Comparisons between Select Studio Microphone Techniques
in
Middle Tennessee State University's Studio A

by

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A thesis presented to the Honors College of Middle Tennessee State University
in partial fulfillment of the requirements for graduation from the University
Honors College

Fall 2015

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Abstract

There are many different factors that affect recording music, including the musician, the room, microphone selection, and microphone placement (Owsinski, 2014, p. 77). To demonstrate how microphone placement can change the tonal quality and apparent loudness of a sound source, I recorded instruments with multiple microphones in different positions during the same performance. I edited the recordings by unmuting one microphone at a time so the listener would be able to hear clearly the difference in sound between microphones. The end result of my project was a five-minute video that outlined my procedures and presented my recordings along with a brief summary of my findings. The video was designed to provide greater accessibility to this work, especially to students, by making it available on websites such as YouTube. I concluded that even small microphone position changes greatly affect recorded sound, as well as made instrument-specific generalizations about positioning relative to tonality.

Table of Contents

List of Figures

Introduction

In a recording studio, it is the engineer's job to capture a musician's or a band's performance. Initially, engineers are faced with several choices. A decision must be made as to where each musician should play; should they play together or individually; and how far apart do their instruments need to be? The engineer must also decide which microphones to use on each instrument and how to position them relative to the sound source. It might be tempting to think that microphone position has only slight effects on the recorded sound; many instruments sound about the same to our ears at multiple distances, though quieter as distance increases. To examine why the microphone captures sound that varies from what our ears hear, it is important to understand not only how microphone level changes over distance, but also how frequency response is altered by microphone position.

Figure 1 shows a diagram of the recording process. First, a sound source projects energy through the air, then a microphone converts the sound in the air into electrical signals. Next, several processing elements, such as a preamplifier, equalizer, and compressor, modify the electrical signal. Last in the signal chain is the recording medium, such as analog tape or a digital hard drive.

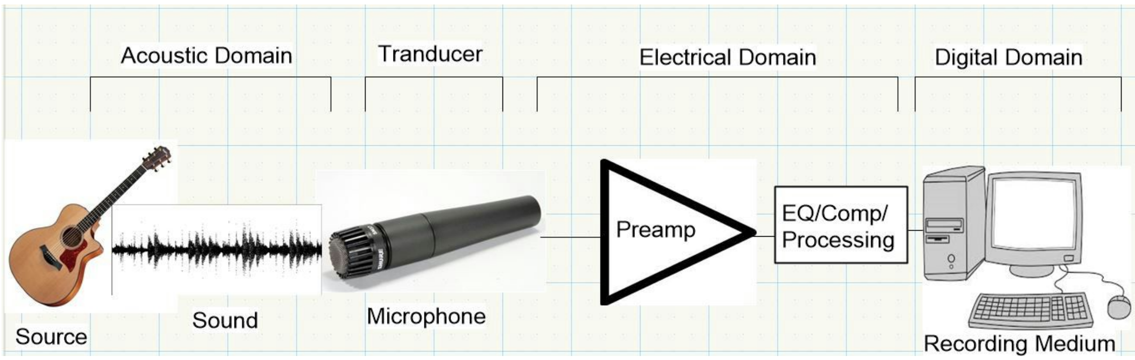

Figure 1 Recording Signal Chain

To capture a performance in a way that the final product will be considered “technically satisfactory,” a common contractual term indicating the recording meets industry standards for professional audio (Halloran, 2008, p. 342-343), all the elements in the chain must be performing in a manner that contributes towards this goal. For example, a good microphone, positioned correctly, sending signal to a computer that faithfully captures the sound from the microphone, will still result in a poor recording if the musician’s playing is subpar. Generally, a technically satisfactory recording comes from a combination of a “good source + good placement + good mic” (Huber, 1998, p. 1). This is contrary to a “fix it in the mix” style of engineering that would try to correct deficiencies in recordings at a later date during editing rather than strive for the best possible raw takes. The reason a “fix it later” attitude often fails is due to the one-way nature of the recording signal chain; every time a signal is modified before it reaches the recording medium, there is no going back.

Since engineers do not affect the proficiency of a musician's playing, the microphone is the first point in the signal chain where they exercise any control. Editing, mixing, or processing will only modify what the microphone captures. Since the microphone is the furthest upstream in the recording chain, anything that happens afterwards will be directly related to the microphone's performance (Touzeau, 2009, p. 35). This means that the frequency content of the final product is heavily based on the initial signal the microphone captures. Changes in frequency content can be expressed not only in the quantitative amount of each frequency present in the audible spectrum, but also in terms of phase response, and the amount of reverberation present in the recording.

Spectral analysis, a means of measuring frequency content, is accomplished using a Fourier transform algorithm that plots bands of frequencies on the x-axis against their amplitude on the y-axis (Cimbala, 2010). This is referred to as a frequency response chart (Figure 2).

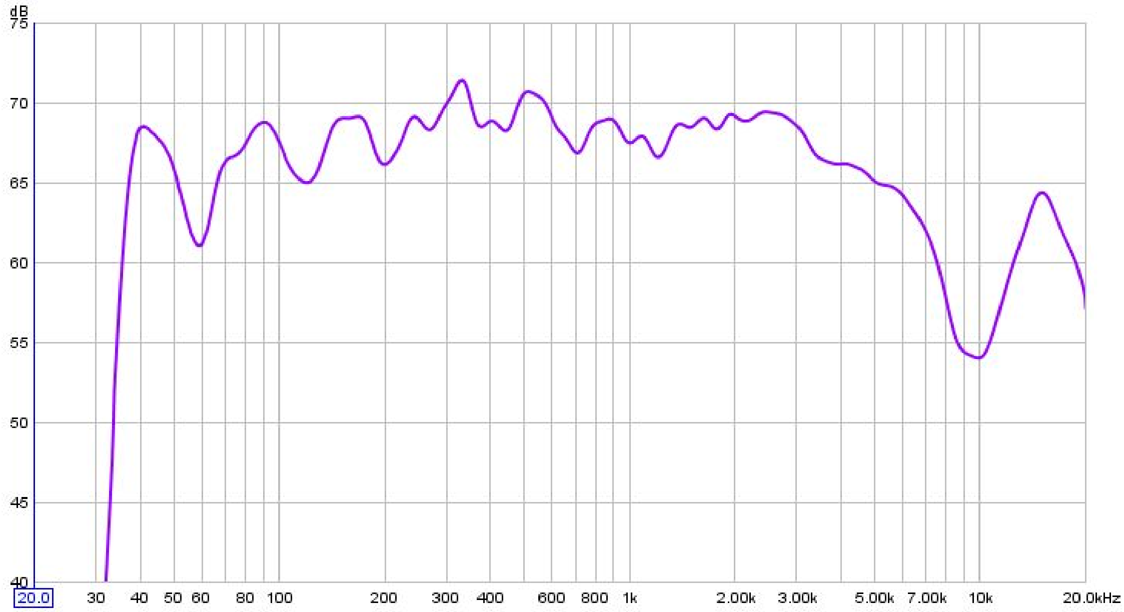

Figure 2 Generic Frequency Response Chart

When the amplitude of a narrow bandwidth of frequencies is manipulated, that change is perceived as a change in tonality, or tonal color. A boost in high frequencies would often be described as making the sound brighter. When the amplitude of all or most of the frequencies in the audible spectrum is raised or reduced, those changes are perceived as volume, or magnitude, differences. Microphone placement exhibits both changes, as the tonality and the perceived loudness of a sound source change with the microphone position. The amount of reverberance the microphone picks up will also change with placement and lead to changes in frequency content.

