AUDIO MASTERING: AN INVESTIGATION AND ANALYSIS OF CONTEMPORARY TECHNIQUES

BY

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Abstract

Mastering is an aspect of music production that is encompassed by air of mystery. There is very little specific information written about mastering and it is oftentimes overlooked in common collegiate Audio Engineering curricula. The purpose of this study is to investigate audio mastering and examine its finer details. In this study a single song was recorded; mixed by four different mixing engineers; and finally collected, analyzed, and mastered. Furthermore, interviews were conducted with five prominent mastering engineers regarding the many specific facets of mastering. These interviews were then transcribed and analyzed for common trends and practices. The goal is to illuminate various techniques for mastering while utilizing these examples in the context of a case study. Throughout this study, several themes emerged including the need for accurate monitoring, specified equipment, experience, and a touch of luck. The information gathered in this study will benefit those interested in understanding mastering. This could range from students wishing to pursue a career in mastering, to artists looking to comprehend how the mastering process affects their work.

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I. Introduction- Overall

Mastering is the least understood step in the production of professional commercial audio. This final step in the professional recording process is highly specified, competitive, and thus notoriously secretive. In the current state of the music industry with increasing numbers of home studios, professional audio mastering has become a topic of significance. This is due to the fact of the increasing numbers of artists who want their productions to be mastered, but often cannot afford to send their project to high-end mastering studios. This creates a demand for low-cost mastering that has the potential to be detrimental to the quality of music production, leading to the employment of inexperienced and ill-equipped mastering engineers. In order to combat the general lack of comprehension of mastering, information on its finer processes and details must be shared; artists, audio professionals, and the general public must be informed in order to make the best decisions for musical production and the audio industry.

Currently, there are several resources that contain information about audio mastering. There are some informal articles and videos that can be found online, such as those by Mike Collins and Johnathan Wyner, and there are a few textbooks published about mastering. Two of the most authoritative textbooks are *Mastering Audio: the Art and the Science* by Bob Katz (2007) and *Audio Mastering: Essential Practices* by Jonathan Wyner (2013). These books provide an introduction to mastering and are vital resources for learning about the

common elements of mastering including effects processing, file formats, and general mastering aesthetics.

These two textbooks are tremendous resources for attaining an introduction to the concepts of the mastering process. Furthermore, Wyner's book features two "case studies" where he walks the reader through the mastering of two sample recordings. This takes the concepts presented in the book and makes them practical, providing the reader with the implementation of specific examples of the mastering process. Even though Wyner includes these "case study" examples in his book, the details and processes of other techniques could be explained further. Because mastering is a diverse discipline and the process is specific to the genre of music, it is not enough to talk about mastering in general terms. Additionally, it is much more meaningful to discuss mastering when the concepts are put into context. Despite these great resources, there is still a lack of variety and depth of mastering literature and resources.

This study further investigates the details of mastering, providing examples for a specific discussion about the mastering process. It provides information relevant to audio engineers, producers, and artists who might not fully understand all that mastering entails. This study is separated into two sections. The first is section is comprised of a series of interviews conducted with veteran mastering engineers. This section allows mastering engineers to talk about their craft, oftentimes citing specific examples relating to their past projects and experiences. The goal of these interviews is to provide relevant and valuable insight into the current state of mastering.

The second section of this study focuses on the recording of one song; four different mixes by four different mix engineers; and the mastering of those four mixes. This section offers insight into the entire audio production chain and focuses on the final mastering of the four mixes of a single song. Furthermore, one of the mixes was a part of an EP match exercise which is another aspect of mastering that requires a differing approach than mastering for a "single."

This study combining these two sections provides relevant information about many aspects of audio mastering. It examines the responses in the interviews of mastering engineers and analyzes the mastering process in the context of four different mixes of the same song. The purpose of this study is to investigate examples that provide a context for specific discussions about contemporary mastering techniques. The following section, "A Guide to Common Practices in Mastering," lays the groundwork for this studies' investigation of the audio mastering process.

