Live Sound Reinforcement R&C Day Transcription

Slide 1:

Hello, this is Javon Jones, and today I will be doing my presentation of Live Sound Reinforcement. These are the approaches to the fundamentals and basics that are needed to know in order to create an effective system for a bunch of different applications, such as: lectures, concerts, live theatre, etc.; and also taking a look at a few field applications/case studies. At least that's the way my project was originally intended. It did have to change a little bit with the switch online, but I'll get into that in a little bit. So, let's get started.

Slide 2:

So my research did have a general structure to it. [It was] broken down into four parts. The first would be learning some of the fundamental concepts of sound and how it acts in spaces as well as some more technical concepts that would be more appropriate for learning how to work with the equipment which is the second part. Where it is taking looks at the individual portions of what makes up the system and making sure that there is an understanding of the parts before you add it to the whole: which is the setup. The setup is essentially taking all of those parts [and] interconnecting them in order to make sure that the sound energy produced from the speaker, an instrument, whatever it is that you are recording actually is propagated through the system and delivered to the intended audience with crispness and clarity. And from there, that being a more theoretical standpoint, there would be the move towards case studies of specific places. I was going to look at places on campus, and I still am, it's just that I've had to switch a bit of how I was going to do it. There was supposed to be a live application of a performance with UMW music that I would do the live sound for. So the switch to online made it so that can't happen, but luckily a lot of my project being theoretically based, and based on fundamental concepts was left relatively unaffected; so I was still able to continue.

Slide 3:

So, to start off the concepts. Some of the concepts that I looked at were what exactly is a sound system in the definition of a "system that delivers sound from the sound source to the audience in a clear and concise way." Then I looked into the science of sound; understanding the basic sound properties; understanding of electricity and how it mirrors sound and how it powers the system; the concept of loudness and how we use that in the language of live sound. And then acoustics which is beyond the science of the sound itself but how the sound

then interacts with ta space. These concepts are s some key ones to more complex and robust, but for the sake of the presentation I am not going to dive too deeply into those. I'll try to keep it as "on-surface-level" as I can.

Slide 4:

So, to start with the sound system, which is also referenced to the public address system because it is not just used for concerts and is used for other applications. It can be used for something as small as say a tour guide with a very rudimentary setup that is on their person to deliver clear intelligible sound to more people than would be capable with just their voice, or in other cases instruments, alone. There are other types of public address systems the first of which being a fixed-installations, which is cite specific. All of the tech and infrastructure is built into the space. It does not change. If it were to change it would be changing because of the additions of speakers and other equipment. But everything else is perfectly laid out and custom-tailored to the space for what it is made. Then there are also the portable public address systems. Going back to what I mentioned, the speaker (the tour guide more specifically) That would be the use for personal PA system. Then there is the band, which is typically what we think of more. It being on the stage, that one can be applied to concerts, live theatre performances, orchestras, and other sort of performance based activities that can happen on a large stage, Then you also have touring/stadium public address system and those have sort of evolved in their own way from the others because of how massive the scale is. If you think of the super bowl, that is a lot of people you have to conver, and they have developed their own system. And while my study doesn't primarily work with them and those differences, a lot of these concepts do apply [to the stadium]. And while you don't understand the complications and differences there, you can go into one and see how a stadium setup works. Because really a lot of public address systems are made up of a lot of the same fundamental parts. Which I will go over later.

Slide 5:

So to start with some of the sound science, you have to understand how sound works. Sound is basically the vibration of molecules. In particular, the sound we perceive and is necessary is the vibration of air molecules. How that ends up happening is that these molecules, once excited by factors like our voice and other instruments that provide vibration ,the excitement causes the sound molecules to bump into each other. And as we can see from the diagram, as represented by both a wave as well as lines that show how they get closer and farther away, when air molecules are excited, they get closer together, that being compression. Then they disperse a little bit until the bump into other molecules. That is what we describe as rarefaction, and it is the flow back and forth between that for how we describe sound. And

we have the compression and rarefaction, but we also the Zero Crossing Point, which is the ambient/atmospheric pressure of the space if the air molecules weren't excited to begin with. So sound is a little site-specific, but in general it is moving around a certain baseline pressure in order to create the vibrations that we perceive. It is important that we know that even though we put it in a two-dimensional field of understanding that this is something that is three dimensional and happens through multiple forms of matter. It gors through our bodies, walls, chairs, even though air conducts sound for us the best, it doesn't mean other things aren't acted upon. Which is something I touch on for acoustics. But let's move on.

