed into a wave. If we take a look at our speaker cone again, this is how *much* the cone is moving. Amplitude controls the volume of a sound. If you look at our wave form, they are getting taller without getting faster. Think of this is in terms of pressure. If we return to our air-as-spongey-dominoes analogy, it's like the speaker is kicking those molecules harder or softer, resulting in the air moving with more or less force. This is also why

loud sounds destroy our ears – high sound pressure quite literally shrivels up the hairs in your ears that you use to hear and causes them to fall off and die. Pleasant.

We measure amplitude in terms of decibels, abbreviated dB. This scale is a little strange though – it's logarithmic rather than linear. That means that the difference between 0-10dB is not the same as the difference between 10-2-dB. Each 10dB jump actually marks a 10-fold increase in sound pressure. Thus, a 10dB sound is ten times louder than silence, and a 20dB sound is one hundred times louder than silence. This seems theoretical and overly technical, but I promise knowing what decibels are actually measuring will come in handy as you start recording.

What are waves?

Another refresher from your highschool physics class. Waves are modeled on graph in undulating lines – I'm sure you've seen this before. This is a mathematical way to express something that is happening in the real world. When we draw these graphs, they are drawn as to show us both amplitude and frequency. Amplitude is the spread of the wave on the y-axis, while frequency (which is time-dependent) is shown on the x-axis. However, it is important to remember that this is a model, not a drawing of reality. Real sound waves are a. multidimensional (they spread out in all directions) and b. areas of compression and rarefaction (the opposite of compression).[1]
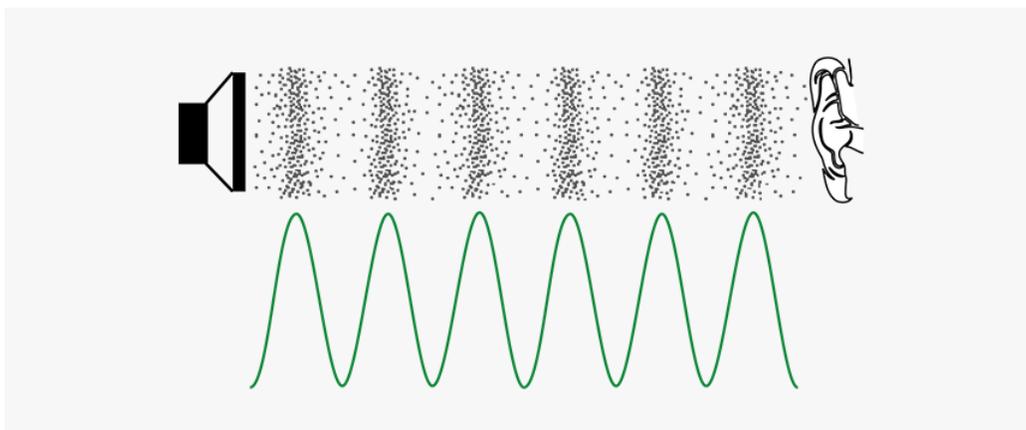
Figure 1: Compression and Rarefaction

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[1]Diagram from https://www.pngitem.com/pimgs/m/12-127156_sine-wave-compression-and-rarefaction-hd-png-download.png

The one aspect of a wave that I haven't talked about yet is phase. This is easiest to understand with a graph. Imagine a wave, beginning with the energy rising (compressing the air) and then falling. This would be modeled as follows:

Now imagine a wave that starts later in this cycle, halfway through. It would begin by that period of rarefaction (again, the opposite of compression), and then in its second half rise to high energy compression. We can express this shift in terms of degrees – 180 degrees shifted, or "out of phase," as we would say, would be shifted half way like I just described. This will be important in a minute.

As of yet, I have talked about amplitude and frequency in a theoretical sense – pure tones, totally regular frequencies. What happens when we want to produce musical tones with texture and nuance? Ultimately, all sound is a combination of frequencies and amplitudes. What is important to understand is that amplitudes are additive. Two waves at the same frequency played starting at the same time will add together to make a wave with twice the amplitude. Conversely, two waves with the same frequency played with one shifted 180 degrees out of phase (again, shifted halfway) will actually cancel each other out. Because there is an equal amount of compression and rarefaction occurring at the same time, the molecules will simply go nowhere. [2]

Figure 2: Phase cancellation

This has two major implications. The first is that any combination of sounds, pitches,

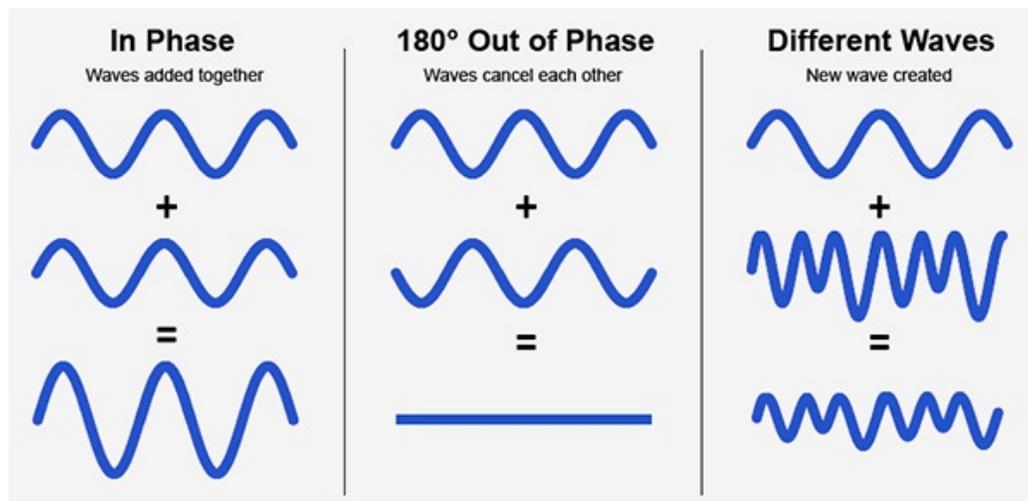and volumes can be modeled by a single wave. This wave will look complicated – take a look at this close up of an orchestra recording. See how the wave is not at all regular, but rather looks like a rollercoaster? This is the result of hundreds of sounds, all with different phases, amplitudes, and frequencies adding together to form one wave.
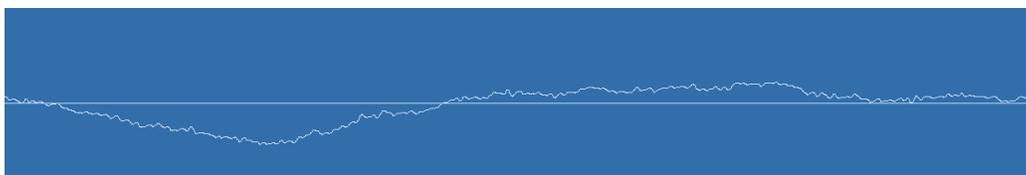
Figure 3: Zoomed in orchestra waveform

The second implication is that, in poor conditions, you can end up with something called *phase cancellation*. This is an important term, so take note and keep it in the back of your mind. Remember how I said that two waves of the same frequency and amplitude will cancel each other out or add to each other depending on where in their phase they are added together? Depending on your recording setup – microphone placement, room configuration, etc, it is possible for this to happen on your recording. It sounds nasty, and is kind of hard to describe. You can look up videos on youtube for examples, or have whoever is the current senior engineer point it out to you next time you're recording. That's all I'll say on that for now, but if you're wondering why your recording sounds thin and strange, phase might be a large part of that.

What is a microphone?

Ok. Deep breath. I promise that was the worst section. Time to move on.

Now that we know what sound is, a microphone will (hopefully) make a lot more sense. Remember how a speaker is constructed of a cone that vibrates the air? A microphone is (almost) nothing more than its receiving end counterpart. Instead of a cone, a microphone has a thin membrane. This is vibrated by the air in the same way that the cone vibrates the air initially. Pretty simple. The construction within the microphone that contains this membrane is referred to as the *capsule*, while everything else is the *body*.

Where it gets a little more complicated is how that analogue, real life sound pressure is converted into something we can use to be captured and recorded. On a basic level, the microphone takes that movement of the membrane and turns it into electrical

signals. Simple enough. There are two(ish) types of microphones – *dynamic* and *condenser*.

Dynamic microphones are fairly simple. They are about as close to the opposite of a speaker as you can get. The membrane moves a coil of wires around a magnet, producing small impulses of electrical current. These impulses mirror the original sound signal and can be recorded through various media to be played back by a speaker later.

Condenser microphones are slightly more complicated. Instead of a coil, condenser microphones contain two membranes: one that is fixed, and one that can move. These plates are charged, and as sound hits the moveable one, create a difference in charge between them, resulting in electrical impulses. Because of this, condenser microphones require power to them, referred to as *phantom power.* On recording gear, this is usually a button labeled *48v*, referring to the 48 volts that the gear supplies the microphone.

The specifics are not terribly important here, and microphones are an infinitely complicated topic, but it is important to know the basics of how the two microphones function. For now, know that condenser mics are typically more detailed and sensitive than dynamic mics, and are able to accurately pick up sounds at much greater distances. For this reason, we tend to pretty exclusively use condenser mics for classical music recording, where detail is imperative and soft sections are common.

What is a room/basic room physics

Rooms! We all live in them.

Understanding the sound of rooms is quite simple. In fact, you can try it right now. Walk through your house, or dorm, or wherever you are, and whistle, sing, or talk out loud. As you move through different rooms, notice each space has its own character and adds something to the noise you're making. Showers are especially noticeable, as are kitchens full of marble countertops and living rooms full of plush chairs and carpeting. Some are quite live and reverberant, while others are much more muted and quiet. Make this a practice in your day to day life – notice how the size, shape, and materials in a room affect the way that things sound. As you get better at identifying these characteristics, start to notice how your place in the room changes this too. Standing in a corner is significantly different sonically than standing in the dead center.

Without getting too technical, this is all essentially a result of the way that sound is

reflected and absorbed by different materials. Some materials, like concrete and glass, are highly reflective of sound. Stand in a parking garage or in front of a mirror and notice how the sound is reflected back at you. Others, like wood, are more neutral, and materials like carpet or fabric tend to absorb almost all the sound thrown at them. In some ways, this is very intuitive – if it's shiny and hard, it probably reflects sound too.

What becomes deceptive is the role of the shape of the room. There are techniques for deadening a room made of reflective material that rely on the positions of the walls. Let's take the recital hall in Glidden Hall (our main recording space – start thinking about it!) as an example. If you look closely, you'll notice that there are essentially no parallel walls. The side walls splay outwards as they go towards the back of the hall, and are angled upwards. The back wall is parallel to the stage wall, but is also angled upwards. The stage wall is relatively straight but curves up at the top, forming the wavy ceiling in that space. All of these are intended to reduce some of the reverberations in the space – while you want a space to have some life to carry the weight of the instruments, you also don't want the sound of the instruments to be buried in the sound of the room.

Unfortunately, the recital hall is not a great room. It's acceptable, but nothing truly sounds great in that room, which is too bad for us. It's up to us to mitigate some of those issues through creative mic placement and critical listening.

If you are looking for a much more live room, check out Galbreath chapel on college green. The chapel is a highly symmetrical octagonal room with high ceilings, allowing plenty of space to bounce around. Large windows and plaster walls add to this effect. This room can sound great in some situations, notably solo violin and some string quartets. As a general rule, lower pitches tend to get lost in high-reverb situations, and the symmetry of the room can actually create problems with the aforementioned phase cancellation (see, it really is important!). Really loud sounds can also become a problem in these kinds of rooms, so something like a brass quintet might have more problems than a string ensemble might. We don't always have choice about the room though, so use your ears and follow what you think sounds best!

DAWs, A/D converters, computers, and other mumbo jumbo

I debated somewhat whether I would include anything about analog recording in here, and while understanding these fundamentals can be enlightening on their own, I decided not to address them out of an interest in brevity.