Microphone placement leads to quantifiable level differences due to the way sound propagates through the air. When an instrument generates noise, it projects sound energy through the air in all directions. The sound waves travel outward in a sphere and further apart. As distance is increased by a factor of two, surface area increases by a factor of four: the amount of sound energy is constant, but more disperse. This is called the inverse square law, as diagrammed in Figure 3.

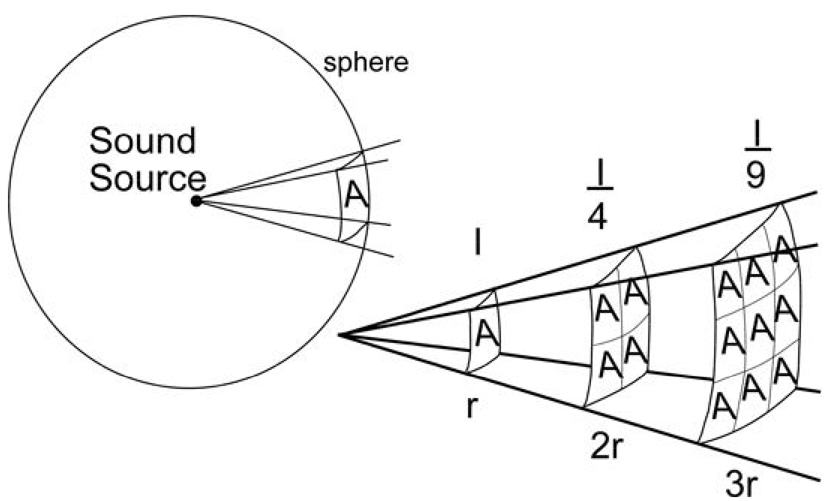

Figure 3 Inverse Square Law

Reverberation is the addition of reflections from the original sound source off objects in the room such as walls, tables, and other surfaces (American, n.d.). The quantitative aspects of reflections are the delay between the arrival of the original sound and the reflected sound, the frequency content of the reflected sound itself, and how long it takes for the reflected sound to dissipate, or decay. The quickness, or attack, of a sound, along with the decay of its initial peak, sustain, and release are collectively referred to as a sound envelope (Figure 4).

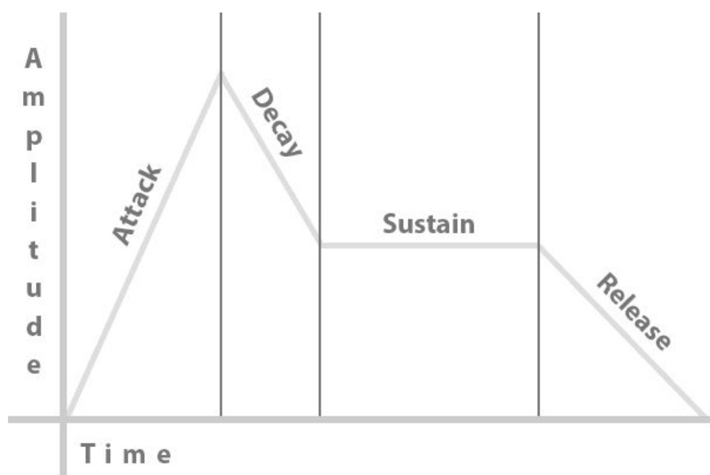

Figure 4 Sound Envelope

An echo is an example of reflected sound with a long delay between the original sound and the reflection. How long the echo “hangs” in the air represents its delay time. Where a microphone is positioned relative to the sound source and the reflective surfaces determines how much of the reflected sounds are present in the final recording. The further from the source the microphone is placed, the quieter the direct sound becomes; the closer it is to reflective surfaces, the louder the reverberant sound. Only anechoic chambers with no reflective surfaces exhibit zero reverberance.

Microphone position is also relevant not only to recording studios, but also to any profession that uses microphones. This includes any amplified sources, such as rock concerts, sound for picture, orchestral sound, churches, DJs, or corporate conferences. Any time a microphone is used to capture sound, its placement relative to the source affects the resultant audio. Positioning techniques vary throughout the different sections of the music industry to achieve different results, but the way a Shure SM57 microphone responds to a guitar cabinet is consistent across a broad range of applications, whether in the studio or concert hall. It is the engineer’s responsibility to choose the right placement for the intended result.

When deciding where to position a microphone on a sound source in a studio setting, the recording engineer has many options. Typically, microphones are placed close to the sound source in order to capture a very controlled sound with little room noise. While a distance of three to six inches may be common, greater distances may be used if the engineer likes the way the original sound interacts with the room and wants to capture that for use in the final product. Such decisions, however, are irreversible. For this reason, it is not uncommon to have multiple microphones recording the same performance simultaneously to provide flexibility during mixing. Specific microphone placement techniques vary between engineers, largely as a result of differing backgrounds and experience.

In the recording industry, professional engineers come from diverse educational backgrounds and differing levels of experience. Engineers use techniques that consistently capture what they desire in an audio recording. Since each engineer comes from a different background, what each considers "good" sound will be different. A hypothetical example of this is someone who primarily listened to music on a car stereo and, as a result, thinks that all music should sound like it does inside the car. When this person enters a recording studio and listens to the same music on high-quality speakers with a much greater frequency range in a quieter environment, that person may think that the changes make the music sound bad, since it is not to what he or she is accustomed. This demonstrates the importance of establishing sound references that match industry trends for professional audio. An individual's sound references are the way a person expects to hear certain musical sources in order for them to be considered pleasant. The goal of my thesis was to provide examples of how select microphone placements sound on several instruments in order to help engineers make faster, more informed decisions on how best to match their individual sound ideal.

When I designed this project, I wanted to accomplish three major goals. I wanted to demonstrate clearly that microphone placement changes how sound sources are captured in recordings; I wanted to draw conclusions as to how microphone position relative to the sound source alters the results; and I wanted my results to be accessible for use as an educational tool. To accomplish this, I created a compilation of individual recordings of multiple microphones capturing the same musical performance at the same time. Although textbooks and in-class discussion can introduce concepts of studio microphone techniques, these provide limited information regarding the specifics of a sound without actual audio. I chose microphones in matched sets and three sound sources that reflect techniques applicable to a wide variety of instruments.

It was important that I use the same microphone for each performance so that the only variable being manipulated was the position of the microphone relative to the sound source. With modern manufacturing technology, frequency response remains relatively consistent across new, mass-produced microphones of the same make and model (Sweetwater, 1999). However, age and condition can affect the response of any individual microphone. Since I did not have access to brand new microphones to control this variable, it is worth noting that some small changes not due to placement may be present in the recordings.

Another aspect of microphones is gain-staging. Identical microphones would not demonstrate recorded level differences if the microphone preamplifier gains and overall send into the recording software were not identical for each microphone. To accomplish this, I generated a sine wave tone in Pro Tools and electrically split it into the signal chains that the microphones would feed into. I then compared the levels across the separate microphone tracks once they fed back into Pro Tools in order to make sure they were the same (Figure 5). If my reference tone had registered higher in one track than the others, that would have corresponded to a microphone that had too much gain. By adjusting the individual microphone preamplifiers with a reference tone, I ensured that the only difference between their level as they were recorded was their relative position to the instrument.