Slide 6:

So another part of sound is how we end up measuring the intensity of the sound and how we end up measuring the pitches and frequencies that we hear. So, if you look at the top diagram, that is the crest (of the compression), of the sound wave. Looking at that tells us that the distance that compression, or the intensity of the compression in terms of the actual space, how far way that gets away from the baseline atmospheric pressure. Same goes for the rarefaction portion, the lower one. That one is the valley. The distance from the atmospheric pressure at those points tells us how strong and loud the signal will actually be. Additionally, we have to think about the pitch there as well, which has to do with how many cycles per second we get. A cycle is a combination of 1 compression section (top the image) and one rarefaction section (bottom of the image). So how many of those occur in a second determines our pitch. So that has to do with our perceptions as sound as either low or high. The range of human hearing goes from 20 to 20,000 Hertz, which is the term used to describe frequencies. We can see in the bottom picture how the frequencies relate to each other. As they move across units of time, there are more cycles per second [for higher frequencies] than are had in the lower frequencies, which plays a big role in acoustics later.

Slide 7:

Before jumping into acoustics and loudness, it is useful to have a basic understanding of electricity; at least for the terminology for loudness which we will end up getting into soon. But the basic electricity that ends up going through a sound system is split into four separate measurements of how we use electricity to go through the system and deliver our sound; because, as it turns out, electricity actually mirrors very closely to sound itself. Now that is because of the compression and rarefaction of Electrons as opposed to vibrations pressing and moving air molecules apart. And in that key understanding there are [four parts] that end up happening throughout the system and that we refer to.

The first being voltage, if we think about it in terms of dominoes, that's the first dominoes you end up pushing or pulling in certain situations but that's what sparks the electricity to

start going in motion through your system, through the cables, and to be produced in other places.

Then you end up having amperes, or amps. That, once the volts have effectively started the moving of the electrons, this is actually the rate at which the electrons move after being incited. There are different amp rating for cables as well as other pieces of equipment. So that is an important concept to know.

Then there are Ohms, which is thought of the impedance against the electricity and the current. Because even as you have these moments of compression and rarefaction, from the rarefaction standpoint, there is going to be some contest between the electrons trying to make its way through when it is being conducted and something that is pushing back.

Ohms [are] the naturally resistance against a current. Ohms are important because that resistance generates friction, and that friction is what we determine as wattage which is the heat energy that ends up powering the system, for volts, amps, and ohms are just measurements for how we move electricity through the system but the wattage is actually what allows our speakers to work, what allows the sound signals to actually be read and taken in; it's what allows microphones to pickup and transfer the acoustic energy into electrical energy and so forth. There are a lot of calculations and equations that go into these 4 fundamental concepts and measurements in order to transfer from one another and understand different parts of the system. Just having an understanding of those 4 will make it easier to start with your understanding of live sound

Slide 8:

Another core thing to understand is how we communicate loudness through sound systems and that is done through the decibel, which is derived from the Bel that was created by Alexander Graham Bell when he was working with electricity and power because decibels really are derived from a ratio between quantities. As I stated with the Bel, that had more to do with wattage and power, but here [the decibel] can be applied to many different things depending on what its relative to.

To start, though, decibels are actually logarithmic by nature which means that instead of it going up at a steady one-to-one ratio, as decibels increase the significance and the overall intensity of that decibel increase as well, ss you can see from the log scale on the right of the image here. Meaning that as decibels increase that the distance from 1 decibels to 2 decibels is different from the distance from say 20 decibels to 21 decibels that something that is important to understand especially when you are editing the levels in your system because [...] you may not need to go up as high as you think considering how they all compound but a thing to understand is that decibels if I were to just say 20 decibels, it doesn't really mean a lot by itself as it needs something to be relative to.

Slide 9:

And there is definitely an abundance of relative scales for what a decibel is. Some of them have to do with more about the electrical power, and even then with the electrical power dBm and dBW you can see that even the amount at which they are scaled to is different because dBm for example it's listed at 1 milliwatt being equal to 0 dBm, whereas dBW 1 watt is equal 0 dBW. And these have some different uses that are important for the different sections of the sound system. dBu (input/output) is really good for knowing how your mixer, which we will get into later, takes in signal as well as how well it puts out signal. The details are here for some of them but for the sake of simplicity and for what allowed the application for loudness is concerned using decibels. We end up using more of the DB SPL, the sound pressure level, which is derived from a Pascal, which is a measurement of pressure, where .00002 Pascals is the smallest amount that we can possibly conceive of a difference in the air pressure. Even that takes a good undamaged ear to actually happen, but that is the one that gets used the most or that I would say is the most significant because it's what we hear once it's gone through the speaker.