A Guide to Common Practices in Mastering

To reiterate, mastering is the most misunderstood step in the recording process. There exist rumors of mythical audio alchemy and Wizard-of-Oz-style "man behind the curtain" trickery. This lack of understanding may be true for some audio professionals, but even more so, it is especially true for audio engineering students who are primarily taught about the recording and mixing processes.

So then, what actually happens during the mastering process? What are some of the mastering methods used to alter a piece of audio? The following is a guidebook that elucidates some of the common practices of mastering, specifically in manipulating a two-track, stereo piece of music. Included in this guide are tips, techniques, and tools used to transform raw mixes into polished masters.

Politics and Communication

Before diving into the technical and artistic sides of audio mastering, there must first be a discussion on the politics of dealing with clients. Mastering does not exist without clients—period. Therefore, one of the most crucial aspects of mastering is keeping the client happy; always remember: "the client is king." Though it may seem like common sense for some, this is not an easy concept to accept for others. However, once the "client is king" mentality is adopted, both the client and mastering engineer will be pleased throughout the entire mastering process.

Therefore it is crucial to not allow ego to get in the way of the client's desires. This is not to say one cannot offer expert, educated opinions. Instead, try to educate the client about the situation as politely and respectfully as possible. In the end, if it comes down to a matter of artistic taste, trust the client. However, if it is a matter of technical limitations or misunderstandings, try to inform the client.

Another area of client-mastering engineer relations is the need for revisions. Revisions are a necessary part of mastering. Once the program material has been received and the mastering engineer does a first pass, a reference should be sent to the client. This gives everybody involved an idea of the direction in which the project is going. The mastering engineer can think about what needs to happen while the artist can get an idea of how the mastering engineer interprets and assesses the mix. Revisions are common, but the goal for the mastering engineer is to try to understand the intent of the artist and get the master exactly how the client wants the material to sound. **Figure 1** demonstrates the ideal route of communication in a music production chain where there is continual contact between everyone involved in the production.

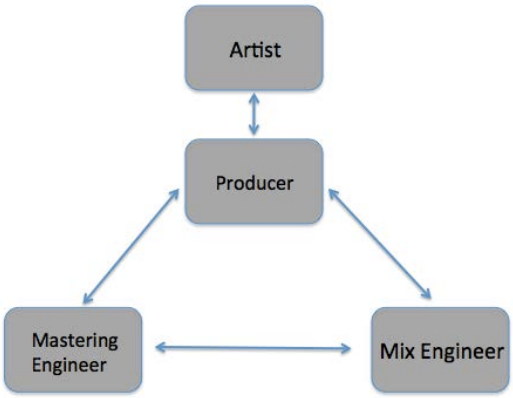


Figure 1. Communication Routes

Lastly, shootouts are another aspect of the client-mastering engineer relationship. Shootouts may not happen often but they do happen and at some point every mastering engineer will take part in one. There are two types of shootouts and for the sake of the project, they will be referred to as: “covert” and “overt.” During a covert shootout, oftentimes clients will send a mix to several mastering engineers. They will then choose whichever master they like best and send the rest of the mixes, either an EP or an LP, to the chosen mastering

engineer. This process can be underhanded when the artist chooses not to mention that the mix is a part of a shootout. In this case it is only after the matter that the mastering engineers realize they were part of a mastering shootout.

While these "secret shootouts" do occur, sometimes shootouts will be known to all of the parties involved. An overt shootout is a great opportunity to try and prove one's skills and to compete for the project. This competition-style shootout can be beneficial for mastering engineers who are just starting out. For example, say a mid-level artist is sending a mix to mastering engineers A, B, and C. Let us say that Engineer A is at a prime studio in New York City, Engineer B is at a similar space in Los Angeles, and Engineer C is a relatively unknown wildcard in Boston. It can be assumed that Engineer C is going to give the mix an extraordinary amount of thought and work. This is not to say that the other engineers will not put much effort into their work, but this gives Engineer C the opportunity to compare his work and abilities to that of more established mastering engineers. No matter the outcome, this type of shootout is a great learning experience for everybody involved.