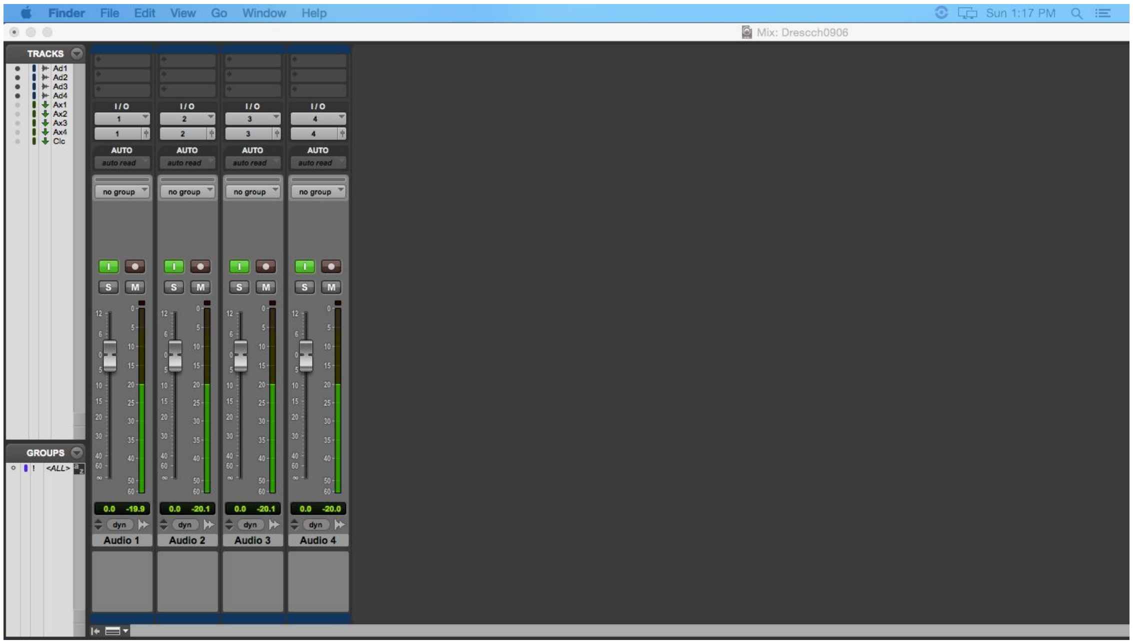

Figure 5 Gain-Staging of Microphone Signal Paths

I performed the recordings inside the control room of Middle Tennessee State University's Studio A in order to eliminate interference from outside sources; the studio is a controlled environment free of wind, most extraneous noises, and is temperature and humidity controlled.

The sound sources I chose to record were an acoustic guitar, an electric guitar cabinet, and a kick drum. These sources represent three broad classes of instruments common in modern recording. The acoustic guitar is an unamplified, stringed instrument with a sound hole that projects the sound of the overall instrument. This is similar in construction to other instruments such as mandolin, dulcimer, violin, cello, and upright bass. The electric guitar cabinet is a reasonable stand-in for a wide variety of amplified instruments that use speakers as their primary sound source. Microphone techniques applied to electric guitar cabinets may also be applied to bass cabinets, keyboard amplifiers, and other loudspeakers. The final class of instrument is the kick drum. Though less applicable to other instruments, the kick drum is so fundamental to popular music in the Western world that it could not be overlooked. Dave Moulton, producer of *Golden Ears* ear training materials, referred to the kick drum as, “the driving force, the rock and roll motor that makes our body move” (Moulton, 1993).

To reach a larger audience, I created a video containing the samples I recorded in Studio A. Since over half of all men and women who use the internet are YouTube users (Google, 2011), I chose YouTube as the host site for the companion content to my thesis. The result was a five-minute video that outlined my recording process and presented the samples with a brief summary of my findings. I also provided a link to the full written portion of my thesis with the video for those interested in more in-depth analysis.

In the video, I featured each microphone individually on each sound source in order to compare the differences between positions. I programmed these transitions, with the exception of the kick drum, to occur slightly after the beat of the music being played in order to make the change more immediate, with the goal of highlighting the differences between the sound at each position.

The abrupt transitions allow the listener to hear tonal and level differences more sharply contrasted, making it easier to determine which frequency bands have changed.

After the recording of microphone samples was complete, I analyzed each using a transfer function in the SMAART computer program. In a SMAART transfer function, two inputs are compared to each other in terms of frequency content and phase relationships. Frequency content is shown as the amplitude difference in decibels on the y-axis and frequency in Hertz on the x-axis. Phase is a measure of time alignment between two related signals. The graph for phase displays phase difference in degrees on the y-axis and frequency in Hertz on the x-axis. In the SMAART program, the phase graph appears in the middle of the screen and the frequency content on the bottom. There is a third graph that appears in SMAART as a narrow strip along the top of the page that deals with the impulse response of a signal, but is not pertinent to this project (Figure 9.1)

The SMAART transfer functions, along with my personal qualitative observations, allowed me to draw conclusions as to how each specific microphone position had altered the way the sound was captured. From there, I was able to generalize how movement on each axis with regard to the sound source affected the overall recorded sound. This information is relevant to music industry professionals and amateurs alike, who desire a technically satisfactory performance from their microphone positioning techniques.

Microphones

Microphones are transducers that convert sound energy in air into electrical pressure, or voltage. There are three main types of microphones: ribbon, condenser, and dynamic. All three operate on the principle that a moving piece called a diaphragm creates electrical pressure differentials inside the microphone which becomes its output. Since dynamic microphones were readily available in matched sets, I chose two different types of dynamic microphones to record samples: the Sennheiser MD-421 and Shure SM-57. Both are specified as cardioid microphones by their respective manufacturers (Sennheiser, n.d.; Shure Inc., 2015.). The polar pattern of a microphone determines what directions the microphone will capture frequencies well and what directions it is designed to reject sound. The most common types of polar patterns are cardioid, super-cardioid, hyper-cardioid, bidirectional or figure-eight, and omnidirectional (Figure 6). The model of microphone will also exhibit differing responses to certain frequencies as part of its design (Rayburn, 2012, pg. 40). The frequency response charts provided by the manufacturers are shown in Figure 7.1-2 (Sennheiser, n.d.; Shure Inc., 2015.).