Slide 10:

It's important understand a few concepts for decibels per sound pressure level. It has gone through a scale ranging often from 0 to 130 dB SPL, although it certainly goes higher, where zero is the threshold of hearing, sort of the baseline like ambient noise when nothing is being when the space isn't being excited, all the way up until 130 dB SPL where that's actually what we consider to be the threshold of pain where the music starting more at 120 dB SPL becomes a little more uncomfortable and 130 dB SPL where it actually starts to hurt and damage our ears. So here we have a few relative scales for how what you can equate these areas of decibels to. And +1 decibel change is the smallest noticeable change that we can possibly have; again, it takes a trained ear to notice but definitely at plus 3 decibels you will notice a change in the sound. Every 10 decibels that are increased in SPL results in a double of our perception of the loudness of the sound. If you look at the scale on the right, it mirrors that in [how] each thing goes a step above and doubles itself. And then when you end up looking at it in terms of space. The Inverse Square Law happens with DB SPL, which results in every time you double your distance away from The Sound Source, you actually end up decreasing the amount of decibels per sound pressure level that you receive there for your perceived loudness as you can see on the image here, first it starts with 4 and then 8, 16 and it [progressively] gets larger and larger [with] distance doubling. So that's important to know spatially to understand loudness and how will that also builds into acoustics?

Slide 11

So is it is important to understand that Acoustics, how sound travels through the space, is an addition to the sound knowledge that we already have because it means that it will go and interact with the space which will result in different modifications that happen to it. Key thing to note, the speed of sound is 1130 feet per second, which means that 1.3 ft is traveled every millisecond through the medium of air. That changes with temperature as well as if you're at sea level because again atmospheric pressure affects all of the key fundamentals of sound but so do the spaces that sound is in so acoustics takes more of a look at that and there are few ways that sound gets affected.

One way is by refraction. That's where the medium that sounds going through from air to let's say a wall results in the speed of sound being different because although let's say like a wall its molecules will technically compress and will refract are there is still a difference in there because they are more compact and there are different materials that react differently, but essentially refraction is when sound transfers through another medium.

Then you have a diffraction which has to be which has to do with how sound gets around objects and it's space so it can go around and over objects because, like I said, sound does [compression] and refraction and it also happens in a three dimensional space. So when we think about it like a wave although it is three-dimensional technically the wave motion still happens, just invisible to us. That means that the amount of, let's say like the amplitude and its wavelength, affects how much it will actually be able to get through certain mediums. For instance high frequencies, hey only tend to have a wavelength measured in millimeters which results in it being more easily blocked and stopped whereas low frequencies are measured in yards and tend to be able to work around more obstacles as well as work its way through more obstacles. This is a lot of the reasons why sometimes you'll hear a lot of the lower sound before you hear the high sounds in certain sound systems because of how they are delivered and how old the system is set up if it's not ideal and might get a little muddy. Muddy in terms of you lose the higher frequencies that give you a lot of clarity.

Then you also have the absorption as I talked about with refraction sound can travel between different mediums, but when it goes through those mediums the energy gets trapped a lot more significantly depending on what the medium is, which has to do with something called the absorptive coefficient and the scale which you can see on this [chart] here. [The chart] actually shows which materials absorb more, which materials absorb less, and overall how that works. These are important to take into consideration when you're building your space because some sounds will just straight-up get sucked out of your system if you're not careful to counteract these absorptions.

Slide 12

So another important effect is actually reflection. And that is when sound comes in contact with another source. Of course, there is absorption. There is refraction that naturally will happen. But in addition to that, some sound will also bounce off the surface. How it does that is based on the law of reflection which has to do with the angle of the incoming wave; because, at least for a flat surface, the angle at which the wave comes in, the angle of incidence, it will reflect off [....] at the same angle in the opposite direction actually; the angle of refraction. You can see in the image below that there is a comes in at a 45 degree angle and leaves at a 45 degree angle. So naturally the more head-on that you get with a with a material or wall, the steeper the angle is going to be and you can even get it where if it goes directly towards the wall it can bounce directly back. So that's one thing happened with reflection […].