So, here we are stuck with computers. It's a digital age and pretty much everything is recorded digitally now. In order to do this, we use something called a DAW, or digital audio workstation. This is the application on the computer that stores and organizes all the recorded audio from all your various microphones and input sources. Within a DAW, you are able to make edits, cut the recorded piece to desired lengths, and make various other sonic adjustments. Essentially, it is an all-in-one editing and recording station. Examples of such programs include Logic Pro X, Adobe Audition, and Pro Tools. Each have their strengths and weaknesses, all are overpriced, and all of them will at some point have you pulling your hair out over their design choices. I won't go into the specifics of how to use an individual DAW here – that's what youtube and google are for – but I will briefly address their basic functions so you know what kinds of things to google.

Tracks: Within a DAW, you are able to create tracks. These can either be mono – one channel, or stereo – two channels, one left and one right. These tracks are a way of keeping separate your input sources, usually microphones. For example, when recording an orchestra you may have a four microphone setup – a pair of stereo microphones in the middle of a hall, as well as one mic closer to the piano and one closer to a string bass. The stereo pair will likely be combined into one stereo track, while the piano and bass mics will each get their own mono tracks. Later, after the recording is finished, these tracks can be individually edited with effects and have their volume levels set relative to each other to ensure a good balance between instruments.

Recording: Before recording anything, most tracks will have to be *record enabled.* Usually this is a red button with the letter R on it somewhere on the track. This primes the track to be written to by recording. It's usually obvious where it is, unless you're Garage Band in which case call me in two weeks when you've finally found that stupid thing buried. Google is your friend.

Input monitoring: This is a function of DAWs that allows you to hear in real time the input of a track. It can be enabled as a way of checking that you are recording what you think you are, and can be used to to monitor your recording as it happens.

Routing/sends: Most DAWs have the capability to send audio from one track to another. Usually the endpoint of this audio is something called a bus. A bus is a dedicated channel that audio is sent to, usually for the purpose of applying effects to a group of tracks. The most common use of a bus is for applying reverb. There are also DCAs and VCAs, which are similar to a bus but different and outside the scope of this article. Google is your friend.

Meters: Meters are a tool to measure the volume of your recorded sound. They aren't something you do anything with other than watch to make sure your track volumes are at an appropriate level. And remember how I said that waves are additive? The more tracks you have, the louder your overall output is going to be. Meters help you to make sure that nothing is clipping (a fancy word for when the audio is so loud that it overwhelms the digital processing and makes a nasty distorting sound).

Inputs/outputs: Your sound has to get into your DAW somehow, and then also get back out again to reach your ears to play it back. DAWs tend to have a couple settings for this. First is at the level of your audio device – which piece of hardware are we using to collect input and output it back? From here, there is a track-specific menu (often of the drop-down sort) that allows you to select which input to use. Outputs are generally pretty static, but they can be changed in some situations. Google is your friend.

Ok, enough about DAWs. While we are on the topic of getting sound into your computer, how in the world does that happen actually?

The answer is something called an *audio interface*. Again, each one is different, so I'll spare you specifics. Remember how microphones take real-life sounds and convert them into electrical impulses? In the old analog days, these impulses could be directly written to tape via magnets – hence, *analog*. Unfortunately, computers don't understand analog input in that way, they need ones and zeroes. In the slightly less olden days (c.a. early 2000s), there was a standalone device called an A/D converter, short for analog-digital converter. The sole job of this device was to convert those electrical impulses into ones and zeroes that the computer could understand.

The modern-day interface is sort of an all-in-one device that includes an A/D converter, among other things. The technology is all compact enough that it all fits in one nice package, so for most applications save for high-end studios people tend to opt for the more compact design. So what are these other elements now?

The main element of the interface, and usually the most obvious, are the microphone inputs. This allows you to plug your microphones physically into the device via a cable and receive input from them. Each input is usually accompanied by a *gain knob*. This allows you to adjust the *gain* of each input – in essence the volume level of the microphone as it is input into the computer (much more on this function later). This is also where you will find the phantom power "48V" button to provide power to condenser microphones that I mentioned earlier in the microphone section. These inputs are attached to something called a preamp. These are small circuits that amplify

the sound input enough that it is readable by the analog-digital converter. They also have some effect on the quality of the sound and are the subject of much debate.

Interfaces now are usually connected to your computer by a USB cable, although this will probably change as we change input methods. Ten years ago firewire was all the rage, who knows what it'll be in ten years from now. Just remember to plug the darn thing into the computer – otherwise you'll be very confused. Interfaces now also have near infinite features across different models. Some allow for MIDI input and output, some are able to take complex digital signals to act as external inputs, and some probably make you coffee. These are outside our discussion though – google is your friend.

Signal flow

Ok, we're finally getting somewhere. I promise this is the last section in this chapter. It's also the most important, so go get some coffee, wake up, and pay attention.

Signal flow refers to the way that your audio signal travels through all your devices. The collection of these devices in order is referred to as your *signal chain*. It's a simple concept, but deceptively tricky to really get into your head. Remember that sound flows in one direction, and it starts at the source. To make this more concrete and clear, here is an example of a basic signal chain:

A violinist stands on stage, producing sound. This sound is captured by a microphone. That sound travels through the cable (called an XLR cable – see the glossary) in the form of electrical impulses. It reaches an interface and travels through the inputs there, being regulated by the preamps before the signal reaches the analog-digital converter. The A/D converter transmits this signal into something readable by the computer. The computer takes this signal and routes it to the DAW, at which point the DAW takes over and routes each individual signal to different tracks. From here, if input monitoring is enabled, the DAW will route this audio signal *back* to the interface, at which point it is routed once again through the A/D converter, but this time it is routed to the outputs of the interface. If speakers or headphones are connected, it will then play back through those and reach your ears.

In this scenario, the following devices are involved, in the following order:

- Violin
- Microphone
- Cable

- Interface
  - Preamp
  - A/D converter
  - USB cable
- Computer
- DAW
  - Tracks
  - Input monitoring
  - Output device
- Interface
  - A/D converter
  - Output
- Cable
- Speaker

This is a fairly simple routing scenario. More complex setups can become daunting and overwhelming, organization is key! It often helps to make a list of all the items that are involved in your recording setup. If any one of these elements are in any way messed up, your signal might not reach where you want it to go. More on troubleshooting later, but keep this in mind when things break.

Chapter 3: Specifics to Glidden Hall

Wow. Made it through the technicalities – now here are some specifics for dessert.

Remember, this is the procedure as of winter 2021/2022. It is very likely that the system will change and this chapter will become outdated very quickly.

Where are and what are (sound) things?

Let's start from the beginning of the signal chain and work our way backwards in the recital hall. Microphones are kept in a black mobile cart in the studio. As of writing, we own three AKG 414, two sE7, and two Earthworks SR30 microphones. The Earthworks microphones have their own fancy wooden box, although this might change. The applications of each will be enumerated later.

Cables are generally kept on the green shelving unit in the studio. As of writing it's a bit of a mess there, but if they are nicely coiled the cables will be just fine sitting on a shelf.

Stands for the microphones live in the corner of the studio. There are two types – the small, boom arm stands and the much larger, heavy-duty stands with no boom. The latter are preferred for most applications. There's nothing sillier than putting $1000+ of microphone on a stand that looks like it's about to topple at any moment. Unfortunately, those larger stands have some non-standard attachments that are originally designed for lighting equipment, so they require an adapter for use with microphones. These adapters should normally already be in the clips for the microphones, and if not, best of luck – those things are tiny.

Speaking of clips, microphones like to be special and have their own proprietary clips. These are generally kept on the green shelf with the cables. There is also a stereo spacer bar that should live with the clips.

The inputs on the stage of the recital hall are a little complicated. There are four inputs at the front of the stage that are the most commonly used. They are numbered three through six and correspond to inputs three through six on the interface. Inputs one and two are on the ceiling, along with a ceiling mounted stand for mounting microphones up there. In my experience, they sound terrible mounted up there and are not particularly useful. Input seven is mysteriously AWOL, presumably destroyed when the ramp was put in. Input eight was formerly an output for a stage monitor on stage left. It does still correspond to input eight on the interface. I have used it on

occasion, it works fine. Input nine and ten are at the time of writing re-adapted to be outputs for a playback system. They are routed out from outputs three and four on the interface, and are run to the hall's PA system in the closet via XLRs. I did this specifically so that they could be readapted to be inputs again if the need ever arose.

The interface is mounted in a wall-embedded rack above the desk in the studio. It is the bright red device with lots of knobs. At the time of writing, the model is a Focusrite Clarrett 8-pre USB. It is powered by a Furman power conditioner (a device that "cleans up" power to prevent humming and buzzing in audio equipment) that is mounted at the top of the rack. The power conditioner also powers the two Dynaudio studio monitors (speakers) that are connected to outputs one and two of the interface. The monitors and interface should be turned on and off with the switch on the power conditioner – the actual power switches for them are on the back and involve moving them around inconveniently.

The Focusrite interface is connected to the Mac Mini via a USB cable. Check to make sure this is plugged in, it's easy to overlook. It should stay plugged in in theory.

Where are and what are (video) things?

As of writing, there are two cameras in the recital hall. One is a mobile Sony camcorder, and the other is a PTZ (pan-tilt-zoom) camera mounted at the back of the hall, controllable by a remote controller. The camcorder is in a black road case along with its mounting gear, some HDMI cables, and the HDMI to USB interface card (this card does to video what the audio interface does for audio. Simply plugging the HDMI cable into a computer will get you nowhere.)

The PTZ camera is wired through an ethernet cable that runs through the walls of the hall into the studio. This ethernet cable both powers the camera as well as provides it the necessary input from the controller. It does *not* transmit any video. In order for the camera to function, this ethernet cable *must must must* be plugged into a power-over-ethernet capable switch (POE – google is your friend on that one, too much for this guide). Otherwise, it will not get power. There is a second cable called an SDI cable that runs from the camera into the studio through the same path. This is the cable that transmits the video signal. It runs into an SDI to HDMI converter. From there, the signal runs into an ATEM Mini Pro that is connected via USB to the computer. Again, this serves the purpose of converting the video signal into something that is usable by the computer as an input. This device (the ATEM Mini Pro) is the little black box with the numbers 1-4 on soft buttons and three buttons labeled "Auto,"

"Cut," and "FTB." It can also act as a video switcher if more than one video device is connected to it. See the more reading section at the end.

At the back of the hall, there is an ethernet cable that is taped to the floor, right around under the PTZ camera mount. This cable runs back to the studio as well, and can be used in combination with a pair of ethernet to HDMI converters that are stored with the Sony camcorder to transmit a secondary video feed from the back of the hall to the studio. Note: these converters are directional, i.e. one is only an output and one is only an input. If it is not working, try switching them around. They also require power from small power adapters.

As of writing, there is only one tripod, since there is only one camera. Remember to bring this if you are going somewhere to record on site.

Livestreaming

Part of this job is livestreaming. A disproportionate part of this job is troubleshooting the livestream when it breaks. Here are the procedures and some tips and tricks when it inevitably breaks.

As of writing, we use a program called OBS to stream performances. It is free, so when you need to stream from your laptop you can simply download and install it. Much of OBS is quite self explanatory – the "start streaming" button starts the stream, the "start recording" button starts recording what is being streamed, etc. However, in order for OBS to function, you are going to need what is called the *stream key* from the School of Music Youtube channel. This is a unique identifier – in essence a password – that you can provide OBS so it knows where to send its data. It can be found on youtube labeled "stream key" when you click "go live now." In OBS, click on the "settings" button in the bottom right corner. Then, click on "stream." From the drop-down menu, select "YouTube (RTMP)" and paste the stream key into the box labeled "stream key." For more updated info on how to do this, consult the OBS wiki/manual pages, which will have more detailed instructions on how to do this. Once the stream key is there, it won't (or at least shouldn't) change unless someone changes it, so no need to do this every time you stream.