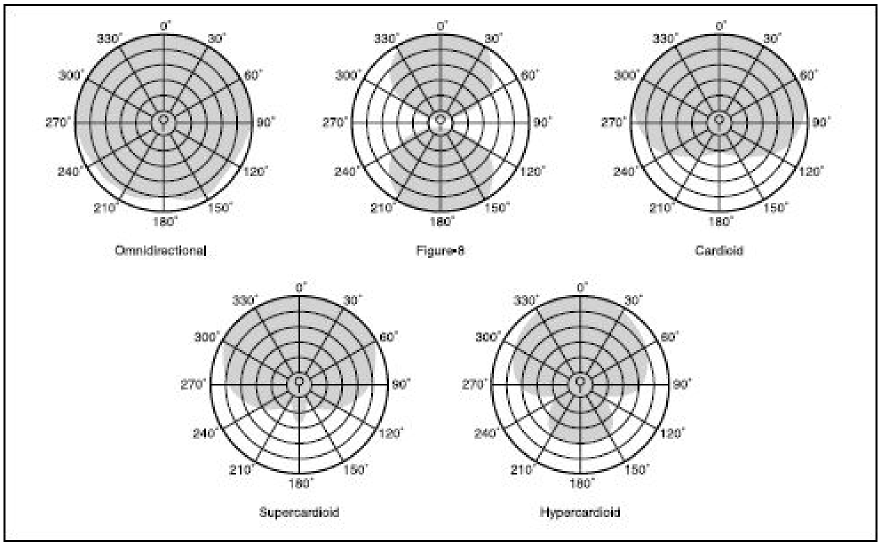

Figure 6 Microphone Polar Patterns

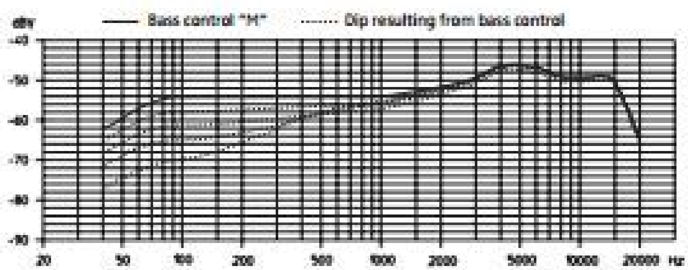

Figure 7.1 Sennheiser MD-421 Frequency Response

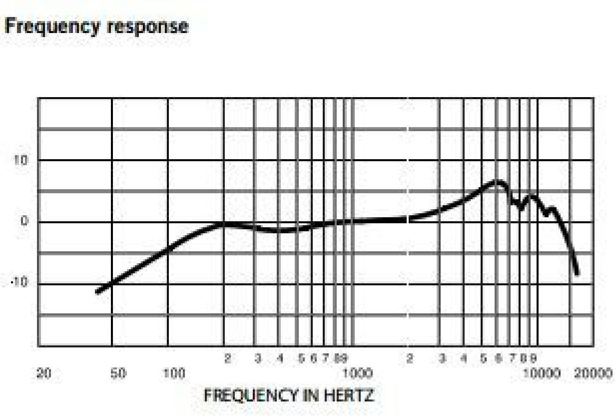

Figure 7.2 Shure SM-57 Frequency Response

Spatial positioning of musicians during recording sessions has been an important topic since before electrical amplification. Acoustic recordings required the musicians to be in the same room and to play for one listening position: the horn acted as transducer, converting acoustic pressure in air into mechanical indentations on a wax cylinder. If a loud element was positioned too close to the horn, the other musicians might not be heard at all. Recording was also an expensive process and did not benefit from individual tracks or overdubbing takes until multitrack recording in the late 1950s. With the arrival of electrical microphones, not only did the musicians have to be positioned properly as a group, but microphones on individual performers had to be considered as well.

Now, with multitrack digital recorders at our disposal, the correct position of the transducer relative to the instrument is easier to achieve than ever. Engineers can listen in headphones to a microphone's output even as it is being positioned. Using this method for positioning microphones, engineers can audition the final outcome before the recording process begins. When one combines this with multiple microphones in different positions all recording at the same time to what is essentially limitless hard drive space, the expected quality of a recording rises significantly.

Recording Methodology

To record samples of relevant microphone positions, I first had to research which microphone placements other industry professionals had found worthwhile. I used the comments and suggestions made by notable recording engineers available online as well as advice on the proper use of microphones from microphone manufacturers to establish prevalent microphone techniques that would be useful for comparison. The following table lists the microphone positions I recorded as they appear in the video by instrument and the primary source that influenced that inclusion (Figure 8).

Acoustic Guitar	1: 3”; on-axis; neck	Owsinski (2014, p. 154)
	2: 3”; on-axis; neck joint	Owsinski (2014, p. 154)
	3: 3”; on-axis; soundhole	White (2001)
	4: 3”; on-axis; bridge	Huber (1998, p. 59)
Electric Guitar Cabinet	1: 2”; on-axis; center cone	Rudolph (2014)
	2: 2”; on-axis; off-center	Clink (Senior, 2007)
	3: 2”; 45° off-axis; off-center	Ainlay (Senior, 2007)
	4: 3’; on-axis; center cone	Shure Inc. (n.d.)
Kick Drum	1: 5”; on-axis; beater strike	Huber (1998, p. 48)
	2: 5”; on-axis; outside edge	Huber (1998, p. 48)
	3: on-axis; inside sound port	Owsinski (2014, p.113)

Figure 8.1 Microphone Positions and Sources

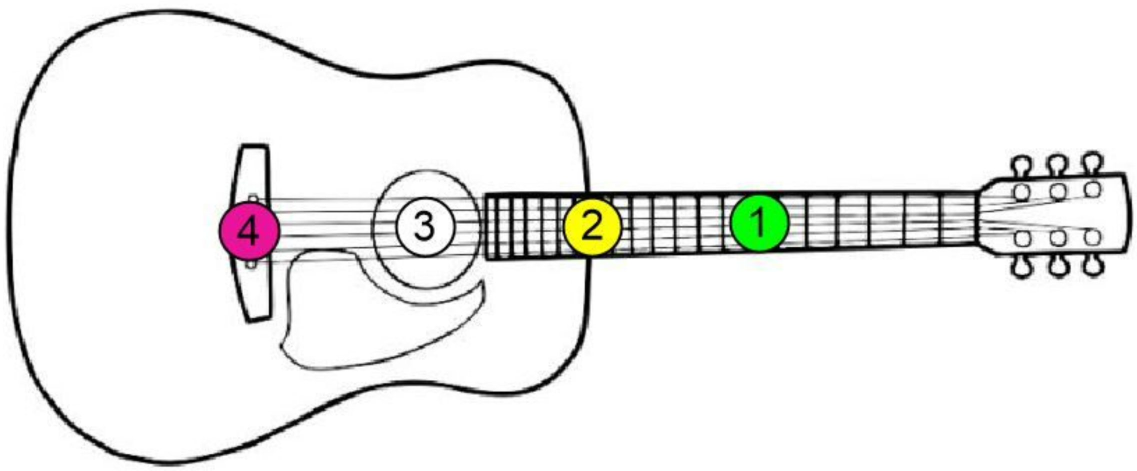

Figure 8.2 Acoustic Guitar Microphone Positions

Figure 8.3 Acoustic Guitar Microphone Positions Bird's-Eye View

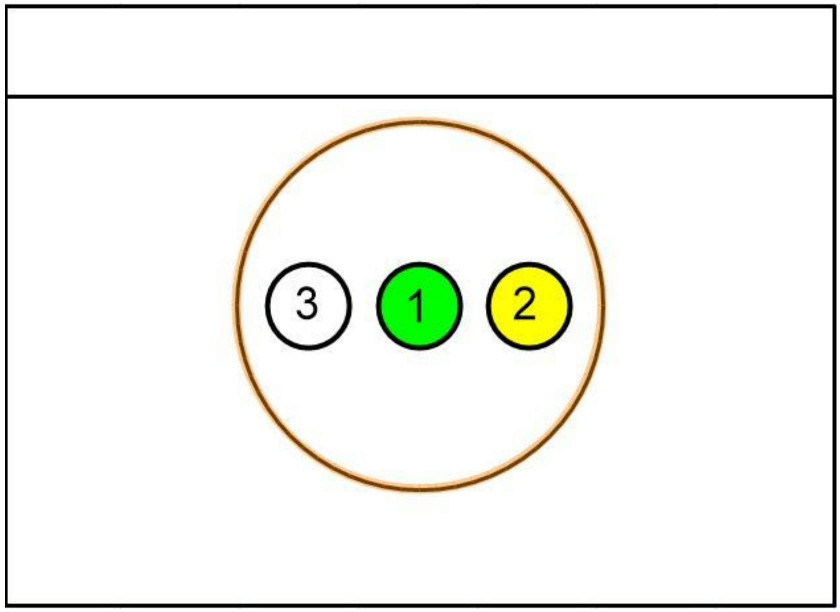

Figure 8.4 Electric Guitar Cabinet Microphone Positions (Position 4 Not Shown)