It also happens that like I was saying my original application had to do with a flat surface that you were hitting, but when you're dealing with convex and concave [surfaces] it can have different effects. Convex [surfaces] end up spreading the sound out, so as multiple sound waves hit that outwardly rounded surface, it bounces them in more direction and spreads to sound. If the sound is curved inward (Concave) the sound will tend to converge on one point from those reflections. Neither is inherently good or bad, but it depends on what you need more for your space. Although concave does have a tendency to be even more problematic as it can make some spots feel weaker. It can make some spots feel like it's gained a lot more sound […], whereas spots that aren't being focused on actually end up getting a less sound because of the reflection.

And one more thing about reflection we have to think about direct sound which is the sound directly from the sound source. It doesn't have a chance to be distorted by refraction or the absorptive coefficient or anything like that. [Direct sound] competes with reflections for clarity and the Precedence effect is important in regards to reflection because if reflections reach the ear within 30 milliseconds of the direct sound or any sound that reaches our ears within the same 30 millisecond timeline [are] averaged together so that we can perceive them easier. Anything beyond 30 seconds and the clarity starts to become a little muddied It makes it difficult to understand the sounds are coming into our ear.

Slide 13

[We can] take reflection further. We take it to the concept of reverberation, which is actually the reflective sound after the sound source has ceased in creating its sound that would be like once you strum a guitar and then the strings vibrate for a little bit; once it stops to the amount of time for the sound to keep bouncing and reflecting past that is known as the reverberation. It's not inherently good or bad, but too much of it can produce problems. There are a few main measures of reverberation; one of which being RT60 which is the reverberation time for sound to decrease by 60 DB SPL in order to be considered [imperceptible], at least in comparison to the noises of the room naturally.

Then you also have critical distance which is where the reverberation equals the direct sound. That has to do with physical position because as we know if you're farther away from a sound source, you're going to perceive the loudness of said sound differently and at a certain points, illustrated by this graph, it will show you that the reflective sound of the space will eventually overpower the direct sound. It's only really a problem once the reverberation becomes 12 decibels above the direct sound. So as long as you can keep that under control, it's a little bit easier and it won't be problematic there. And that is also something that can't be corrected by solely increasing the decibels or the loudness of the sound because as you increase the loudness of the sound you will also increase the intensity of the reflected sound so really you'll be increasing both of them which will result in the same graph, just at a higher sound level.

Slide 14

Through the understanding of some of those basic concepts we can move in the understanding some of the pieces go into the system that we use. So the system can actually be broken down into a few key parts, some of which being inputs where we actually take the acoustic energy and transfer it to electric signal; or in the case of line level signals, the instruments that are used keyboards and other such things just automatically transfer a sound of signal the mixer where it takes in all the signals that allows us to route them into different places one of which being processors where we can physically affect sound either on the mixing board or through outside processing equipment and move it back into the mixer to send towards the amplifiers and crossovers: amplifiers boosting the sound for the speakers and then crossovers making the decision of where to send the sound to the speaker line-up. And the speakers, of course, taking the electrical signal and transferring it over to the acoustic sound that we can then process with our ears.

Slide 15

So to begin with the first part of the chain of the system is the microphone. Microphones are important because they act as transducers that take acoustic energy and transfer to electrical energy. Really without microphones the system when I have a chance to even begin really [..] so it's all of its essential but it's the first step really of making the system work.

The microphones are also […] divided into two different types one of which being dynamic. Dynamic microphones don't need power, are good for high intensity sounds, and are extremely durable. Think about the situations where you end up having a live performer on the stage vocalist singing with the mic and they drop it. If they are to drop a dynamic microphone, sure it might sound a little unpleasant but the thing will still keep on ticking and things like drums or other sort of percussive instruments can also be used with Dynamic microphones.

Then you have the condenser microphones which require power specifically to them because of they internally pick up sounds. of a dynamic microphone picks up sounds just buy a metal diaphragm inside the casing of it that just takes the acoustic sound as is and sort of puts them through or starts sending them through in relationship to a magnet it interacts with. The condenser instead needs electricity in its diaphragm and it doesn't work so much with a magnet more so then how the diaphragm when acted upon [...] by acoustic energy […] it affects the electrical signal [and then] is transferred, hence it needing power.