As far as the actual operation of OBS goes, there are two main components: scenes and sources. Scenes are collections of inputs – images, video captures, audio captures, etc. They are what you select as the currently active item and what OBS will send to youtube to stream. Within scenes are the sources – these are what provide the actual input. Generally, we have a pre-show slideshow (the images for which can be

found either from the chief engineer or at my github repository), as well as the main video/audio stream. *Important:* video and audio have to be added separately under the sources panel. Video does not automatically capture audio. Make sure you are selecting the correct device for each input. In the recital hall, this should be "Blackmagic design" for video and "SoundDesk virtual cable (stereo)" for sound. Generally, we also have a transparent Ohio University logo in the bottom right corner. This is a minor detail, ask an engineer for help with this if you have trouble.

The most convenient way to use OBS for our purposes is in what is called "studio mode." This can be activated by the button in the lower right corner labeled "studio mode." When selected, OBS divides your window into two sections: the preview (on the left) and the program – what is being currently streamed (on the right). You can switch between the two with the "transition" button in the middle. The preview allows you to make changes to and select the next scene without the viewers of the stream knowing.

Ok, now on to the idiosyncrasies. OBS has a really stupid feature (or maybe more like a bug) that it only collects input from the first two inputs of an audio interface. This means that if you are using microphones plugged into inputs three through six in the recital hall, they will not be picked up by default. I searched for a solution for months for this, and ended up buying a program that allows you to do some virtual routing of inputs and then delivers it to OBS as a stereo channel. This program is called SoundDesk (remember the weird "SoundDesk virtual cable" thing from the last paragraph?) and acts as a virtual mixing board. It must be open while OBS is running on the recital hall computer in order to get proper audio.

If you are streaming from your laptop, it is likely that you are on site somewhere and only using two mics anyway, so no need to worry about that. However, as of writing, if you are streaming from your laptop it is currently necessary to bring your own interface to stream, or otherwise stream from your phone. This is a work in progress, consult whoever is in charge for the most recent recommendations.

OBS also likes to crash. If this happens, don't panic, the stream will be ok. Just re-open OBS and calmly press "start streaming" again. The stream will pick up again and the viewers at home will miss a little bit of content but otherwise everything will be fine. If you were recording video as well as streaming, it's likely that the recording continued while OBS was closed (not sure how this happens but I've seen it happen). You may be able to re-upload this video later in full. Also – if you are recording video, make sure to press stop recording at the end, or else OBS will not close the video file that it is creating, rendering it unusable.

It can be hard to tell what the audio is like that is coming through OBS to be streamed. Much like how DAWs have input monitoring, OBS has the same feature. It is also really really bad. You can tell where your volume levels are by the meter in the middle of the OBS interface, but this can be deceptive. Aim to make maximum levels at about -3db. In settings, under audio, you can choose a monitoring device. Then, when you have an active audio device, if you click the little gear next to the active audio meter and select "advanced audio properties," you can select "monitor and output" from the dropdown menu. Just be aware that this feature is buggy and often cuts out, continues when you don't want it to, is delayed, and is overall not great. Note: if you're having trouble getting it to work, especially with headphones, try restarting OBS. Obviously you can't do this during a stream, but that's why it is good to get there early.

The audio level that is provided by the interface/SoundDesk is often too quiet as well. OBS provides a gain setting for your audio input that allows you to crank up the volume a little bit so its audible for the audience. This can be tricky, as its easy to overdo it and end up clipping and causing harsh distortion. This is probably my least favorite part of running OBS, but it is manageable. On whatever audio device you are using (with it selected), click that little gear icon next to the meters. Then, click filters, and under "audio" click the plus symbol and select "gain." This will allow you to adjust the gain of the audio input. I have found that a gain of around 5-10 db usually gets the job done. Avoid moving this around too much during the performance if you can help it, as it isn't very smooth sounding and makes things suddenly very quiet or very loud.

Procedure - General

This is a general outline of the procedure for recording/streaming.

Preshow

I like to arrive thirty minutes to an hour early, depending on the complexity of the event. Once you get the hang of things, thirty minutes is more than enough time for your basic Oboe/Trumpet/Piano/etc recital. For anything more advanced – if there's even a sliver of a doubt that things might be complicated – it is good practice to shoot the faculty member/student managing the performance an email beforehand to touch base. Too much communication is better than none at all. Given that you are reading this, you are probably new and faculty won't know to contact you. If you are chief engineer, they will be directed your way, but until then it is good to reach out yourself.

Once you get there, scope out what is going on. Are there speakers on stage? Are there a lot of set changes and movement between pieces? What kind of instrumentation is happening? Is it a solo piece, chamber pieces, or large ensembles? Again, communication is key. All this will determine where you place your mics and how you set up.

After you've figured out what in the world is happening that day, start setting up! Microphones are a good place to start. A stereo pair of mics is a good default for most things, although the specifics of this will be a discussion later. Keep in mind when you place microphones the visual aspect – not only for the camera, but for members of the audience as well. Nobody wants to go to a concert for their child only to have a massive array of microphones placed in front of their face.

If you are somewhere where you need to set up a camera, do this now too. I like to get all my hardware set and in position before I even touch the computer. Make sure you have enough power outlets at your disposal.

At the point when your mics are set up where you like them and they are plugged in to the right inputs, head over to the computer. Open up the DAW you are recording in – most likely Adobe Audition – and create a new multitrack project. Generally Audition will prompt you to do this. Save the file in the following format:

> YY.MM.DD [Type of recital: SR/FR/GSR/VA/USR] [Name of performer], [instrument]

The abbreviations above refer to the following:

- SR: Student recital
- FR: Faculty recital
- GR: Graduate recital
- VA: Visiting Artist
- USR: Undergraduate student recital

Ask an engineer if you are unsure what kind of recital you are recording.

Next, make sure you are getting input from those microphones. Don't be shy about asking someone (nicely!) to play something and talk in front of them for you. Most people don't mind at all; it takes thirty seconds and can spare you a world of headaches later. Use this time to set your input levels on the interface. Ask them to play as loud as they expect to during the performance, adjust accordingly, and then back it off 10%. People always underestimate how loud they will actually play.

If all is good in the DAW, check out OBS. Make sure you are getting video signal, and if not, try to isolate the problem. If the camera is on and otherwise working, it's often just a matter of unplugging/replugging things and restarting OBS in different orders until it works. Check your audio levels too. See above section on livestreaming for more help.

Now is the time to double check everything. Assuming you arrived with enough time to spare, you should have at least fifteen minutes before the show to reassure yourself that everything works and get settled. I recommend taking a sip of coffee at this point. Does the audio come through right? Does everything sound and look like you want it to? Are your microphones horrendously blocking people's view?

This is also the time to post to Facebook. This is easy to forget. The formula looks something like this:

> Join us live at [insert time] for [insert name], [insert instrument]!
> Livestream available here: https://www.youtube.com/c/OhioUniversitySchoolofMusic

> Join us live at 8pm for Roger Rogerson, Flute! Livestream available here:
> https://www.youtube.com/c/OhioUniversitySchoolofMusic

Posts can also be scheduled ahead of time. This is useful if you have multiple concerts to attend to in a day. Google is your friend – "schedule posts facebook page."

If you haven't already, open a browser on a computer that is signed in to the Ohio University School of Music Youtube account. Contact an engineer/manager for access – same goes for the Facebook account. Click on the camera shaped button with a plus symbol in it at the top right corner of Youtube's homepage, and then select "Go live." This will give you the opportunity to name the stream something appropriate. At the time of writing, this looks something like this:

> Ohio University SoM - [Type of Recital: SR/FR/GSR/VA/USR]
> [Name of performer], [instrument]

> Ohio University SoM - SR Roger Rogerson, Flute

You will see a preview of the stream on the Youtube channel. Assuming you haven't started the stream yet, this will be black with a loading symbol.

It is also possible to schedule streams ahead of time. This can be useful if you have many streams in one day. However, you have to be careful – if you try to go live with another stream from the "go live now" section before the scheduled stream, it will co-opt the scheduled stream with all its associated data. In other words, if you are

scheduling streams, **schedule all your streams**. This is a really easy way to mess up your data.

On OBS, assuming you have followed the previous instructions about setting up both a pre-show slideshow scene and the live video scene, make sure you have the pre-show slideshow scene active and displaying in the program section of OBS. You don't want to start a stream with an empty stage showing, it is quite awkward.

Now double check that your DAW is set up right too. Are you *still* getting input to the tracks you have created? If not, go through your signal chain – is everything powered on? Are you getting signal to OBS? Refer to the later *Troubleshooting* section of this book for more help here. Also, double check that your tracks are record armed – are the little red "R" symbols on the tracks lit up red?

Assuming everything is working, now the wait begins. Generally it's good practice to start streaming about five minutes before the performance time. That give the audience at home a little time to be reassured that the stream is in fact happening and also for our slideshow to play for a little bit before the concert. If you haven't done so already, now is also a good time to find the program. As of winter 2021/22, they are all digital, and should be in a Google Drive folder in the School of Music Google account (same as the youtube account). You will need this later.

When you are ready, head to OBS, and click "start streaming" in the bottom right corner, as well as "start recording." *Important:* if you used Youtube's stream scheduling function, make sure you head back to your open browser window and click the blue "Go live now" button in the top right corner. *Youtube will not start streaming without this.* If you did not schedule the stream, Youtube should pick up the stream automatically and simply begin streaming within a couple seconds. Usually the stream is ~10-15 seconds behind real life, so don't be scared if it takes a second to show up. Once this is working, go to Audition and click record. You should see the play head (the vertical line with a triangle at the top) start moving and in its wake the tracks will be red. *Important:* do not hit the spacebar while Audition is recording!!! This will stop the recording with no confirmation and you will interrupt the recording of the performance.

During the concert

If you couldn't tell, the majority of work comes before the concert. Don't fall asleep yet though! As you count down the final moments to the start of the concert, watch the stage for the performer to enter. It is good practice to transition from the slideshow to the live feed in OBS right as the performer takes the stage. They may

check in with you if they are good to go on stage, so use your discretion. At the point when they take the stage, there is very little to do with OBS until intermission besides pray to your chosen deity that it doesn't crash.

During the performance, you will want to mark the beginning and end of pieces and movements in Audition. This will save you literally hours of work later. There is nothing worse than opening a concert recording for editing only to find that you didn't mark the beginning and end of pieces. At first, finding the beginning of movements and pieces can be challenging, especially if you are not familiar with the piece or classical music in general. Your most noticeable changes will usually be tempo changes – if the piece stops and when it resumes suddenly gets faster or slower this is likely a change. Discerning pieces from movements can be challenging at times too. A good audience will only clap between pieces, not movements, so use this to your advantage. However, often someone's grandparent or relative who has no idea will start clapping and ruin this for you, so don't rely on it. Watching the performer can be a help as well – do they totally relax and bow, or do they seem like they are poised to begin playing right away? This is often the most reliable way to tell the end of a piece. If in doubt, mark it, make a note to yourself, and when you get a chance after the performance ask the performer.

Markers can be placed in audition by pressing Shift+M. Be very careful when doing this not to stop the performance with the spacebar. For ease in editing marker names, you can open the marker winding via the Window->Marker Window menu option. Markers should be named in the following format:

> [Track number] [Composer], [Title of piece] - [Movement number and title (if applicable)]

> Ex: 01 Bach, Partita No. 5 in G, BWV 829 - II. Allemande

If you are unsure, follow what the program has to say, but make sure to follow the above format for archival purposes. Pieces and movements should have both a beginning and end marker – this will come in handy later. Continue the marker process throughout the entirety of the piece.

Intermission procedure may vary, but generally it is a good idea to at at least mute the audience noise for the stream. Depending on current procedure, there may an overlay, slideshow, or short video. Ask an engineer. Note: do not stop the stream or recording during intermission, leave everything running.

Once the concert is over – and really, for sure over, with no more encores – I like to

fade to black on OBS with the "fade to black" option in the middle of the screen. Wait five-ish seconds, then click "stop stream" and "stop recording." It is very important that you do both of these things, otherwise OBS will hijack the School of Music Youtube stream indefinitely. Now you can also hit the spacebar on Audition to stop the recording. *Save your work.* And then save it again. You don't want to lose this file.