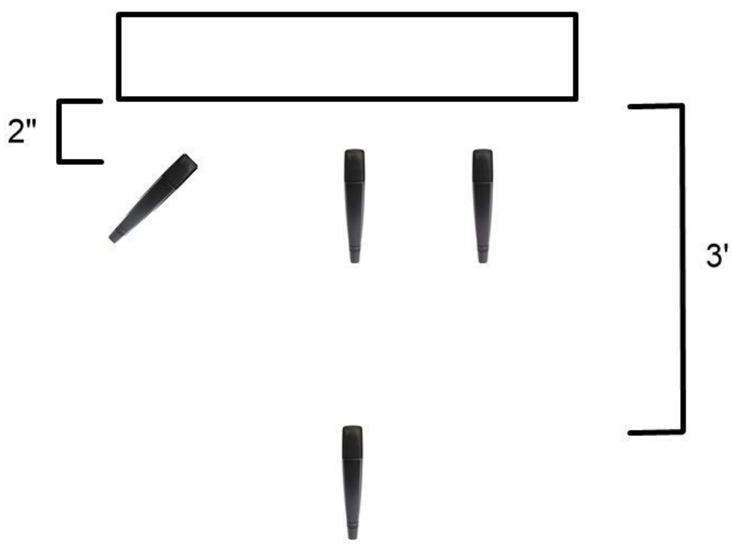

Figure 8.5 Electric Guitar Cabinet Microphone Positions Bird’s-Eye View

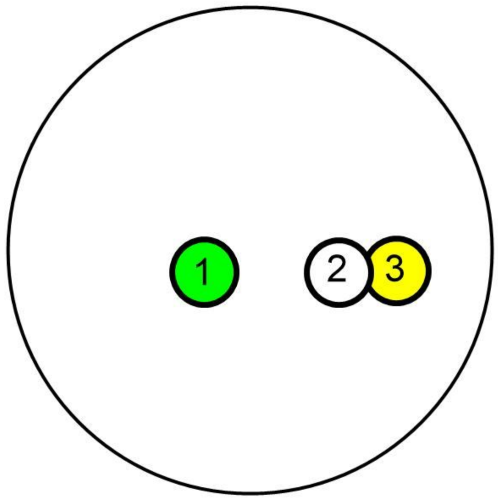

Figure 8.6 Kick Drum Microphone Positions

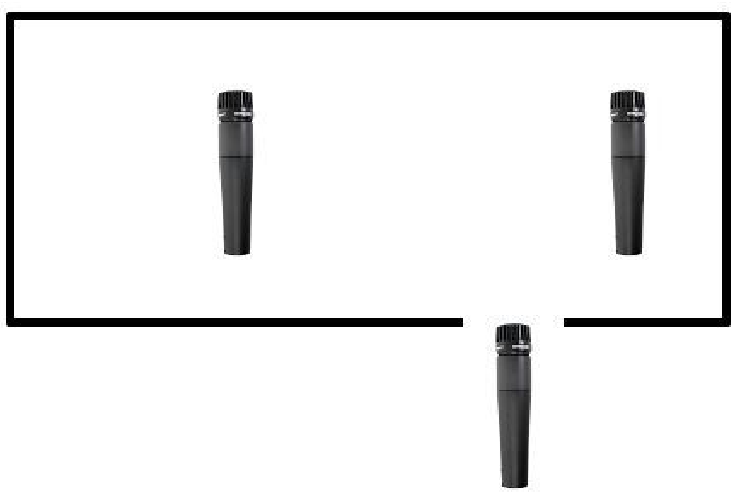

Figure 8.7 Kick Drum Microphone Positions Bird's-Eye View

Samples were recorded at Middle Tennessee State University's Studio A. I used Sennheiser MD-421 microphones for the acoustic guitar and electric guitar cabinet recordings, and Shure SM-57 microphones for the kick drum, due to their ability to handle high sound pressure levels from the instrument (Shure Inc, 2015). Distances from the microphones to the instrument were measured in order to ensure accurate placement. Before recording, I matched the input levels of my signal paths going through the API Vision console into Pro Tools using a reference tone (Figure 5). Audio was recorded and edited in Pro Tools 10 and 11; video was edited in both Pro Tools 10 and MAGIX Movie Edit Pro 2015 Plus. I used a Canon DSLR Camera to record video.

The instruments I sampled were a Larrivee acoustic guitar, a Carvin electric guitar plugged into a Bugera cabinet, and a Mapex kick drum. The musical selections were intended to be generic while demonstrating broad spectrum sounds, such as strumming a guitar instead of picking individual notes. The length of the musical selection was determined by estimating the amount of time it would take a listener to acclimatize to each new sample being played and, after establishing that reference, transition to each subsequent sample in the same manner. All music was performed to a click-track to ensure that a poorly timed recording did not distract from the focus of the project. Photographs were taken of all the recording setups and were included in the video portion of the project.

In order to record different microphone positions for the exact same performance, I positioned sets of microphones on the sound source and recorded them simultaneously while playing the instrument. For the electric guitar cabinet, I was able to record the guitar directly into Pro Tools first and then play that performance back through the guitar cabinet where I recorded it with multiple microphones in different positions. This is a technique known as re-amping. The result was the same as recording microphones simultaneously during a live performance. Since I played the role of both musician and recording engineer, re-amping allowed me to focus on engineering once the performance had been recorded.

The audio/visual portion of the project was recorded during two separate sessions at Studio A. After recording was completed, I used a transfer function in SMAART to determine quantitatively how microphone position affected frequency content. The audio was then time-aligned to its appropriate position in the video in Pro Tools and exported to MAGIX where the different video segments were arranged and pointer, number, and transition effects were added. The final product of MAGIX went back into Pro Tools in order to adjust the listening level of certain portions before being exported as a finished .mov file.

Conclusions

When I recorded samples with multiple microphone positions, each one contained variations from the others in both tonality and level. In order to measure quantitative differences between microphones, I used SMAART to obtain the average frequency of each transfer function for the full performance. Since any measurement microphone is subject to the same positional effects as the test microphones, it is difficult to create meaningful neutral references for real-world sounds. Consequently, I compared each microphone to the first microphone position as it was presented in the video and Figure 8. This resulted in three different transfer functions for the acoustic and electric guitar and two for the kick drum. I included a screenshot of each as well as one of all the transfer functions together per instrument (Figure 9.1-11.3)

The acoustic guitar frequency response chart readings (Figure 9.1-4) demonstrate quantitative differences between the positions. Without hearing the recordings, it is possible to use these graphical representations to predict how each microphone position might sound. Using the first microphone position, the one directly over the neck of the guitar, as a reference, the transfer function illustrates how the other microphone positions are dissimilar. When SMAART analyzed the difference between the neck-joint position and the neck position (Figure 9.1), a boost of 12 decibels is apparent between 63-80Hz, which gradually slopes down to 0 decibels around 150Hz. The rest of the audible spectrum is a series of troughs ranging from 6 to 12 decibels less than the reference position. These dips are due to the phase differences between the two microphones, as shown in the middle graph of Figure 9.1. The long, diagonal slashes represent frequencies that cancel due to sound waves superpositioning with each other at 180 degrees phase difference. Areas where the phase graph is discontinuous represent phase cancellations. The soundhole position microphone, when compared to the neck position, has a 17-decibel boost at 63Hz that slopes downwards to 0 decibels of difference from the reference around 200Hz. The phase difference between the two microphones is also greater and the frequency response chart reflects this with sharper dips and peaks. 250Hz, 400Hz, and 1.5kHz all stand out from the troughs present in the transfer function. The last microphone, the bridge position, exhibits a cut of between 12 and 6 decibels between 63 and 80Hz as compared to the neck position. There is a 6-decibel boost around 150Hz. Above 500Hz, the phase response becomes very decorrelated and peaks and troughs in the frequency response are the result.