[Condenser microphones] are better for more quiet applications and also ones that need a lot of high-frequency detail. Take, for instance, like a flute you might want to use a condenser microphone for a flute as opposed to a dynamic microphone. The only issue is that condenser microphones are not as durable as their Dynamic counterparts. So, if you're going to use them in a live sound situation, we need to make sure that it is somewhere where it won't get hit or knocked over and that it's secure, and that you know it's placement is where it's definitely going to be the entire time. Otherwise, you'll have a lot of problems.

And not listed here microphones also have their own pickup patterns in terms of how they take in sound. There are few with narrow pickup patterns, there are some that go around the entire microphone, some that are only […] do the front and the back but not the side. Those are important to your setup which comes up a little bit later.

Slide 16

So mixers are key in the fact that they take the electrical signal and they sum them up together and allow you to route them out to other places or even route them out as individuals to be processed through the processors mentioned earlier, put back into the board, and then sent out as your final signal towards the speakers. As you can see here, through this rudimentary diagram, some of the inputs that would come in from vocals, guitars, drums.

Some of the concepts, like fold-back monitors, [as] monitors are essentially speakers for the performers to hear themselves, but [the signals] can be sent out to other places too; headphones, effects racks, processors, a recording if you're doing it and then to the stereo outputs for your speakers.

There are two types of mixers. Analog which are the first types of mixers to come out because originally in terms of sound technology, we only worked with electrical signals. We didn't have any sort of digital means to interpret that sound data. So […] what you have on the board is what you have and if you wanted any to do any effects to the signals, you would need to send the signal out to an effects rack.

And with modern advancements we've moved more of a digital system where you still end up working with that electrical signal that can come in but now we have it in place where we can take that electrical signal and then code it to binary and […] computer language is essentially and have the same information and be able to send it and edit it.

Where digital mixers have a difference is that they'll often have a screen on them so that you can go between like different interfaces and different effects on the screen as well as having some of the hands on components as analog. There is not inherently a good or bad, it's just a matter of preference. Analog tends to have what some call a warmer color to the sound just because of how it retains the signal; whereas digital it's allows for more effects to occur and it allows for more layering and for you to make more use of space. So [mixers are] slowly moving more towards digital, and even some digital mixers are emulating analog styles just to give it that sort of old-style feel. But overall it's just a matter of what you end up having, analog or digital. They both get the job done if you know how to work with them. And digital has built-in system effects which eliminates need for an effects rack. Although it certainly helps.

Slide 17

Though there are the processors that I mentioned earlier that can [be] either analog as if you have the effects rack I mentioned well as a separate entity that acts on the electrical signal or,

built into the mixer, you have a digital edits that you can make to signal. There are three different types of edits each with their own subsections and categories of what they actually do.

There are spectral edits which have to do with working with the frequency, in other words the pitch that we end up here so we can Equalize, which has to do with editing how much of a signal we hear. We can boost to hear more of a certain signal (frequency) or we can decrease a certain signal if we find it to be problematic for our space or we just need to hear more of something.

Auto tuning and Pitch shifting which has to do with correcting the frequencies in real-time. This is often more of a musical application, Equalization too; most of these processes actually are more musically oriented, but they can find their way into not musical use if the need should arise.

Then you have you Dynamic, which has to do with the dynamic ranges that you work with mainly loudness. You have your compressors which if you have a large dynamic range, what compressors can do is that they can sort of squish the range into a smaller amount so that the that the difference between your lows and your highs (in terms of amplitude) aren't as a significant but also it can shift [...] the overall sound of that signal where you want. So if you squish the range of of it, and then you go "Hm, I squished the range, but I still wanted to be up in the higher (louder) part." You can move it so the [lowest volume amount] is a little bit higher and the [highest volume amount] is a little bit higher in addition to making sure that the range is still the same. Then you have your limiters which honestly are most of the time the saving races of sound systems because they prevent signals from getting way too high or they prevent sound signals from getting way too loud, which can blow out your speakers and render your whole system useless. One also not listed here are Noise Gates which actually end up having they cut out sounds below a certain decibel level so that if there's any noise, like hissing or humming that's happening, it doesn't get picked up and the microphones the signal will only be allowed to go through once it's at a certain threshold.