Post-show

After the show is over, it's generally best practice to wait a little bit before you storm into the audience to grab your microphones/other gear. People tend to mingle, and you don't want to make people feel like they are unwelcome. However, you may also use your discretion with this – some people tend to mingle for a *lonnnngg* time – there is nothing wrong with telling people that you need to lock up at a certain point and that they may continue to mingle in the lobby.

Coil your cables nicely, pack up your microphones into whatever you brought them in, fold up your stands, and make sure you're not leaving anything out. If you're in the recital hall, turn off the Furman power conditioner and interface. There really isn't much to do at this point if you are in a hurry, so feel free to leave and do the editing later.

Editing

Generally people like to have their recordings within the week. I try to get them done earlier, but obviously your time is valuable too. Just don't leave anyone hanging, and be communicative.

For all its faults, this is where Audition shines, so I will go through the process for editing in Audition.

With your multitrack session open, go ahead and double check that the markers for pieces are all in the right place. If in doubt, it can help to pull up the piece on Spotify/Youtube and verify that the beginning is in fact the beginning. Pieces and movements should start with about two seconds of silence. Movements with no applause should end with about 3 seconds of silence, and movements/pieces with applause should end with about 10 seconds of applause. Check by listening from the beginning of each marker that it sounds right.

If the markers all check out, click on the "mixing" or "mixer" tab at the top left. This is a highly subjective and complicated subject, so I will keep this extremely minimal – refer to the *Best Practices* chapter for more information. At the very very least, a low cut around 50-80hz is almost mandatory. This is not complicated but sort of outside

the scope of this section, so either Google or later chapters are your friend here. All I will say is don't overdo it. Classical music is (usually) about the performances as is, not your fantasies of crazy effects. If you have multiple tracks (say from multiple microphones), adjust the balance to your liking.

When you are satisfied with the mix, head back to the "waveform" tab at the top left corner. Assuming the markers are in the right place, open up that marker window again (Window->Markers). Click on the first marker and then command-click the second marker to highlight it as well. At the top of the marker window, there will be a little button that creates a range out of the selected markers.

Repeat this process for each of the beginning and end markers for each movement/piece. At the end, you should have half as many markers as you started with, as they were converted to ranges. Return to your first range marker, and double click it. It should highlight the first range in the waveform window. Double click the waveform itself between the ranges, and if you have multiple tracks, command-click the remaining tracks within the same range. They should highlight to indicated they are selected. Press command-k to cut them along the range marker lines. Rinse and repeat for the remaining ranges.

Returning once again to the first range, select it along with any other ranges on other tracks in the same spot, and apply a cosine fade. This is done by command-clicking the small dot at the beginning and end of the range. See image below:

Fades should be about 1.5 seconds at the beginning of pieces/movements, 2-ish seconds at the end of movements with silence, and 3-4 seconds for movements/pieces with applause. Use your ears to make sure it sounds graceful and not awkward. Rinse and repeat, and blow dry just to be safe.

Finally, select all your cut tracks (excluding the parts between pieces) and go to the "Mixdown" menu. Select "Mixdown Session to New File" and then "Selected Clips." This will create what is called a mixdown – a single audio file that has all your changes to the original audio written as a new file. This includes all the changes you made in terms of EQ, compression, reverb, and balance between tracks. There should be a small knob somewhere on the track waveform itself (it is floating, small, and moves around – you may need to google "amplify knob audition mixdown" for an image). Use this knob to amplify your session to its final volume. We call this normalizing. Keep the highest volume points of the waveform at or below about -2.5db to -3db. Once this has finished processing, you will be ready to "bounce" the tracks. This is the term for creating a final export of the tracks into audio files that are readable by

programs like iTunes.

To bounce the tracks, go to the File menu, and select "Export" and then "Audio within Range Markers." This will create a popup window with some options. Make sure that you are exporting to 48000khz with 24 bit depth, and that "use marker names in filenames" is selected. Also be sure you are exporting as a Wave PCM (.wav) filetype. Select the folder that you created the Audition project in as your destination, double check everything, and then click export.

At the end, you should end up with a number of tracks in the same folder you created your session in. As for archival procedures, ask the current engineer where these tracks should go. Some visiting artists like to give you their email to share the tracks with them – at the time of writing all students have near-infinite storage in OneDrive. Tracks can be shared directly with artist/faculty/students this way if all else fails.

That concludes the general procedure – it sounds complicated, once you do it in real life a couple times it will become second nature.

Below are some notes on the various spaces that you may find yourself recording in.

Procedure notes - Recital hall

This is the space you generally have the most options in. Stands, cables, and microphones are all at your immediate disposal, so don't be afraid to experiment somewhat.

The current recital hall setup requires that SoundDesk (the application) be open during streaming to feed audio to OBS. Consult the OBS documentation above for more info. SoundDesk will allow you to create a decent mix for the stream, as well as handle multi-microphone input for OBS.

Generally, I find a stereo pair of microphones in either A-B or ORTF configuration (more on these later) to be more than sufficient in this space. If you have to put microphones on stage, always ask the performer if they are ok with this.

If you need to do playback from the studio computer into the hall (for example, for the pre-show safety message), this can be done, but it is a little complicated. Outputs three and four are routed out to inputs nine and ten on the stage, but go through a mixer. (NOTE: we should figure out a better way to do this before I write this).

At all times, there should be a backup drive attached to the recital hall studio computer. Make sure recordings/audition sessions are backed up here. Remember, a digital file only exists if it exists in two places.

Procedure notes - Memaud

At the time of writing, Cullen Beach is technical director at Memaud. He is knowledgeable and will be able to help you with almost any question about the space if you are communicative. Consult him for the most recent configuration in Memaud, it changes frequently and tends to be a little complicated.

He can be reached at beachc@ohio.edu. He's very busy, so don't overburden him – he'll get back to you.

Memaud tends to have multi-camera setups. The specifics of this are likely to change, but just remember that you have options here for some cool videography during the show.

You will also need to access the stream key to confirm that their computer with OBS on it is configured to stream to our account. Refer to the above OBS setup section for more info.

While Memaud has pre-set cameras, you will need to bring our microphones/stands. They have plenty of cables, so don't worry about that. Setup here is usually pretty easy – the fifth row is a good place for microphones as it is wider than the rest and a good distance from the stage.

Procedure notes - Methodist Church

This space can be finicky, but there is a booth to set up with power. You will need to bring a power strip, an extension cord, plenty of XLR cables (the run from the booth to the front of the stage is around 100 feet), an interface, a laptop, camera/tripod (with all its accessories – don't forget the HDMI cables and HDMI-USB converter), microphones, and a stand. Usually only choral concerts are held here, two microphones and a stand is more than enough.

Microphones can be set up directly behind the directors podium, generally in ORTF configuration (again, refer to later sections for what that means).

There is a wifi network, ask the choral director for more info on how to connect to this.

Procedure notes- Field recordings (Galbreath, Scripps amphitheater, College green, etc)

These spaces are in some ways the most fun and also the most annoying.

Bring everything you could possibly need. This includes power strip, an extension cord, plenty of XLR cables, an interface, a laptop, camera/tripod (with all its accessories – don't forget the HDMI cables and HDMI-USB converter), microphones, stands, ethernet cables, ethernet-usb adapters if you need one, and any spare items you might think of. Bring plenty of water/coffee/tea as well, it really helps.

Galbreath has an ethernet port for high speed internet, it is located behind the door to the right of the organ. There are tables and chairs in the basement, this is a small comfort that goes a long way and looks professional.

Concerts on the green are difficult due to the spotty internet. Usually Cullen Beach from Memaud is involved (see above), contact him if you need anything. Often streaming from your phone is the best option here.

Scripps amphitheater is similar to the green, but has good enough wifi for streaming and plenty of power if you bring an extension cord. Be communicative with the organizer/faculty member in charge of the program so you are prepared for whatever is happening here.

Chapter 4: Best Practices

Congratulations! If you've made it to this part, you've graduated from being a mere instruction reader to becoming a real artist! Yay!

This section will illuminate some of the more cryptic terms and practices that I've vaguely referenced throughout this book so far. As always, this section is in no ways comprehensive. It's a starting point meant for you to continue your own research and experimentation. And at a certain point, no amount of reading will teach you as much as simply doing recordings. Use this as a guide, not a bible. Your ears are your best friend with this. Learn to trust them. If something sounds wrong, try to figure out why. If you are struggling to even discern what is happening, take a break. Fresh ears are essential. Listening fatigue is a real thing, and can be extremely disorienting. You'll build up a tolerance, but even the most experienced engineers take breaks.

Sound organization

I'd like to start this guide off by encouraging you to be organized. Actually, encouraging might be too weak of a word – demanding might be better. Organization is essential in any recording situation, and doubly so when it is live. You will need to respond quickly to any challenges that might arise, and the last thing you want to be doing is scrambling to figure out which microphone is going where because you plugged everything in illogically. It may feel silly, but don't be afraid of labeling things and taking notes. Keep track of what microphones you're using, where they are going, and what programs are open. Where is your sound going – could you explain piece by piece and quickly the signal flow of your setup? This is essential to any setup.

As you continue to work, you will develop some kind of routine. Encourage this, and nurture it. The more you do things the same, the more effective you will be in responding to problems. An mediocre engineer who knows their setup is better suited to live recording than a great engineer who has no familiarity with the situation. Your setups will become faster and more efficient, and not only will this aid you in creating good recordings, but you will look more professional as well. People trust an engineer who comes in is quickly ready to record, and doesn't spend forty minutes fighting with their software. Not that this won't happen to experienced engineers, but if you know your setup you will start to know what tends to go wrong and be able to fix it faster.

It is also good practice to learn how to coil your cables right. This keeps them both

aesthetically pleasing and also prevents wear. *Never* coil by simply wrapping the cable around your hand and elbow like you would a rope. Consult an engineer if you don't know how to coil – they will be able to teach you – or look up "over under coiling technique" if you'd rather learn on your own. It will feel awkward at first but soon become second nature.

Finally, keep a journal or at least notes about what kinds of microphone techniques you used on which recordings. Even if you don't revisit them, being mindful of what it is you are doing will train your brain to know what kinds of setups are appropriate for which situations. While reading about techniques can be a good starting point, there is no replacement for experience and critical listening. This is its own kind of organization – how are you going to remember what setup you used if you just threw something together and forgot about it immediately? I often name the tracks in my DAW with information about the setup. For example, I might label a track "Neumann km184 ORTF lip of stage" to indicate the type of microphone (Neumann km184), mic setup (ORTF), and placement in the room (lip of stage). When I return to my session to mix or for any other reason, I am constantly reminded of how I made that recording, and am forced to associate the technique with the sound. This is only one way of many, but find a way to make this happen for you, and you will quickly find yourself much more familiar with the way certain setups sound.

Critical listening

Over the course of this book, I have encouraged you to use your ears – listen to rooms, listen to recordings, etc – and to learn to associate these things with their sources. I want to further encourage you to listen to the world around you. Take a walk, listen to the various sounds. Can you pick them out? Not just one or two, but really everything. Cars, lawnmowers, birds, people talking. Listen for the soft details, the low rumbles, the strange high pitched noises. Drive yourself a little crazy listening to everything at once – dive deep into the sounds until you can't take it and have to come up for air.

This sounds silly, but it really does help. It's such a simple daily practice, but the fact that you can do it literally anywhere at any moment means there's really no reason not to. When you get to recording, start to do the same thing. When you enter a room to record, take note: are there air conditioners making noise, low rumbles, high pitched whines from the lights? Hint: the recital hall has a couple of these, sometimes more prominent and sometimes less. These are considerations when recording and when placing microphones.

Also, listen to lots of music! Get yourself a decent pair of headphones, take a walk, and listen intently to a recording. What sounds good? Does anything sound off? Can you tell where the microphones were placed? Do some research, and see if you can find the answers to some of these questions once you've made some guesses. All this will help align your ears to what is really there.