The Mastering Process: Typical Signal Path

It is important to remember that mastering is a personal ordeal for the mastering engineer. He or she typically spends years, if not decades, honing their craft and dialing in exactly what workflow works best. Some people prefer to work entirely in the digital domain, others prefer analog, and most mastering

engineers utilize both analog and digital processors in the signal path. This is called a *hybrid* signal path when both analog and digital components are used.

Keeping in mind that every mastering engineer has his or her own workflow preference, the following is an outline of a typical hybrid signal path. Again, this is the most common process in mastering today, especially for non-classical applications. This outline highlights the process of working with a client who has delivered the mixes as digital audio files.

The mastering process begins when the client sends the files to the mastering engineer. This can be done via any kind of digital transfer medium: a flash drive or more typically, a file uploaded to web-based cloud storage like Dropbox, Google Drive, or WeTransfer. Oftentimes larger mastering facilities will have their own secure servers to which the client can upload files.

Once the files have been received, the mastering engineer will import the file into his or her choice of Digital Audio Workstations (DAW). There are a myriad of DAWs that can facilitate mastering, but the two that are most common among professional mastering engineers are Magix's Sequoia and Merging Technologies' Pyramix. Both of these DAWs are known for the ease of operation in terms of workflow and editing capabilities. For example, the fade editing on these programs are intuitive and flexible to the point where one can create any desired fade effect. What separates a mastering DAW from any other is its ability to author disc information. This includes metadata entry, track spacing, and the creation of DDP (Disc Description Protocol) files.

When a mastering engineer receives a mix from a client, they look for the following four things:

- **Lossless Format** Typically this would be either .aiff or .wav files.
- **Correct Sample Rate and Bit Depth** The rule of thumb here is that the mix engineer should not do any sample rate or bit depth conversion at all. It is best when the mix is sent to the mastering engineer in the same format it was mixed in. For example, if the song was mixed with a sample rate of 96kHz, it should be exported and sent to mastering at 96kHz. It is also crucial to have the file with a bit depth of 24 bits. This allows for optimal resolution in processing, especially in the digital domain. Additionally, this allows the mastering engineer to do the final conversion for CD quality, Red Book standard, 44.1kHz 16 bit files. The mastering engineer will be able to utilize the best sample rate conversion and dithering for this process. Lastly, beware of ultra-high sample rates and bit depths like 192kHz 32-floating point because some plug-ins cannot operate at that high resolution. Sample rates and bit depths of 96kHz 24-bit are perfectly adequate for non-classical applications.
- **No Fades** Sometimes the mixing engineer will want to add the fades before mastering. This can cause some problems with the level of the signal while processing. Without the fade, mastering can be completed without any drops in level before it hits the processing chain. The mastering engineer will add the fades to the file after the file has gone through the signal processing.
- **No Limiting** Another temptation of the mix engineer is to put a limiter on the output of the mix in order to keep the peaks of the mix from clipping.

Alternatively, the mixing engineer could lower the level of the entire mix so that the peaks do not go above 0dB FS (Full Scale). It is preferred to have files that have peaks of around -3dB FS, but it is acceptable to have mixes that reach, but do not exceed, 0dB FS. A mix of -3dB FS will allow the devices in the mastering chain a fair amount of headroom so that there is no distortion or clipping at the input stage of the devices or plug-ins.

Lastly, it is important to distinguish the difference between mix bus compression and limiting the mix. It is perfectly acceptable to receive and work with a mix that has appropriate mix bus compression. The mix engineer may decide to use type of compression as an artistic decision by to "glue" the mix together.

Assessing the Mix

Once the mix has been received and meets the above criteria, the mastering engineer will import the file into the DAW. He or she will then observe the mix, listening for anything needing to be altered, and will look for any abnormalities in the file itself, making sure that the file is ready to be mastered. This is an important step in the process when first impressions are made.