Then you have your temporal effects, which have to do with time. This has to do with any echo or reverb effects that you would add to your sound. I talked about Reverb being something that happens acoustically when it interacts with the space, but the Reverb effect can also work in how it just adds a […] slight delay to the sound that also sort of coats the sound and gives it more a depth and feeling as if it's coming from a larger space or a more reflective space. So those are three of the processors that we can work with.

Slide 18

So after we have gone through the processors, then we end up sending over to the last three parts of our system which are the amplifiers, crossovers, and speakers.

Amplifiers take the finalized signal that we have that we are happy with from the mixer and it sends it to reproduce it at a higher power wattage and ends up sending out towards the speakers at which point, before reaching the speakers, reaches the crossover which sends it to the appropriate speaker whether […] because there are different speaker types that handle different frequencies better, we get the choice of where and ends up going.

And there are two types of crossovers. We have active crossovers which occur before the amplifiers so you can see that down below that the signal will be split into two and those two being split will need their own amplifiers; or there is the passive crossover which happens after the signal has been Amplified. So it's really just about where it falls in the chain there.

When you think about the speakers, the speakers have four types and they are also transducers of electrical energy to acoustical energy. They are the output of the system. This is where everything the work that has gone through ends, and that everything should be coming through crisp and clear. And are 4 types of speakers to subwoofer which range which handles frequencies between 20 and 150 Hz and then you have the woofers, midranges, and tweeters which when you look at the diagram on the right, you can see that their ranges are from for the woofer 50 to 2000 Hz, Midrange 2000 to 5000 Hz, and then tweeters 5000 to 20000 Hz and the drivers and cones that are in each have different diameters to their drivers […] that actually deliver the sound to us

Slide 19

The system actually wouldn't mean very much if we didn't have the cables to connect all of the pieces together. Now the cables all have […] different connectors at the ends of the cables 3-prong XLR, 1/4 inch Jacks, and other types of connectors are used to take the signal and transfer it between different pieces of tech among sound system, but even amongst all those differences there are two fundamental […] divisions.

[They are] balanced and unbalanced. Unbalanced signals are best for short distances and saying best means really the only good for short distances because the signal in an unbalanced cable doesn't travel very far. And in addition to that noise gets picked up very easily by these cables. This was sort of like the original before they developed the more advanced, balanced cable technology which retains the signal strength longer and naturally counters the noise from the environment because of how it processes the signals and creates two copies that actually run side-by-side and add to each other at the very end of the cable run to create a cohesive signal. Those are very important to the sound system and should not be neglected or overlooked at all.

Slide 20

Having touched on the pieces and now comes to putting them all together to set up an effective public address system and the key to it really is that this is all the work that goes in theoretically and happens even before you get placed on the stage. So you have to think of your event, know the space that you're working, and make some preemptive decisions of how the work around some problems; Know your Basics so that you can adapt and get creative if you need to […] should something happen at the venue. That's a little different because even as you have all the knowledge, all the sort of things to keep in mind rules of thumb, etc. That doesn't mean that you're going to be guaranteed the good sound that your predictions lay out for you. It comes down to what you hear and you have to be ready to adapt in the space.

Slide 21

If you're Outdoors, you'll have some factors to consider, although most of them will be things that are very difficult to actually compensate for; Wind being one of them. If you think about wind in terms of what it actually is, wind is a shift and movement of air molecules. So if you go on that line of thinking with knowing the basics of "Okay, the sounds are actually vibrations of the air molecules in the movie together" It can actually happen that really strong wind comes by and it can shift your sound from going in one direction to another or shifted slightly to the side.

Now, because of the size of high-frequency sounds being in millimeters for the wavelength. Those are the ones that get affected the most by high winds because of the fact that both frequencies are actually measured in yards, those ones hardly get affected by large winds. […] And even the effects that you have on those high frequencies tend to be minimal unless it is like hurricane level winds or probably even a little lower than that. But by the time you get to that point, you probably shouldn't be at a concert or at whatever the outdoor event is.

You have the temperature and the humidity which end up bending the sound for temperature. Let's say you have it's the morning the ground still cool, but now that he's above the starting that are the air above starting to heat up that can affect where the sound gets pulled to.

Humidity sound can also bend and it affects the speed at which the sound travels as well.

Those are things are difficult to account for and sometimes you kind of just have to deal with it and be able to live mix in the moment in order to fix it. And then one that's more within your control is noticing if you don't have any reflective surfaces around your space that means you can either if you have it available to you, you can add like little panels architecture

that have things to bounce off of or you just make sure that your speakers are hitting the audience directly.