One last thought: there is a book called *Drawing on the Right Side of the Brain.* In it, the author instills in the reader a simple concept: draw what you see, not what you think. She uses the example of drawing faces – many people draw an eye or a face based on what they believe to be important in a face. It is analytical, rather than perceptual. As a result, people draw an eye that looks nothing like a real eye, but rather a confused copy of one. I think this is applicable to music as well. We often think we know what something sounds like, and then go off of a half-formed idea of what it truly is. Learning to truly listen is exactly the same process. If you can identify what is really there, you will have a much easier time learning to reproduce it.

Microphone techniques

If you're like me and have spent any amount of time on obscure corners of the internet trawling through forums, you'll know the exact way that people tend to become quite opinionated about certain subjects. Microphone technique is one of the most secretly contentious subjects out there – there are about as many techniques as there are engineers. However, there are a couple of more standard choices and guidelines out there that I will attempt to do justice to. As you progress, you will form your own opinions, and you may even think I'm completely wrong. I encourage you to write me and tell me why I'm wrong if you come to this conclusion.

For this section, I will focus only on condenser microphones. Despite some engineers probably using a dynamic microphone on some classical recordings, it is a very unusual practice. I would wager that 99% of the recordings you will ever make of classical music will be with condenser microphones. And I would further wager that another 90% of those will be made with a single stereo pair of microphones – uncomplicated, but effective.

If you already know a thing or two about recording, I still want you to read this section. In fact, I especially want you to read this section. Chances are, unless you came from a really strange high school, any recording experience you have is probably with bands/singer-songwriter types, not classical. The approach is quite different, although the physics remains the same.

Microphone basics

Wait, didn't we already do this section? Sort of... we talked about the physics and technical side of microphones, but have yet to address the basic functionalities of microphones themselves.

Within the world of condenser microphones, there are a few sorts to consider.

All microphones have a directionality, what is referred to as the "pickup pattern." This can be graphed and modeled, but ultimately simply refers to the directions from which the microphone picks up sound. Some microphones are omnidirectional, meaning they pick up sound equally from all directions. Others pick up sound with more discretion, only in one general direction. Some microphones, like the shotgun microphones commonly used in TV and film, are so highly directional that their pickup pattern is closer to a spotlight than anything else.

Common pickup patterns include:

- Omnidirectional (all directions equally)
- Cardioid (directional with about 120-180 degree spread)
- Hypercardioid (like cardioid, but narrower)
- Figure-8 (pickup on front and back but not sides)

Looking up microphone pickup patterns will yield hundreds of images visualizing their patterns – do this now.

Besides simply being useful for avoiding sounds you don't want to pick up – audience noise, for example – pickup patterns also have an effect on the sound. As a pickup pattern becomes more focused, it emphasizes frequencies based on its location. This is especially noticeable in the lower end of the frequency spectrum. Known as the *proximity effect*, directional microphones will emphasize lower frequencies the closer you put them to the source. Conversely, they will drop lower frequencies past a certain point. This can be used to your advantage, but also can cause problems. Put too close, a source can sound boomy and tubby, too far and you lose the richness of the bottom end.

There are also two major types of condenser microphones: small diaphragm and large diaphragm condensers (often abbreviated SDC and LDC). At Glidden hall, we own some of both. The AKG 414 microphones – the larger black and silver mesh microphones with the switches – are large diaphragm, while the sE7 microphones – the small and thin gray ones with the red square on them – are small diaphragm. Generally, SDCs are said to capture more detail and are more "realistic" than LDCs. This

is due to their membrane physically weighing less, as well as their body imparting less of an effect on the sound. Yes, the body of the microphone shapes the sound: it too is a resonating surface. LDCs are well suited to a "larger" sound, with more bass and more *oomph*, so to speak.

Some microphones – like the AKG 414 – allow you to change pickup patterns. This can be useful, but also a source of confusion if you forget to check what it is set at. Keep this in mind.

Placement

Let's go way back, some thirty-odd pages ago – do you remember the first "real" recording advice in this book?

Let it breathe. Here we are again. I want to reiterate this point, only because I think it is so essential to so much of classical recording.

Think back to what we've learned about the way that rooms affect the sound within them, and the way that microphones pick up sound: when an instrument produces a sound, that sound is reflected and absorbed by all the various walls and materials and objects in a room. In some ways, it is useful to think of microphones as ears on a stick. They are not quite the same, but they are similar enough for this analogy. Imagine you put your head way up close to an instrument – like, inches away. What would this sound like? Next time you are near an instrument, try this. See what it does to the sound. You'll probably find you are hearing things you wouldn't normally hear when listening to the instrument. In a piano, you might hear the pedals moving, or the hammers initial attack on the string. You will also lose some of the body of the sound – low frequencies need space to complete their cycle. Now move back from the source – suddenly your ears start to pick up more reverberances, less detail and more whole. This is how you normally experience music, standing in a room, hearing sound from the source in front of you.

I want to be clear here: there is nothing wrong with close micing techniques. Used artfully, they can provide details that placements farther away cannot pick up. Sometimes close techniques are combined with room mic techniques to provide a balance of both. This is all open to experimentation, and I encourage you to do so. However, on the whole, classical music is a live endeavor. Think of yourself as in some ways a kind of anthropologist, seeking to record the true and accurate experience of being at a live concert. Especially given that many of the recordings you will likely be doing for the School of Music will be accompanied by video in a livestream, it is important that the recordings give a sense of space. Micing farther away emphasizes the sound of a

room in a natural way that's coherent with the live experience.

Microphone height is another factor in making recordings. You want a stand that is able to easily put microphones 10-15 feet in the air without feeling like it will fall over. Different scenarios will call for different heights, so experiment with this. As a general rule of thumb though, your microphones should be no lower than 4-5 feet off the stage. Given that most stages are about 4-5 feet off the ground, your microphones will usually end up being around 10 feet off the ground. Too close to the ground, and you will end up with strange reflections, too high and you will start to lose definition in the sound of the room.

Placement is not just about room and height though – it's also about what we call the *stereo image*. This is a metaphor that we use to describe the way that audio feels in stereo. Any given stereo recording simulates to your brain what it is like to be in a space. The balance between left and right of an instrument or set of instruments where in space you believe that sound to be coming from. In the next subsection, I will go over various techniques to achieve this effect.

Stereo techniques

As almost every recording these days is in stereo, it is imperative that these techniques become second nature. Out of all the technical jargon that you could learn, this might be among the most important. These techniques have some goofy names, usually after the inventor of the technique or after the shape of the configuration. I will go over in brief what each technique achieves, its strengths, and its use cases. In any stereo technique, at least two microphones are required to achieve the desired effect. Each microphone is then panned left and right in your DAW to achieve the desired effect. Attached with each technique is a diagram of the proper setup (not to scale). [3]

A/B: Perhaps the simplest stereo technique, this configuration is simply two microphones placed parallel to the sound source with a distance between them. More descriptively, this technique is also known as a *spaced pair* of microphones. See fig. 4 for placement. This technique is usually achieved using omni-directional or cardioid microphones. Omnidirectional microphones will pick up more of a bass response, but keep in mind – they are omnidirectional, which means that any and all audience noises *will* become part of your recording. This technique can also be more visually obtrusive as it generally requires two microphone stands. As a rule of thumb, the dis-

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[3] Diagrams from https://www.thirdcirclerecordings.co.uk/wp-content/uploads/2020/04/blog-image-9.jpg, https://isaacfmp.files.wordpress.com/2015/04/ab_stereo-svg.png, and https://lossenderosstudio.com/img/decca-tree.jpg

tance between the microphones should be about three times the distance from the source to the microphones. As the distance between the microphones grows, as will the "width" of the stereo image. This technique is susceptible to phasing issues, so take your time when setting it up. This technique can be useful in situations where you have a wider spread of instruments, such as medium sized ensembles.
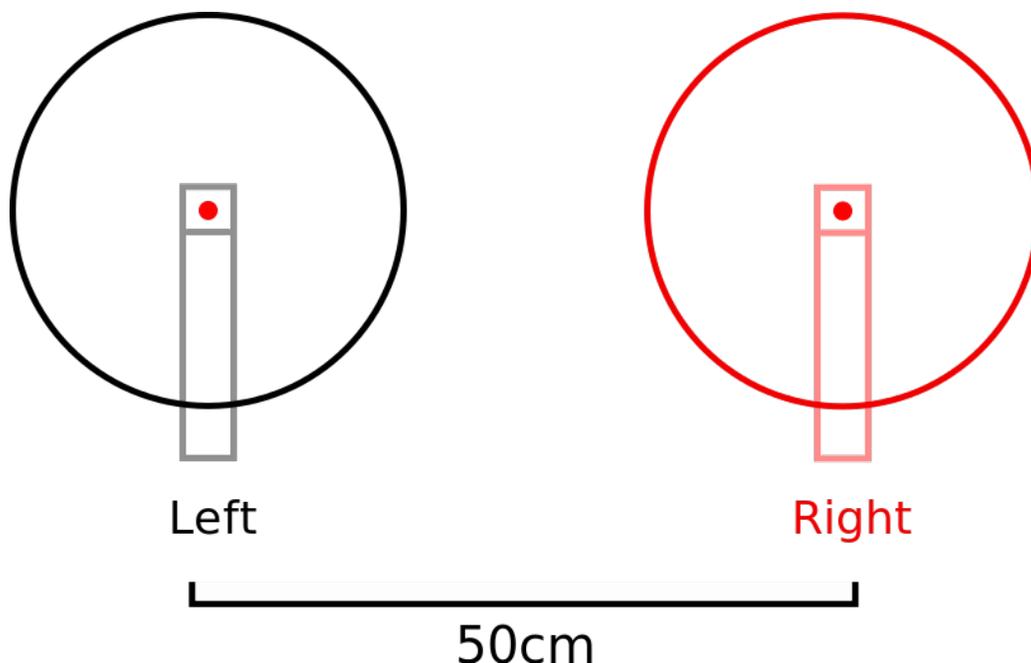

Figure 4: A/B setup

X/Y: Another of the most common techniques, this one is more commonly used when close micing instruments. It is unlikely that you will use this to record classical ensembles, but can be useful when recording solo string instruments. As the name suggests, two cardioid microphones are placed in an X/Y-shape, as shown in fig. 5. These microphones are usually mounted on something called a *stereo bar,* a small bar that has two microphone mount points that can be mounted on a single stand. This technique has the advantage of being easy to manage in terms of phase – the close placement of the microphones solves this problem. This technique is usually achieved using small diaphragm condensers, in part simply due to the need to use small microphones to put them close enough together.