Figure 9.1 SMAART Transfer Function Acoustic Guitar Positions 1 & 2

Figure 9.2 SMAART Transfer Function Acoustic Guitar Positions 1 & 3

Figure 9.3 SMAART Transfer Function Acoustic Guitar Positions 1 & 4

Figure 9.4 SMAART Transfer Function All Acoustic Guitar Positions

The quantitative measurements of each position's relative frequency content is a useful tool to describe what individual frequency bands are modified as a result of microphone position, but it is the qualitative aspects of each sound that are of the most use to the audio engineer. To describe a sound qualitatively uses words that make generalizations about the character of the sound. For example, it is widely agreed among engineers that sounds with more low end frequencies are considered "darker" and more high end frequencies result in "brighter" sounds. There are also less universal terms that can be used to try and paint a word picture of a sound. When audio has a lot of energy in the 8kHz range, that could be described as "sizzle" and frequencies around 16kHz produce "airy" sounds. Engineers have also been known to describe frequency ranges as colors, shapes, and even foods. This is why recording samples was absolutely crucial to this project; no matter how descriptive the language used is, the lack of universal understanding of these terms means that something as multifaceted and complex as audio can never be properly expressed through a description. Without making the attempt at a qualitative expression of sound, however, engineers would lose a great deal of communicative ability, even if it is less than perfect.

With the acoustic guitar microphone positions, the neck position offered a bright sound with little low end support. The overall impression of the instrument being played is that it sounds fairly natural for a smaller body acoustic guitar. There is audible string buzz which distracts from the cleanliness of the instrument. The second microphone is placed directly on axis with the neck joint. This microphone contains more low end than the previous one and the resonance of the guitar is emphasized more. The string buzz is still present but not as loud relative to the strumming of the guitar. Overall, the sound is reasonably balanced with an exaggeration in the lower midrange of the frequency spectrum. The third microphone is positioned over the sound hole of the guitar. This position sound overly boomy, with excessive low-mid and low end frequencies. The string buzz is less noticeable, but still present since it was part of the performance. Overall the sound hole position sounds unnatural, as if listening to the recording in a box. The third microphone is clearly the loudest of the four. The final acoustic guitar microphone is positioned over the bridge of the guitar, where the strings anchor into the body of the instrument. This position is similar to the sound hole position in its unnatural, boxy sound, but the lower midrange frequencies being boosted are different, resulting in a different tonal color and overall feel. There is less low end than the previous example.

SMAART analysis of the electric guitar cabinet (Figure 10.1-10.4) resulted from four different microphone positions. From a quantitative perspective, the second microphone, the on-axis, off-center one, on the electric guitar cabinet differs from the on-axis, on-center microphone primarily in a 5-decibel boost from 40-100Hz and an 11-decibel cut from approximately 5-6.5kHz. The midrange of 150Hz-1.25kHz is similar to the reference. The transfer function between the first position and the off-axis, off-center position exhibits similarities within 4 decibels for the majority of the audible spectrum from the low end until 4kHz, where a 13-decibel cut is presented along with a 20-decibel trough at 6kHz. The fourth microphone position is at a three-foot distance, on-axis, on-center. When compared to the first position, this microphone demonstrates a 24-decibel cut at 63Hz that slopes upward, reaching within 3 decibels of the reference microphone around 400Hz. The exception to this slope is a relative peak around 150Hz within 3 decibels of the reference that then slopes downward from 150Hz before resuming the upward climb to 400Hz. From 400Hz upward, the fourth microphone position stays within 4 decibels of the reference.