Slide 22

If you are indoors some things that you have to worry about, of course, are the effects of reflection and reverberation that occur within the space. Again, even if you have all of the theoretical, the absorptive coefficients and sort of the Baseline knowledge of how it's going to work in the space, you can't on 100% percent plan for how it's going to sound and especially if you have some difficult architecture that you're dealing with such as some of the entryways and stages, bends, corners and multiple areas that you have to hit, shown in this in this diagram here, then that'll be another thing to have to sort of work around and get creative with in the moment. Should you find that isn't quite going how you expected.

Slide 23

Even if some things can't be planned. There are plenty of setup things that you can actually work on and do before you even get into the space. Knowing what you have and making a list of it and making a simple ground plan of where you're going to set things up even something as rudimentary as this [ground plan] is something that is very, very helpful to visualizing what you're going to do so that you don't walk into the space when you're setting up and devote more time figuring out where you going to put stuff than actually where you're going to set it up.

Additionally, you can also make a channel list for where all of these things are going to plug in so you can keep track of the signal flow of the audio signal to make sure that you know, what's going in to what and if something goes wrong and you can troubleshoot.

Slide 24

When you do that set up you need to make a lot of essential choices for the space that you have you and make sure you don't block any essential areas and try to keep clutter to a minimum. This means not only where you place the equipment but how you also place and tape down the cables. You have to make sure that they're not a safety hazard; you need to make sure that you're following guidelines of not blocking fire exits and other such things.

You have to make sure that your speakers are place where they need to be in a need to be secure on-the-ground so that they do not fall over and hurt anyone.

When you're making those decisions to tape cables and other things you need to try and make sure that you know it is going to be permanent. There's nothing that's more terrible than trying to make a plan, taping stuff down and realizing you're going to change your plan and wasting tape.

Mark your cables clearly so that you can track them along the chain, and also leave extra cable in case you do need to move something.

Setting up smartly and safely is just a good practice in general to make sure that you're flexible as well as everybody else is not in any danger.

Slide 25

During your setup, you'll have to make choices about your microphones.

When you end up making those choices it's about how many you have and what types of microphones you have. Take, for instance, the Dynamic vs. Condenser. If you have only one condenser microphone the rest are dynamics, think about what you need to get a cleaner higher frequency sound out of sight, so you know, "Okay, [..] I should devote the condenser mic here.'

Then you also need to think about the unintended sounds going into mics when you're setting levels of the sound that has to do with how much signal you will get from the singer or the instrumentalist, r speaker, at a certain distance and in general there is a 3-to-1 rule for that because when you set that you no matter what you do, no matter what level you set, there is always the risk for unintended sounds to reach into the mic from either other performers or other outside sounds. And in general you can follow 3-to-1 rule for the distance at which you place the sound source from the microphone and the general like "Okay, […] they're in general going to be this distance away."

All other sound sources need to be at least tripled that distance from the microphone in order to give yourself space. And to avoid bleed (unwanted noise pickup) in your system.

Then there is the inverse Square law for microphones to take into consideration. It's a little bit different, […but]the fundamental principles the same for how it works for sound traveling from a speaker to a person or any sound source to any person listening but the distance between as the distance between the mic and the sound doubles, there's a drop in 6 dB SPL. The inverse is also true as well, and that as you get closer the sound and the signal that you pick up will increase.

It's also important to note that the closer you get to the mic, the more lower frequencies get picked up as the proximity effect and it's something to take into consideration [..] to make sure that sound doesn't become too bass-heavy and some things like drum kits depending on the venue might not even need to be mic'd. So if you need to really try and make sure that you're not using too many microphones or you don't have enough microphone the drum kit and other percussive element can be left out if they are allowed enough for the venue.

Slide 26

So there is the speaker set up to think about where you end up having both the monitors and the public-address speakers to consider where you place them. Again, monitors are for performers to be able to hear themselves during the performance and the PA speakers are for the audience to hear what the performers are doing. You want to put an emphasis to put the PA speakers in front of the performers and towards the audience. This is to make sure that you don't end up sending the sound from the speakers to the performers that can confuse them and overwhelm them they won't be able to hear themselves as well. That's why you have the monitors so that they can hear themselves, but […] with your microphones you have set up results [can produce] feedback, which results in a continuous loop of "Signal that goes in through the chain, comes out the speaker at a louder intensity, and then the louder intensity goes back into the microphone which continuously boosts it and makes it louder up until the point where if you don't have a limiter you damage your equipment and you blow out your speakers.