ORTF: If you are to know any stereo technique well, let it be this one. The most versatile, best sounding (in my opinion), and overall most effective micing technique,
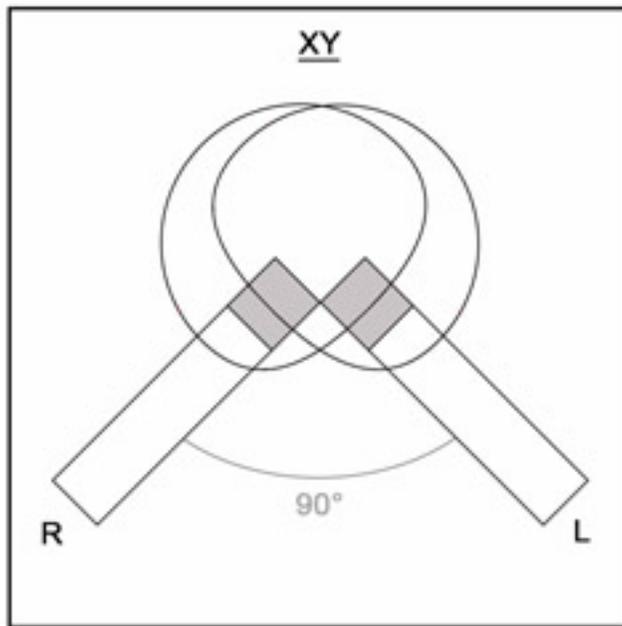
38

Figure 5: X/Y setup

ORTF can be used for everything from solo performances to large orchestra and jazz ensembles. Two microphones are placed at a 120 degree angle to one another, facing outward, with the distance between the tip of the microphones (the capsules) being approximately 17cm. See fig. 6. In this configuration, as with X/Y, the microphones are mounted on a stereo bar on a single stand. This technique is named after the French radio company that invented it, the *Office de Radiodiffusion Télévision Française.* This technique is highly flexible in terms of distance from the source and the size of ensemble. Again, this technique is largely used with small diaphragm condensers, although large diaphragm condensers are by no means unusual. ORTF is my go-to technique if nothing seems out of the ordinary – orchestras and string ensembles sound fantastic with this setup, and it's pretty forgiving in terms of phase issues. Due to only requiring one stand, this technique is also nice visually. I can't emphasize this enough – *learn this technique*.
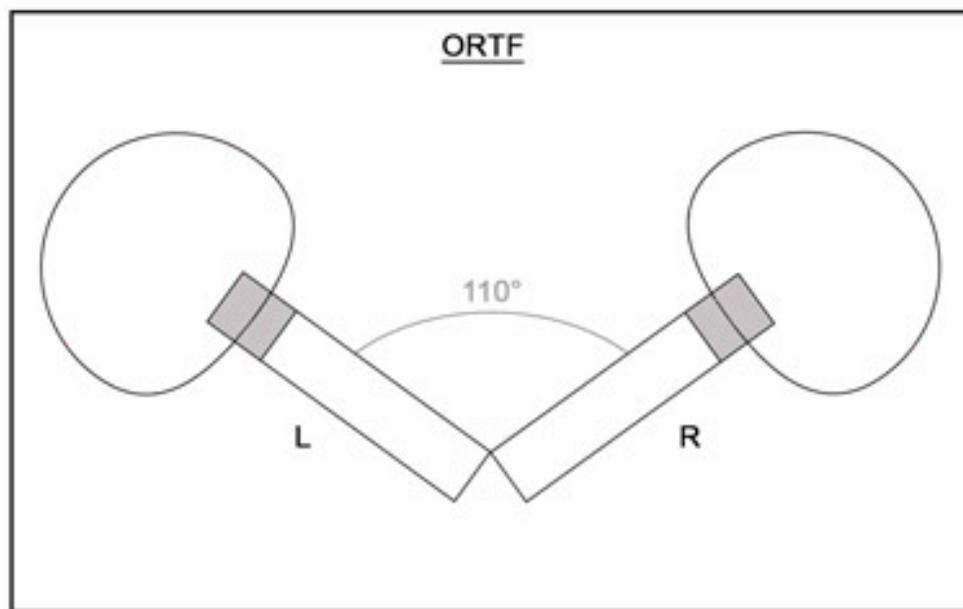

Figure 6: ORTF

Decca Tree: This and the following two techniques are less usual, although still important to know. The Decca tree technique was developed by Decca studios in the mid 1900s as a means of recording large ensembles such as orchestras. In essence, it is an A/B microphone setup with a third microphone placed in the middle slightly

to the front to fill in the middle section – in regular A/B configuration, it is easy to lose the middle of an ensemble. Decca trees were designed to mitigate this effect. Unfortunately, Decca tree mounts require a special (and very expensive) mount, so it is unlikely you will come across this technique unless you are in a more professional studio. See fig. 7 for visuals.

Mid-side: Unlike the Decca tree, mid-side configuration requires special mounting hardware to achieve. However, it does require a microphone that supports a figure-8 configuration. The AKG-414 microphone is commonly used for this, as it supports multiple pickup patterns. This might be conceptually the most difficult to understand of all the stereo techniques. To achieve it, a standard cardioid microphone is placed facing the sound source. Directly on top of or below this, perpendicular to the cardioid microphone, a figure-8 microphone is placed. This might be a little hard to visualize, see fig. 8 for details. The caveat with this technique is that it requires special post processing to create the final stereo effect. This is not difficult, but is beyond the scope of this guide – there are plenty of guides to doing this processing on the internet. The advantage of this technique is that there is almost no chance of phasing problems, especially if the recording has to be converted to mono at any point.

Blumlein: This technique is very similar to X/Y technique, but instead of cardioid microphones uses figure-8 patterns. The two microphones are placed one on top of each other at a 90-degree angle, with one microphone picking up the front left and back right of the room, and the other the front right and back left. This technique produces a highly spacialized sound, much more than X/Y. Blumlein combines the phase mitigation of X/Y with the stereo spread of something like ORTF, and thus makes a very good choice for recording in situations where room ambience is important. Because it often requires two stands to set up, however, it is not particularly visually appealing. See fig. 9 for details.

MUSICAL
ENSEMBLE

CENTER
MIC

3 ft

LEFT
MIC

RIGHT
MIC

6 ft
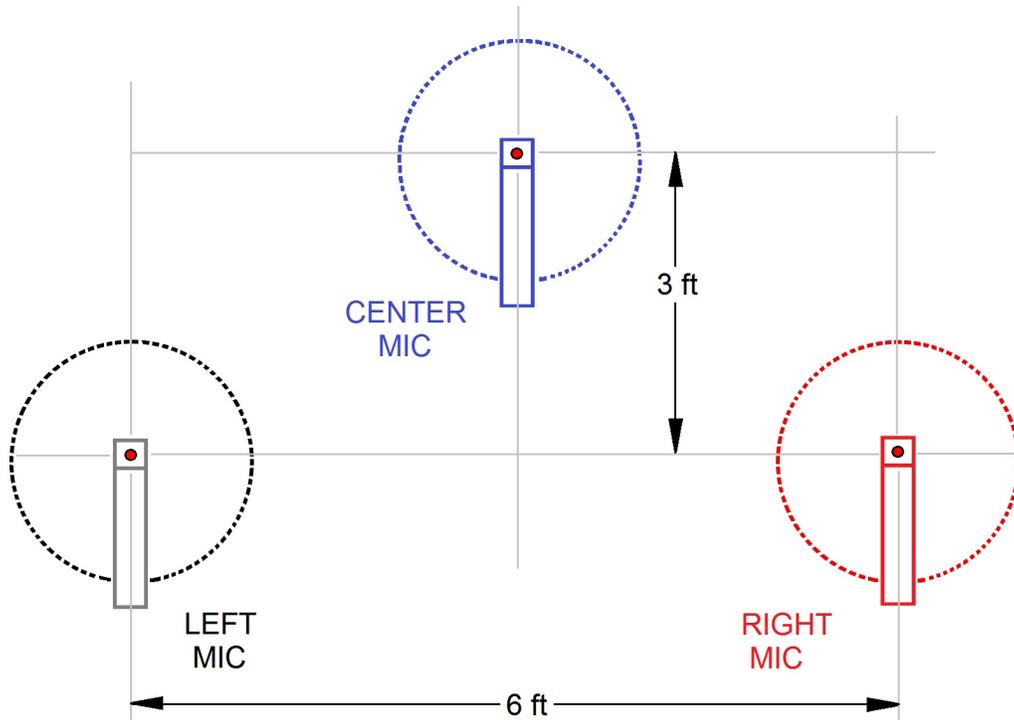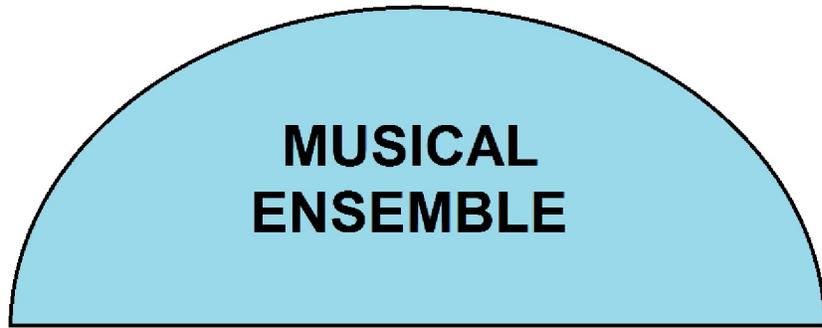
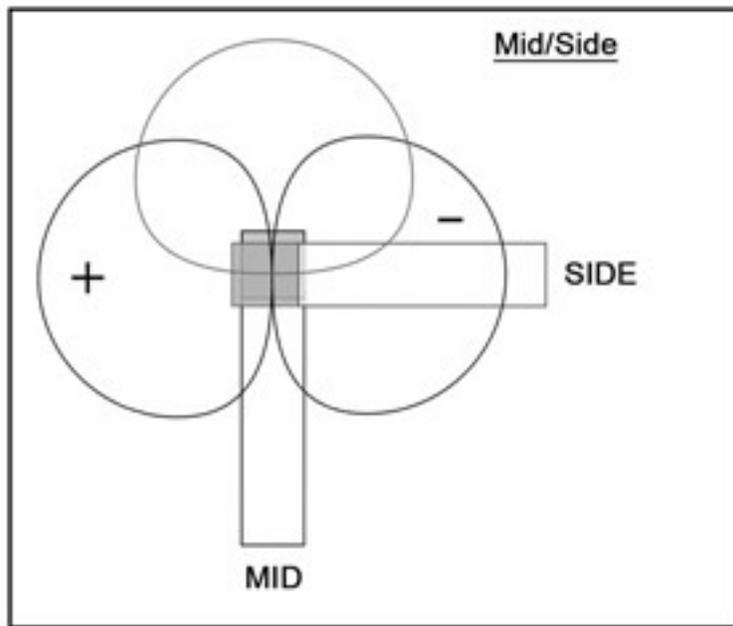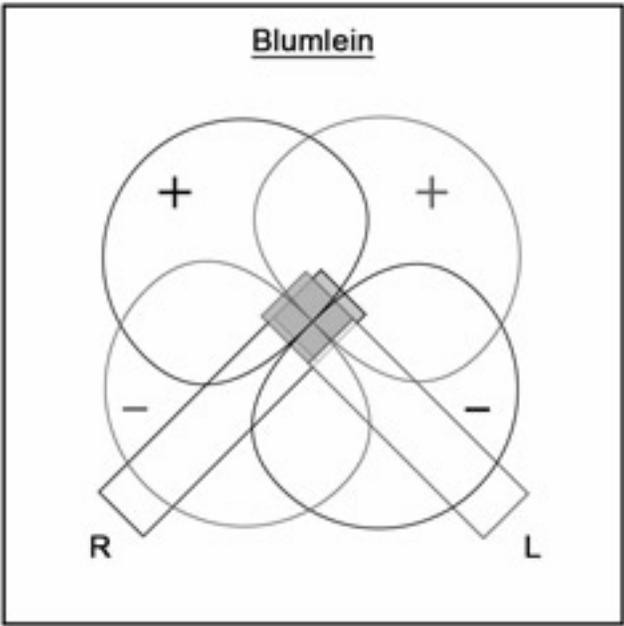Decca Tree

Figure 7: Decca Tree

Figure 8: Mid-side

Figure 9: Blumlein

Microphone choice

How do I know what microphones to use? There are hundreds, if not thousands, of microphones, brands, models, and designs. Fortunately, they are expensive, so it unlikely you will have more than a couple to choose from. Even so – what if I'm buying a microphone??

I will again focus on condenser microphones here again, simply because dynamic microphones are hardly used for classical recording.

Let me start with something a friend said to me years ago when I was first contemplating buying a microphone – *buy quality, borrow quantity.* What this means is that you are far better off spending money on just a few higher quality microphones than lots of lower quality microphones. It is likely that if you were to need many microphones for a project that there would be someone or somewhere you could borrow or rent a larger quantity. I feel that this mentality also speaks to the necessity of having a good source – putting ten poor-quality mics on something is so much worse than a single quality microphone with good placement.

With that said, what makes a good microphone? This is a complicated question, and a lot has to do with extremely technical specifications about the build of the capsule and other internal components. As a rule of thumb, stay away from cheap Chinese brands and microphone "kits." They are tempting, but it really just isn't worth it. Companies with very good reputations include AKG, Neumann, Earthworks, Sennheiser, and Shure.

Look for features that might be useful for your recording scenario. As an example, the AKG 414 has a selector switch allowing you to switch pickup patterns. This can be very useful in a field recording scenario where you may not be sure what to prepare for. Short of having this kind of feature, think about go-to microphones for many scenarios – a good pair of small diaphragm condensers can fill this role very well. The Neumann KM184 microphones are a well-established choice in this realm – although I'm sure many engineers would disagree with me and have their own personal favorites. In fact, any matching pair of higher-quality cardioid condensers is a good addition to a microphone cabinet.