The mastering engineer will get a sense for the direction and goals the client is striving to achieve. The engineer should then ask the following questions: What genre is the material? What kind of processing is needed? How will the processing affect the mix? What kind of processing is appropriate for this style of

music? These are important questions because a seasoned mastering engineer will know exactly what kind of processing is best for each style of music. For example, he or she will probably not apply a lot of compression or distortion to a classical or jazz piece. However, they will probably add a fair amount of compression and distortion to a rock or pop mix. This assessment can be difficult at first, but experience will expedite the situation once the mastering engineer is familiar with his or her room, monitors, and gear.

Another helpful tool in assessing a mix is having reference material. This usually consists of previously released commercial material that is of a similar genre. If the client sends a mix of hip-hop music, the mastering engineer should have a collection of hip-hop recordings that he or she is familiar with. The mastering engineer would know every characteristic of the references and would then compare the mix to the reference. When comparing, the mastering engineer may ask himself: What is the timbral spectrum of the mix compared to the reference? What is the level of the reference and can the mix attain a similar loudness? It is most helpful when a client recommends a reference recording. That way the mastering engineer will know the exact direction the client is moving toward and then no assumptions need to be made.

The loudness level at which a mastering engineer listens to the music is critical. We know via the Fletcher-Munson Equal Loudness Curves that human hearing is best perceptive across the frequency spectrum at around 85dB SPL (Sound Pressure Level). Much farther above or below 85dB SPL will skew how the music is perceived. For example, if one listens louder than 85dB SPL there

will be an increased and exaggerated representation of low frequencies. Meanwhile, the opposite is true; listening at lower levels will result in a lack of bass presence. **Figure 2** shows the Fletcher-Munson curves with the X-axis being intensity in dB SPL and the Y-axis is the audible frequency range.

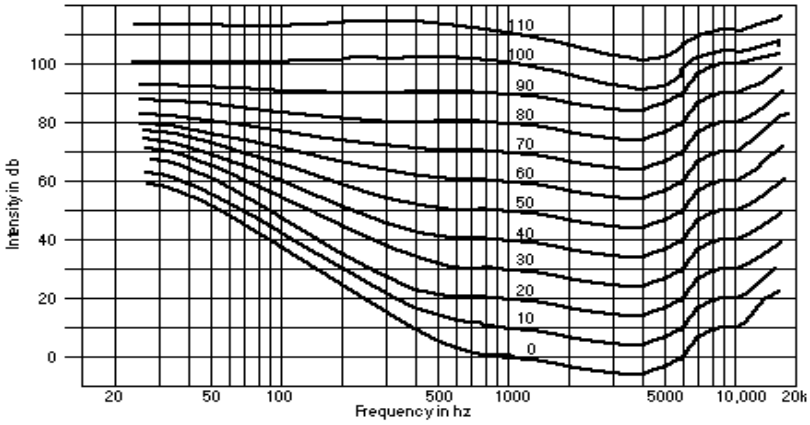


Figure 2. Fletcher-Munson Curves

With that said, mastering engineers should know exactly where 85dB SPL is on their monitor controller. They may have a piece of tape marking the listening position or they may simply be accustomed to listening for that loudness and are familiar with it. Either way, it is important to have the loudness level remain constant throughout a mastering session as this gives the mastering engineer a clean and even slate from which to work. If he or she plays a reference track and it comes across as being louder than the master, he or she can then attempt to get more loudness from their master (if appropriate to the music).

The Expertise of a Mastering Engineer

Before diving into the finer aspects of mastering, it is important to state that the most important elements in mastering are the engineer's critical listening skills and musical intuition. This takes time to develop and is what separates decent mastering engineers from great mastering engineers. For example, a mastering engineer needs to be able to tell the difference between compression ratios of 1:1.5 and 1:2; between an EQ cut of 0.3dB and 0.4dB; or the difference between Redco or Siltech cables. These seem like small differences, but they are decisions that are made on a daily basis and can have a tremendous impact on the material.