Precedence also needs to be given to the high-frequency speakers so that they can be seen by the audience because chances are if the audience can see the speaker that means that it has a clear line to them, and since higher frequencies are considered directional whereas lower frequencies are considered omnidirectional, the higher frequencies need to be able to get directly to the audience for clarity.

And even as we're dealing with the frequency range from base all the way to travel with a high, they should all ideally arrive within the same amount of time. Think of the 30 millisecond average that happened with the reflections I was talking about earlier with the fundamentals. As we hear things within that range we will some them together and think of them as being together anything if they arrived outside of that range, then the rhythm can get thrown off, clarity can get thrown off and things just start to not match up. If there's an issue with the space being […] too long or too wide for effective spreads, you may need to set multiple speakers on a such as Fills or Delay speakers in the space in order the make sure that you can travel it further and that you can hit some of the dead spots. The only thing is is that

for depending on the placement you might have to put an artificial delay on the speakers in order to make sure that they do not conflict with the 30 millisecond average rule.

Slide 27

Mixer setup is really where when you take in the signal. It is sort of the midpoints [and] the point where you have the most control over the sound because once you have the microphone setup, it's up to the performers and then the speakers can only do what you allow to go through. So you need to know what your channels are. Like if performer one is getting a little bit too intense, you can turn them down on your on the board.

You can fine-tune your sound for instance the EQ (Equalization) so you can EQ in real time in addition to EQing before the sound before you actually set it up and before they're performing yet to know what you're mixing for. You'll mix differently for an orchestra then you'll mix for a rock band because you don't want to take away the sound that they are known for.

Then you also want to give yourself some Headroom, which is essentially giving yourself some dynamic space. You can control this by how much of the audio signal you allowed to pass through and at what intensity. You just need to be aware of the fact that depending on the signal strengths, they'll add together. So you don't want to dip too much into like that extra space and cushion that you've given yourself to not reach the maximum level.

And you want program as much as you can to do the work for you, whether it be the noise Gates that keep the smaller sounds out, the compressors that limit the dynamic range, and the limiters to make sure you don't go above a certain threshold, but you still want to be active and make sure you are paying attention to what is occurring.

Slide 28

And there is nothing more to be said then the fact that all the fundamentals all the theories all the live stuff. You can plan as much as you want, but in the end the only way that it's really going to improve and that you're going to be able to rely on this knowledge is through experience because you know, like I was saying before their moments when you have to get creative improvised and frankly you can't be ready for everything that's going to happen. And if you could then you wouldn't need all this detailed knowledge on hand. You could just make sure you read your book set everything correctly and then just kind of let what happens happen. And the rules of thumb there lead to…

Slide 29

…The case studies and the actual applications in the space.

Now what this is sort of where my project in this presentation takes its end because I was going to apply all of this knowledge towards […] a setup of a public address system for a live performance with UMW Music at the amphitheater. That would be an outdoor application and all the same processes like knowing the basics, knowing the system, and […] how to set up for the space would apply then but the switched online has resulted in that not being a factor or not being possible.

What is still possible and what is still being worked on for me at this point is doing research of Klein (Theatre), the HCC Digital Auditorium, and Dodd Auditorium on campus. Naturally. I can't do an in-space research at this point, but I can still focus on pictures and diagrams as well as talking to some of the people in charge of that space to get an idea of what's occurring. That's something that will happen within these last few weeks. It's just as a result of the change that the at the university took I had to end up trying to think of some shifts to my approach and that's what I have determined so far.

Slide 30

And there you have it that is my presentation for live sound reinforcement, knowing the basics, knowing the parts, knowing how to set it up, and references to the work I will be completing in terms of examining and doing case studies of spaces, albeit from home. And without a physical application to apply it to, the information is still something as very important and very key to knowing for my projected forward career with music technology and sound in general. These were the two resources. I use basically live sound reinforcement the sound reinforcement handbook. The sound reinforcement handbook actually was it's more based than analog and diving into definitions of concepts, whereas the basic live sound reinforcement where I got a lot of my live setup and live application ideas from. […] All the pictures and diagrams with in this presentation was used from the basic live sound reinforcement book. And yeah, that is everything. Thank you so much for tuning in and for listening to what I have been working so hard on this semester. Take care.