In terms of specifics, how do you know which microphones to use on which performances? This is a question that doesn't really have a concrete answer, and I can only provide very vague guidelines. As said before, large diaphragm condensers tend to do well with louder sounds, but have slightly less detail. Think jazz bands, brass ensembles, sometimes piano, and anything big and thumpy. Small diaphragms deal

well with strings and small sounds. Think chamber music, string quartets, solo flute, and sometimes piano. Other than these two very general recommendations, you are going to have to use your ears to get to know the microphones you have. In my experience, the AKG 414 has a more present high end, while something like the sE7 is a little more mellow and flat. These are both microphones the School of Music owns at the time of writing, but take even this recommendation with a grain of salt – they may sound different to you.

Post-production

Finally – welcome to the world of post-production. Ever heard someone say they will "fix it in post?" Post-production includes all the editing and processing of the audio that you recorded (and hopefully saved...) Some find this process rewarding; others find it tedious and annoying. If you did your job right during the recording, it should be at the very least painless. This process is sometimes also known as mixing, although this is a sort of misleading term in classical music recording.

Take frequent breaks, trust your ears, and allow your mixes to stew for at least a day before finalizing anything. It is easy to get lost in audio and have everything just sound like mush. Whatever effects you apply, bring them up to where you think they should be and then dial them back 30-50%. You very likely are over-processing. With that in mind, here are the most common post-production processes...

EQ

So what does *EQ* mean? EQ stands for equalization – the balancing of the frequency spectrums in an audio source. Remember how each pitch has its own frequency? As it turns out, the volume of the various areas of frequency can be balanced after something is recorded through features in a DAW. This is also possible through analog gear, although I will not get into that here.

You may be wondering why we would do this – after all, the microphones pick up what is there, and the instruments should already sound good by themselves? There are a number of reasons for using EQ, but ultimately its almost always about cleaning up the sound. It is true that a microphone records what is there, but there can be times when there are undesired frequencies captured. Depending on the room, there may be some frequencies that seem over-exaggerated due to reflections off the walls and floor. Or, there may simply be an AC unit somewhere nearby producing a low hum that you would like to remove.

The first rule in using EQ is *don't overdo it.* It is easy to deceive your ears into thinking that you need to boost or cut a specific frequency way more than you actually do. In fact, it's best to avoid boosting frequencies – most often if you think something needs to be boosted, it is actually that a competing frequency needs to be eliminated. This will go far to clean up your sound.

If there is one thing that you should almost always do without fail, it is applying something called a "high pass filter." Also known as a low-cut, this eliminates the frequencies below a certain threshold. Generally, around 80hz is a good cutoff point, as almost nothing plays notes lower than 80hz. Of course, use your judgement – it is possible to go higher in the 150hz range if recording something higher like solo flute, and at times necessary to go lower when recording electronic classical music. Cutting those low frequencies will aid your mixes in sounding clean and professional.

As far as other specific frequencies to look out for, if something sounds boomy, it is likely 200hz. Cellos like to make strange noises around 800hz, and violins and violas can get harsh in the 2khz (2000hz) range. These are all just generalizations though, and the only way to know what your mix needs is to use your ears. Unfortunately, frequency identification is one of the hardest skills to acquire, and really just takes practice to fully understand. Experiment, and when in doubt you can cut or boost a frequency really dramatically to see if it seems like the offending contestant – they made the undo button for a reason! And though EQ is a huge topic, nothing can replace experience and familiarity with frequencies. There are countless website purporting to offer secret EQ tricks, but your ears are as always your best guide.

Compression

If EQ is peanut butter, compression is jelly. The two live in harmony and play very well with another. Between the two, you are likely able to fix 99% of the problems a recording may have. Clever usage of the two in combination yields recordings that sound like those you would hear on a professional release. You would be surprised how many recordings make use of very little of anything else.

Compression, in a very brief sense, mitigates overall volume. When things get too loud, compression steps in and puts on the brakes. Imagine a pipe with a certain diameter connected to a water pump – up to a certain point, the water can flow freely as fast as it is being pumped, as it has not yet filled the pipe. In fact, up until the point that the pipe is completely full, the water is completely unrestricted. Once that pipe is filled, the water is restricted, and not all the water can flow through at full force.

This is roughly how compression works, except that in the world of compression the

diameter of the pipe is adjustable. We refer to this as the threshold – how loud must the sound get before compression starts to put a limit on the volume? This parameter is combined with a number of other parameters: attack, release, ratio, and something rather stupidly called "knee." Attack is how quickly this volume cap kicks in when it hits the threshold, and is generally measured at the scale of milliseconds. Release is how long the compressor waits before lifting this volume cap. Ratio refers to the amount by which the volume is reduced – compression is not so much a wall as it is a recommendation. A higher ratio will limit the sound by a larger factor. And knee – well, it's not a super relevant term, but it is how we measure how exact the threshold level is. This level can be a sharp kink, or it can be a curve – don't worry about this one too much.

These parameters all affect the sound differently. A faster attack will cause the sound to be punchier and heavier, while a slower attack allows the sound to breathe more. A high ratio will sound more processed, similar to what you might hear on pop vocals on the radio, while a lower ratio again allows the sound to breathe more. Can you guess how we like to mix in the classical world? The number one rule of compression is *don't overdo it.* Are you sensing a pattern here?

Compression being another subject that is a book in itself, I will offer only a couple of words of advice here. Often times, compression is hardly necessary in classical music recording. I will use compression when it feels like the sound is begging for something a little harder hitting – a dramatic brass choir comes to mind. Light compression can also be useful on quieter pieces to bring out some of the detail. Be careful though, as it is easy to end up obscuring detail through compression with too fast of an attack and too high of a ratio.

A brief note on compression – there is something called a dynamic equalizer. It is essentially an all-in-one EQ and compressor that allows you to apply compression to only specific frequencies. It is very useful, and in some ways the best tool for the job in classical music. There is a plugin for DAWs called TDR Nova that is an incredible dynamic equalizer and it's free. I recommend downloading it and playing with it – I never leave the house without it.

Reverb

Of all the processing techniques you might know of, reverb is probably the most easily recognizable by an untrained ear. Because of this, reverb is also easily the most over-applied of any effect.

Reverb comes in two main forms these days: natural and artificial. Natural reverb is

what we've already discussed – the sound of a room. All enclosed spaces have some amount of naturally occurring reverb. If at all possible, this is normally the desired kind of reverb in classical recording. However, there are always situations in which the room has no natural reverb, or at least has very little. In these cases, it might be advisable to use artificial reverb to smooth over the space slightly. There are many plugins in DAWs that will achieve this, and I won't go into detail about that here, but keep in mind that you should never feel overwhelmed by reverb. You are creating a sense of space, not scribbling all over the performance.

A note on signal flow: In post processing, always EQ before compression, and always compress before reverb. You are free to experiment switching the order of these things, and in some kinds of recording this can be used to creative effect, but generally switching the order of these things will provide undesired (read: nasty and ugly) results.

Gain staging

Quick note on terminology: gain refers to input signal levels, while volume refers to output signal levels.

Moving backwards in our signal chain here, lets return to the source – your microphones and pre-amps. At multiple levels of your signal flow, there is a chance to adjust what is called gain. This is essentially a volume control for an individual piece of equipment. The most common place to adjust gain is on the pre-amps of your interface. This is by and large the most important one to get right, as it is sort of permanent in the recording.

When recording, you want to avoid something called clipping. This is when the signal volume becomes too loud and overwhelms the equipment, causing nasty distortion. Avoiding clipping is one of the reasons getting a quick sound check in is actually really important. On your DAW, you should be able to see meters of the input levels – consult your DAWs manual if you have trouble finding them. Metering is done in negative decibels, with 0db being clipping. Shoot for somewhere between -15db and -8db for your maximum inputs for the time being. You can always bring this level up later. However, too quiet and you'll start to lose detail and introduce unwanted noise from the equipment.

When you are finalizing your mix in post processing, pay attention to the input levels of your eq and compression, as this can affect how these processes function. I don't have a hard and fast rule for this, so look into your plugins and experiment to see how

the gain changes their effect. At the end of your mixing, it is usual to do something called normalizing, which is bringing the recording up to a normal level. Typically, for classical music, I like to bring up the overall level to where the loudest points are hitting about -3db. In DAWs like Adobe Audition, this is very simple and visual, but consult your DAWs manual for specifics. *Do this after you have done all your processing.* You want headroom to do some of this processing, in case you need to mix things a louder, and don't want to end up clipping. Fortunately, digital audio is forgiving and the undo button is very powerful.

Videography

I am not a videographer. I understand the basics of video editing, and I know how to make things look decent. That being said, here are a couple tips on making your videos look good.

Like audio recording, it starts with the source. Make sure you have good lighting. Backlit sources are difficult to work with, front light is generally easier although can make things look flat. A subtle combination of both is usually good.

Look for visual obstructions. Mic stands, tall audience members, and other equipment can obstruct the view of the camera. Obviously audio quality generally takes priority, but keep in mind where you place a mic stand.

When livestreaming with multi-camera setups, go slow. You're better off staying on a shot for too long than you are disorienting your viewers with a million quick cuts. Same goes for panning – go slower than you think.

In this same vein, make sure you have a "parking lot" of sorts – a static shot that you can always return to if something goes wrong. It is easy to get lost in close-ups and slow pans, so it's nice to have something to rely on when you have no idea what is going on and allows you to reset your other cameras.

I have very little else to add here, consult a real videographer if you want real advice on this end.

Chapter 5: Troubleshooting

I hate to break it to you, but probably close to 50-80% of your time as an engineer will be troubleshooting. It's a job that nobody notices until things go wrong, so it's best to have a plan when everything comes falling down. The good news is, like I've said before in this book, if you know your equipment you'll have a much easier time troubleshooting. The point of this section is not to provide specific solutions, but rather to refresh your memory about what can go wrong.

Remember your signal flow

Knowing what is putting audio where will save from countless troubleshooting headaches. Whenever I have a problem, which in the audio world usually involves no audio where there should be, or lots of audio where there shouldn't, I mentally go through my signal flow and check each part to see that it seems to be functioning right.

The trick to this is isolation. Start at the beginning – is everything plugged in? Cables are your number one culprit here. If there is no sound, try switching out the cable. Broken cables are a dime a dozen. If this still does not change anything, go through each component of your signal flow and test it individually. If you change more than one aspect of your signal flow at once, you will not be able to identify the offending dissident. Think scientifically – try bypassing certain elements to see if that fixes it. Chances are if your signal flow works as intended without a certain piece of equipment, it is that element that has failed and needs to be repaired/replaced/thrown away.

Common Problems

Distorted audio

Distorted audio is most commonly caused by levels that are too high. Look for the places in your signal chain that have the opportunity to boost gain – notably your preamps and any plugins that you might have active. Look at your level meters in your DAW and elsewhere (OBS, interface) to see if anything is clipping. If so, dial it back to compensate. It is easy to want to make things loud and then suddenly end up distorting everything.

If this doesn't seem to be the case, it is time to isolate. Try turning off your plugins (if you are using any) one by one to see if that fixes the problem. Computers can have

issues sometimes with plugins creating distortions. Also try restarting your DAW and other involved programs on your computer. If all of this still doesn't fix the problem, check your cables. A bad cable can cause crackly or distorted sounding audio.

If all else fails, try a new microphone. It is possible (although unlikely) that the microphone capsule may be broken. This is unfortunately not something you can fix yourself very easily.

No audio

No audio is scary. Fortunately, 90% of the time it just means that you forgot to turn something on or plug something in right. The other 5% of the time it is a problem with the software or the computer not recognizing your interface. And then the final 5% of the time something is actually broken.