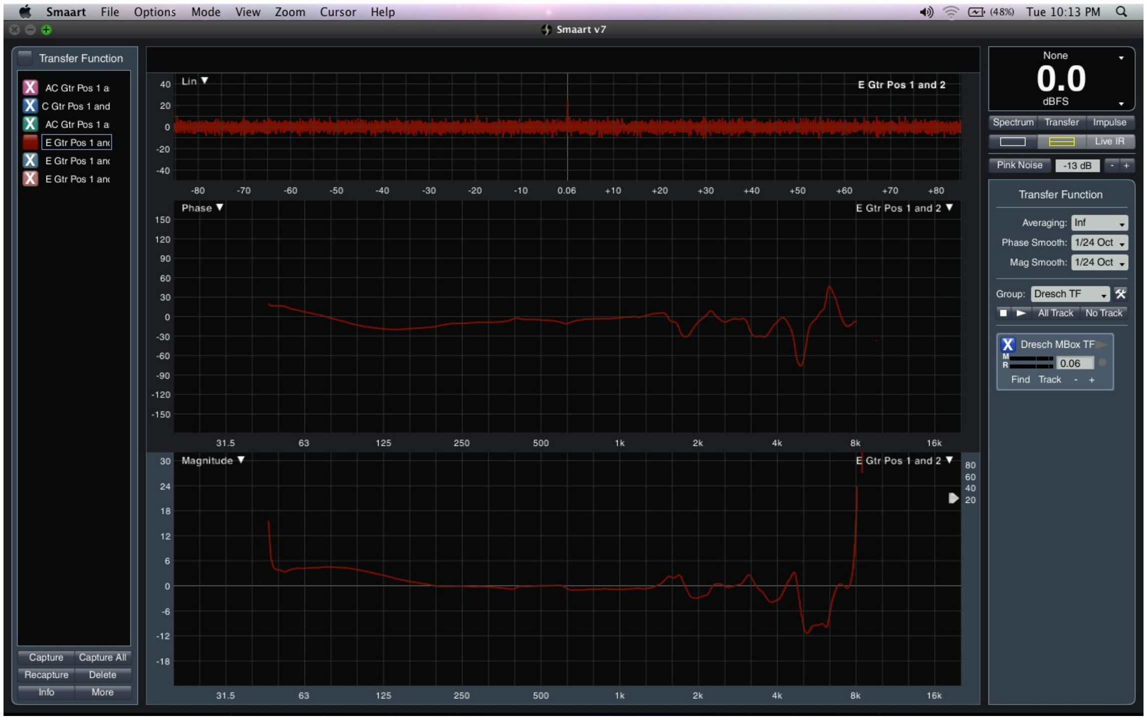

Figure 10.1 SMAART Transfer Function Electric Guitar Positions 1 & 2

Figure 10.2 SMAART Transfer Function Electric Guitar Positions 1 & 3

Figure 10.3 SMAART Transfer Function Electric Guitar Positions 1 & 4

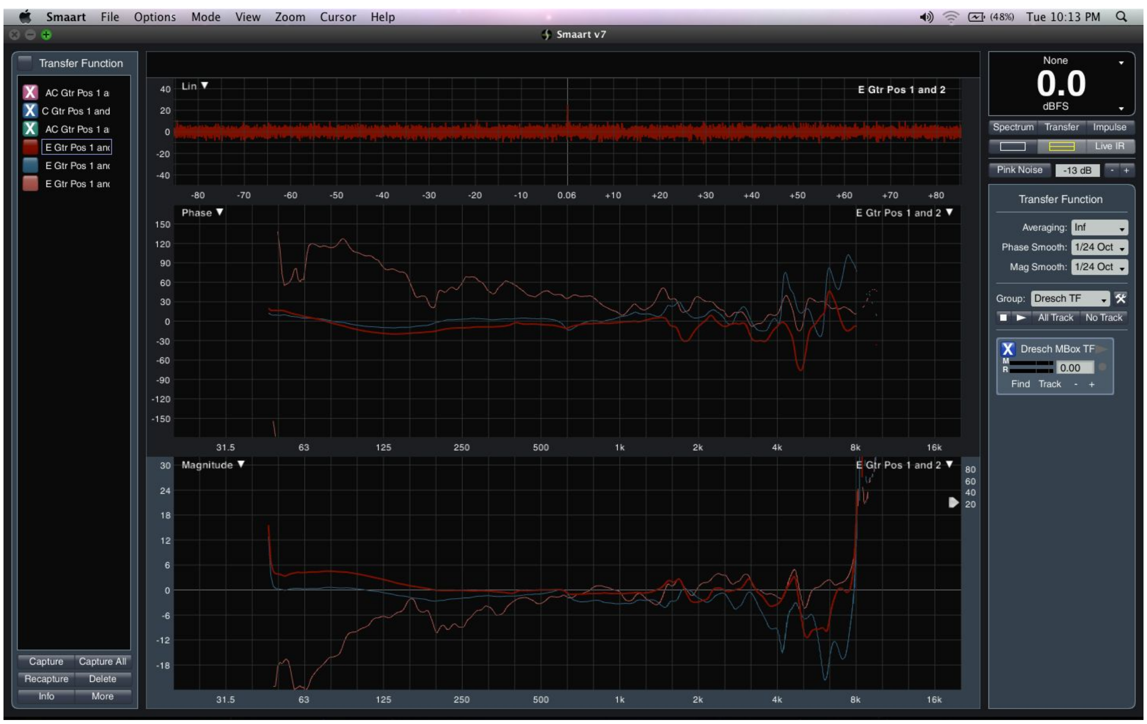

Figure 10.4 SMAART Transfer Function All Electric Guitar Positions

Qualitatively, the first, or on-axis, on center, microphone position is the brightest and loudest of the four recordings. The guitar feels directly in your face and the scraping sound of the pick on the muted guitar strings is sharp. Overall, the sound is slightly harsh, aggressive, and a little thin and bright. The second microphone position is softer, less edgy, and warmer than the first. It exhibits more lower midrange frequencies There are fewer high end frequencies. Overall, the sound is gentler than the first position but slightly dark and muddy. The 45° off-axis, off center position is even softer and more rounded than the second microphone. It is distinctly muddy and feels unnatural. There is little high end content and the sound lacks breathability. It is as if someone placed a pillow over your ear before you listened to the guitar play. The fourth microphone presents a unique challenge when attempting to describe it in terms relative to the other microphones. Without level-matching, it sounds so completely different that it is like comparing completely unrelated sound. First, it is very quiet, requiring 17dB of digital gain before the file becomes usable in any sort of commercially viable manner. The sound is small and tinny. The listener perceives a greater distance between the cabinet and the listening position than with the other samples. The overall characteristic of the sound is soft and without clarity. Once the fourth microphone is brought up to listening level, the result is bright and spanky, full of upper midrange frequencies and without any low end support. This is interesting because as the microphone moves farther away from the cabinet the sound is not perceived to be getting brighter since it is also getting much softer.

The final instrument is the kick or bass drum. When we quantitatively compare (Figure 11.1-11.3) the first and second microphone positions, on-axis with the beater strike area and on-axis with the outside edge of the interior drum head, respectively, we find that the second microphone is similar in frequency content to within 6 decibels from 20-250Hz, but has a large trough of approximately 12 decibels between 250 and 400Hz. Additionally, from 500Hz and above, there exist sharp, sporadic peaks of about 24 decibels above the reference. Since both the frequency and phase response charts are drastically discontinuous and varied above 500Hz, it is likely that these measurements are unreliable. The third microphone, positioned slightly inside the sound port, compared to the first, shows a 16-decibel boost centered around 20Hz and a 10-decibel cut at approximately 40Hz. From 250Hz and upward, the transfer function of positions 1 and 3 look similar to the transfer function of 1 and 2. With regard to frequency content, these microphone positions are similar in the range of 50-250Hz.