Double check that everything in your signal chain is correct – is everything plugged in where it should be? Did you turn your interface on? Are your monitors on? *Is the volume up...*

If that all seems fine, see if the computer is recognizing the interface or other peripherals. Unplug the interface, plug it back in, turn it off and on, just experiment until it connects right. If that connects right, you might have a broken cable. This is the most common item to be broken as they are somewhat fragile and are frequently used and abused. Experiment swapping out pieces of your signal chain until you get audio. Also try shortening and simplifying your signal chain – does a microphone plugged directly into the interface work? If so, that cable, microphone, and interface are all working fine.

Audio where it is not desired

This is almost always a problem of plugging something in the wrong place. When it isn't, it's usually the computer recognizing the interface wrong. Seeing a pattern here?

Audio in the wrong place can also happen in some older consoles and equipment due to a problem called "crosstalk." This is caused by interference between wires that are near each other. There is not much to do about this besides take it to a technician, however it is very uncommon in newer digital equipment.

Buzzes/hisses/hums

Buzzes come in two forms: internal and external. External buzzes are those that are in the natural environment. For example, the lights in the Glidden Recital Hall buzz, as does the air conditioner. These kinds of buzzes can be toned down somewhat with

EQ and some plugins, but really your only option is to just deal with it. Do double check though that someone isn't also running a fan or vacuum cleaner. If they are, kindly ask them if they would do it elsewhere. Condenser microphones are surprisingly sensitive and will pick up even the quietest sounds from dozens of feet away.

Internal buzzes are more complicated, and usually have to do with electrical issues. It may be that your electrical system is not properly grounded. This results in some electrical buffoonery causing hums and other strange noises. Power conditioners are supposed to help with this, but they don't always work. Check that you are using a grounded outlet, and try switching outlets.

Certain pieces of equipment may also introduce hums and buzzes, so if you have added anything unusual to your signal chain try removing it and seeing if it resolves the issues.

Audio cuts in and out or is crackly

Crackly audio is usually a result of a broken cable. As connections wear, the soldering between the cable ends and the wire itself tends to break down or fall off entirely. Obviously if the wire is not connected, the signal can't flow. Switch out the cable, and see if this solves the problem.

If the cable seems fine, try turning any knobs that might be in your signal chain, such as gain knobs. If the audio crackles as you turn them, it is likely a corroded contact in the knob. Corrosion stops electricity from flowing (or at least limits it), causing a crackly sound. This can sometimes be resolved with something called DeOxit, a spray that dissolves corrosion. It can be had online for relatively cheap and is often all that is needed to fix such a problem.

Occasionally crackly audio will be caused by sample rate problems in your DAW. Make sure that your sample rate in your DAW matches your sample rate in the interface, as well as in any other equipment you are using. If you are playing back audio or importing it into a DAW, again double check that the sample rates match. Sample rates are tricky and a total pain usually, so spend some time outside this book getting to know how they function.

Audio screams

If your audio is making a loud screaming noise, either a part of your signal flow is extremely broken or you have created a feedback loop. This happens when audio from your input is routed through to your output and right back in to your input, creating an infinite loop.

Most commonly this occurs when doing live audio for amplification (such as for a rock concert) as opposed to recording/livestreaming. However, it is possible for feedback to happen if you have routed something digitally back to an input, or if you have placed a monitor on stage for the artist to hear what is happening in the microphone.

Googling

Learning to be good at Googling/otherwise searching for information on the internet is one of the most important skills in technical troubleshooting. Time and time again, I hear from experienced technicians, engineers, and IT professionals that they would be nothing without Google. It feels like faking it, but I promise it isn't. It's a skill in itself and if you can get good at it, you can solve almost any problem.

This is partly why I have emphasized certain keywords. Knowing that the microphone that is acting up is a condenser microphone will help you formulate your search – "condenser microphone cuts out when moved." And it seems dumb, but knowing the brand names of your tools is just as important. The previous search could be refined to "AKG 414 cuts out when moved." Chances are, you aren't the only person who has had this problem with this particular model of thing. You will soon find yourself googling strange phrases like "OBS refuses to connect to ATEM Mini pro late 2018 model."

While official product websites and handbooks can be a good reference for technical specifications, your best bet when troubleshooting is usually forums. Audio engineer forums are bountiful on the internet, and don't discount places like Reddit, Quora, or even Yahoo answers. I know teachers in elementary school probably warned you that these are not reputable sources, and although this might hold true for doing research for a science project troubleshooting is another matter entirely. The world's basement dwellers are your greatest asset in those moments of utter panic.

Some questions to ask yourself

Audio

- Do you have your microphones plugged into the right inputs?
- Is phantom power on for the inputs you need?
- Is your gain set right?
- Is your interface on?
- Is everything plugged into power outlets?
- Did you turn it off and on again?

- Did you unplug it and plug it in again?
- Did you unplug it, turn it off, plug it in, and turn it on?
- Did you try any number of combinations and orders of this? Often times computers are picky about when you turn on a piece of equipment.
- Is the volume up on your monitors? And to that end, are your monitors on?

Streaming

- Is your stream key correct?
- Did you click "Go live" in the YouTube streaming console?
- Is OBS getting signal?
- Sometimes if you start a stream and then do not have the YouTube console active in your browser, the stream will not start. It will usually start if you return to it.
- If there is no audio, do you have the right audio input connected in OBS?

Don't panic. It's gonna be ok.

Exactly what the heading says – it really is going to be ok. Doing live sound can be stressful, as can recording. Artists can be grumpy and parents want to hear their child play their solo. This isn't to say to be completely blasé about the process, but to know that if you mess up, it's not the end of the world. After you do a couple hundred of these, you'll find your rhythm, find yourself messing up much less, and realize that the times that you did mess up really are ok. You're young, your inexperienced. Nobody can't and don't know what you don't know. As hard as it might be, remaining calm is your ticket to resolving your issue and keeping your PR nightmares to a minimum. Plus, if you get practice staying calm here, it tends to conveniently leak into other areas of your life...

CHAPTER 6: CONCLUSION/OTHER INFO

Conclusion:

I hope that you learned something. And I hope you're at least a little confused. This stuff is complicated, and it's impossible to convey all you need to know in a single 50 page document. Use this guide as a starting point, and do your own research. Dig into internet forums and get to know the terminology. If you get a chance, talk to some experienced engineers, and if you really get interested, see if you can shadow an engineer at their job. And above all, keep listening. It gets easier, and soon enough it will feel natural.

I want to congratulate you for making it this far in this book, and extend the invitation to contact me at any time if you have further questions. I can be reached via email at samdebatin@gmail.com or at book@samdebatin.com. Please feel free to reach out with questions, suggestions, or simply to ask about where I am now. I'm curious too.

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Thank you for taking the time to read this, and happy recording!

All the best,

Sam

Further reading

Hah - you thought this was the end, didn't you?? There are infinite topics that I didn't get a chance to cover in this book, so here are some rabbit holes for the par-

ticularly motivated. But first, more generalities...

Becoming familiar with computers and how they operate will serve you well. You will save a lot of time if you already know your way around the Mac operating system and can fix computer problems. Understanding things like networking will save you headaches when it comes to internet and some digital audio problems.

If you are interested, learning how to solder and fix equipment can be a huge boost. Not only do you save money, but you make yourself indispensable for whoever hires you if you are saving them money this way. With this, if you are scientifically inclined, learning some basic physics of sound and electricity beyond what I wrote here can be a help as well. Seek out your niche, and you will be rewarded.

Terms to look into

These are concepts and terms I didn't the chance to bring up, but are somewhat important nonetheless for understanding more advanced concepts. I will simply put them here for you to research on your own.

- Unbalanced vs balanced audio
- Line vs microphone level
- Dante audio
- Digital vs analog audio/video
- Sine waves and sound production
- Nyquist theory

Books, articles, and other media

- First, check out this thesis by former OU student Steven Van Dyne. It explores in more depth some of the techniques I have already covered. Available here: https://etd.ohiolink.edu/apexprod/rws_olink/r/1501/10?clear=10&p10_accession_num=ouhonors142
- *Microphone Manual: Design and Application*, David Miles Huber
- *The Technique of the Sound Studio*, Alec Nisbett
- *Modern Recording Techniques*, David Miles Huber
- Any number of audio recording forums and magazine websites - gearspace.com, soundonsound.com, tapeop.com

Recommended listening

Absolutely not definitive, but here are some things I like:

- Anything by the Berliner Philharmoniker, in particular the Brahms First Symphony
- Anything by the Cleveland Symphony Orchestra
- Anything by the President's Own Marine Band
- The Netherlands Bach Society on Youtube. Great recordings, great videography, and overall a great template for how a well produced classical recording with video should be.
- Glenn Gould recordings, 1980s, especially the Goldberg Variations and the Well-tempered Klavier. Gould was notoriously picky in this regard. This sound is actually quite unique in that it is quite dead, either due to close micing, a small room, or both. Gould also had quirky ideas about stereo image – worth researching yourself.
- Sviatoslav Richter recordings
- Kronos quartet – much more modern sound, but very well recorded.

And also pretty much any album nominated for the Grammy for best engineered classical albums. Duh.

Above all, seek out pieces that you like, and listen to different versions of them. Listening critically is less about learning a specific sound than it is about understanding what it is you like about certain sounds. If you like a piece that the Cleveland Symphony Orchestra does, see if the New York Philharmonic has a version. Compare the two, and see if you can figure out what makes them different and similar. Where are the microphones? How many microphones? What configuration are they using?

List of terms/Glossary

Below is a very short reference of important terms in both classical music and recording technology. Further lists can be found by googling. Again, this list is for familiarity more than it is for deep definitions. Don't feel bad if you just skim through the list for now and come back to it later – things will make more sense when you actually use them.

Technical Terms:

- Gain - the level of the input signal of a source
- Volume - the level of the output signal of a source
- Condenser microphone - a microphone that uses an electrically charged plate to produce signal. Most common type of microphone in classical music recording. Requires phantom power.

- Dynamic microphone - a microphone that uses an electromagnetic coil to produce signal. More commonly used for rock recording with louder sound sources
- A/D converter - Analog to Digital converter, converts analog sound to digital signals so your computer can comprehend it
- XLR cable - the three-pin cable that is used to connect microphones
- HDMI cable - the cable most commonly used to connect video equipment to computers
- TS cable - Short for tip-sleeve cable. Also known as 1/4" cable or instrument cable, the cable that is used to connect guitars to amplifiers, as well as some audio equipment
- EQ - short for equalization. Allows you to balance frequency levels in a sound source.
- Low cut - an effect that removes all frequencies below a certain frequency threshold
- DAW - Digital Audio Workstation. The software you use to record audio on your computer from external sources.
- Sample rate - How many samples per second are taken from your analog audio signal for digital conversion
- Bit depth - The "resolution" of a sound sample – essentially how much information is taken in each sample
- Phantom power - the power source provided by microphone pre-amps to power condenser microphones
- Pre-amp - the first level of amplification that microphones receive
- Audio interface - the device that interfaces with your computer to move signal from your microphones to your DAW for recording.

Classical Music Terms:

- List of tempos[4]:
    - Grave – slow and solemn (20–40 BPM)
    - Lento – slowly (40–45 BPM)
    - Largo – broadly (45–50 BPM)
    - Adagio – slow and stately (literally, "at ease") (55–65 BPM)
    - Adagietto – rather slow (65–69 BPM)
    - Andante – at a walking pace (73–77 BPM)

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[4]Taken from https://symphonynovascotia.ca/faqs/symphony-101/how-do-musicians-know-how-fast-to-play-a-piece-and-why-are-the-terms-in-italian/

- Moderato – moderately (86–97 BPM)
- Allegretto – moderately fast (98–109 BPM)
- Allegro – fast, quickly and bright (109–132 BPM)
- Vivace – lively and fast (132–140 BPM)
- Presto – extremely fast (168–177 BPM)
- Prestissimo – even faster than Presto (178 BPM and over)
- Concerto: a classical piece that features a solo instrument backed by an orchestra
- Chamber music: music usually played by smaller ensembles
- Movement: a unitary section of a piece, usually marked by a break in music
- Sonata: a piece written for solo instrument, often with accompaniment