Figure 11.1 SMAART Transfer Function Kick Drum Positions 1 & 2

Figure 11.2 SMAART Transfer Function Kick Drum Positions 1 & 3

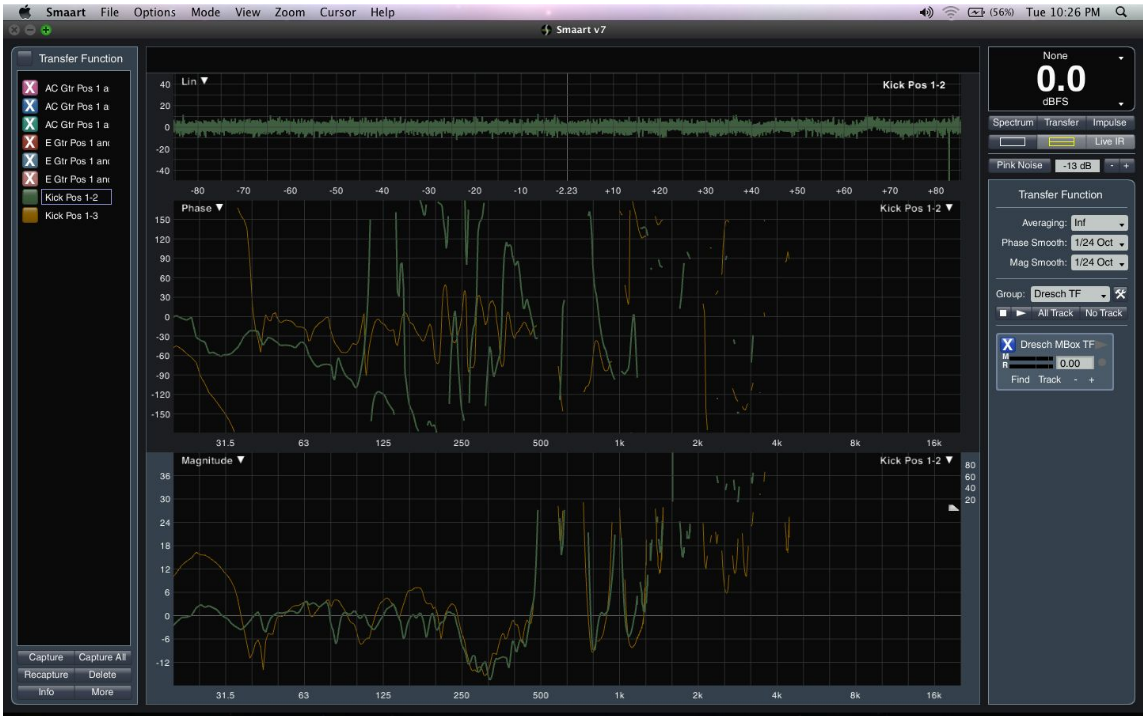

Figure 11.3 SMAART Transfer Function All Kick Drum Positions

Speaking in qualitative terms, the first kick drum microphone position emphasizes the sound of the beater on the interior drum head. The result has more of the plastic sound of the head. Since the microphone is positioned close to the center of the drum, the reflections off the interior walls of the drum are also more present in this recording than either of the others. The overall sound is tight, sharp, and has a live feel. The second microphone, the one positioned on the outside edge of the interior drum head, offers a stark contrast to the first microphone. This microphone sounds thumpy, deeper, and much more dead, or having fewer reflections. The first positioned sounded similar to bouncing a large plastic ball off a hard surface, but the second sounds more like kicking a cardboard box. The third microphone, positioned slightly inside the sound port, sounds very similar to the second microphone with more of the resonance of the drum added. This makes it feel a bit deeper and more live. It is also quieter due to the distance from the beater strike area. Of the three, the third microphone position most closely resembles the natural sound of a kick drum.

Discussion

The results of my observational analysis demonstrate how drastically microphone placement can affect recorded sounds. With regard to the acoustic guitar, each subsequent microphone position towards the sound hole became louder and contained more low frequency content. This lead me to generalize that the further towards the neck of the guitar the microphone is positioned, the brighter the resulting recording will be, and the closer to the soundhole the deeper the sound will become. On the electric guitar cabinet, microphone placement differences of a few inches resulted in sounds that ranged from crisp and aggressive to more warm sounding the further away from the center cone the microphone was placed. As the microphone is moved away from the cabinet, the overall level decreases dramatically, causing the sound to appear small and thin, but, when the gain is increased, the sound can be quite bright and harsh. The kick drum sounded very different in a position close to the beater strike area than when the microphone was farther back. Generally, the closer to the beater the microphone is, the higher and sharper the drum sound. To capture more of the body of the drum, the microphone must be positioned further back to allow the strike of the beater to mix with the resonance of the drum itself, not just the beater head.

Many of the samples I recorded do not sound like something an engineer would want to use on an album. Both the acoustic and electric guitar had one or two positions that sounded normal in a musical setting, but the kick drum had none of the power and thump that is expected from the foundation of popular music. This is due to the fact that one would not normally use the microphones I used for a kick drum. It is also unlikely someone would select a Sennheiser MD-421 as his or her sole acoustic guitar microphone. Most people consider a condenser microphone more appropriate for acoustic instruments since they are better able to bring out the clean, airy characteristics of real wood and steel resonating together. Likewise, the small-diaphragm dynamic SM-57 does not have the low end response necessary for capturing the fullness of a kick drum. Larger diaphragm dynamic microphones that can handle high sound levels are the gear of choice when dealing with a bass drum (Huber, 1998, pg. 47). Most of these mics also have large boosts in the low frequencies. The goal of this project was to demonstrate how tone changes with position. As long as the only thing that changed from sample to sample was the position and not the microphone or its gain staging, whether the recording is musically pleasant or not is irrelevant. Instrument-appropriate microphones will still demonstrate similar responses to position relative to the instrument or sound source they are recording.

The video portion of this project presented several challenges that were novel to me during the completion of this thesis. There were small things that broke up the continuity of the presentation, such as the clothes I wore changing from transition to transition and, at one point, a graduate student setting up drums in the background of the studio. The microphone I used to capture my speaking voice was positioned relatively far away from me in order to keep it out of the video shot, which was something I noticed had happened in an early recording. As a result, when I increased the gain of the microphone to a normal listening level, the reflections off the walls and table of the studio control room added noise to the recording. If I were able to redo that portion of my project, I would use a lavalier microphone or other body-worn microphone to achieve a better signal to noise ratio. The conclusions I presented in the video were also simplistic and brief, with no real explanation as to how I had reached them. I think that to create a video that was truly self-contained and provided sufficient exposure to this project would take much longer than the five minutes I used, and, as a result, would have been time-prohibitive to script, record, and edit. I think a better way of disseminating this work would have been to simplify the media format and use online audio streaming services such as Soundcloud instead of YouTube to distribute my recorded samples. This way, I could have focused on samples of more instruments and a more concise learning experience for anyone viewing my work, instead of trying to learn how to work in a media format with which I was unfamiliar.

My goal was to establish comparisons for studio microphone positioning techniques. By listening to the samples I recorded and becoming familiar with the differences between each position, an individual can clearly see what I set out to demonstrate. When it comes to microphone placement, inches matter, and understanding how recorded sound is captured differently due to microphone position is an important weapon in the audio engineer’s arsenal.

Reference List

American Speech-Language-Hearing Association. (n.d.) Classroom acoustics. Retrieved from http://www.asha.org/public/hearing/Classroom-Acoustics/

Cimbala, J. M. (2010). Fourier transforms, DFTs, and FFTs. Retrieved from https://www.mne.psu.edu/me345/Lectures/Fourier_Transforms_DFTs_FFTs.pdf

Google. (2011). Demographics: The rundown. Retrieved from https://www.thinkwithgoogle.com/products/youtube-demographics.html

Halloran, M. (Ed.). (2008). *The musician's business and legal guide* (4th ed.). Upper Saddle River, NJ: Pearson Prentice Hall.

Huber, D. M., & Williams, P. (1998). *Professional microphone techniques.* Vallejo, CA: Mix Books.

Moulton, D. (1993). Principles of multitrack mixing: The kick drum/bass relationship. Retrieved from http://www.moultonlabs.com/more/principles_of_multitrack_mixing_the_kick_drum_bass_relationship/

Owsinski, B. (2014). *The recording engineer's handbook* (3rd ed.). Boston, MA: Cengage Learning.

Rayburn, R. A. (2012). *Eargle's microphone book: From mono to stereo to surround: A guide to microphone design and application* (3rd ed.). Boston, MA: Elsevier.

Rudolph, B. (2014). In the studio: Microphone techniques to get great electric guitar sound. Retrieved from http://www.prosoundweb.com/article/studio_microphone_techniques_to_get_a_great_electric_guitar_sound/P2/

Senior, M. (Interviewer), Ainlay, C. (Interviewee), & Clink, M. (Interviewee). (2007). Guitar amp recording: Pro techniques put to the test. Retrieved from http://www.soundonsound.com/sos/aug07/articles/guitaramprecording.htm

Sennheiser. (n.d.) MD 421 II. Retrieved from http://en-us.sennheiser.com/global-downloads/file/2405/MD_421_II_GB.pdf

Shure Inc. (n.d.) Microphone positioning: Amplified instruments. Retrieved from http://www.shure.co.uk/support_download/educational_content/microphones-basics/amplified_instruments

Shure Inc. (2015). Product specifications: SM57 cardioid dynamic microphone. Retrieved from http://cdn.shure.com/specification_sheet/upload/81/sm57-specification-sheet-english.pdf

Sweetwater. (1999). Matched pairs of microphones. Retrieved from http://www.sweetwater.com/insync/matched-pairs-of-microphones-2/

Touzeau, J. (2009). *Home studio essentials.* Boston, MA: Cengage Learning.

White, P. (2001). Recording acoustic guitar. Retrieved from http://www.soundonsound.com/sos/aug01/articles/recacgtr0801.asp