Monitor Path: Flat and Pristine

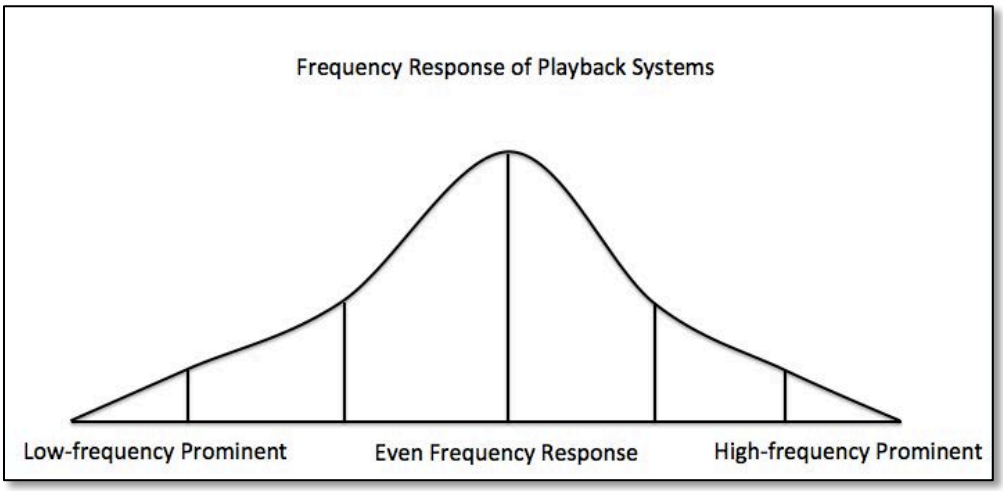
A discussion on mix assessment cannot be made without emphasizing the importance of a pristine monitor path and listening environment. A mastering studio must absolutely be designed to be as neutral as possible, incorporating a Zen-like harmony of acoustical design, treatment, and frequency response of the monitors. This neutrality of frequency response is essential in order for everything to be heard evenly. It is not uncommon for a mastering engineer to make a decision to one-half of a dB in both frequency and dynamic domains. This slight alteration may be subtle, but can have a dramatic effect to the material as a whole.

These moves can only truly be understood when the monitor path is neutral and accurate enough that a change of a half-dB can be heard. Thus, again, the slightest move can lead to a dramatic outcome to the material and will influence how the music translates and is perceived on various listening systems.

There are some whose school of thought is that mastering should be done with a monitoring system similar to what the audience is using. That way, the mastering engineer can make the music sound as good as possible on an average playback system. After all, most people will be listening on laptop speakers and inexpensive ear-buds. This at first sounds like a good idea, but it only works if everybody is listening on that same exact playback system: the master will not translate across other kinds of playback systems.

The only way to have consistency is to master with monitors that have a completely flat frequency response. One common way to think about this issue is with a bell curve graph. The extremes of the bell curve show playback systems with that are low-frequency prominent and then other playback systems that are high-frequency prominent. Within these extremes lie all the different playback systems with a completely flat frequency response is in the middle. This shows that most playback systems fall near the center of the bell curve, see **Figure 3.**

Figure 3. Playback System Bell Curve



There are many elements that factor into the equation of listening room neutrality and these elements are not to be overlooked. As a result you will find that a majority of the budget for equipping a mastering studio is spent on room design, speakers, amps, and cables etc. Mastering-grade speakers, amps, and cables especially are most often found among the "audiophile" market and not necessarily the typical pro-audio market. For example, renowned mastering engineer, Bob Ludwig, at Gateway Mastering, worked with loudspeaker manufacturers, EgglestonWorks to create a custom speaker, *The Ivy*. EgglestonWorks' client base includes both mastering engineers and people investing in high-end listening environments like home theaters.

First Steps: Signal Routing

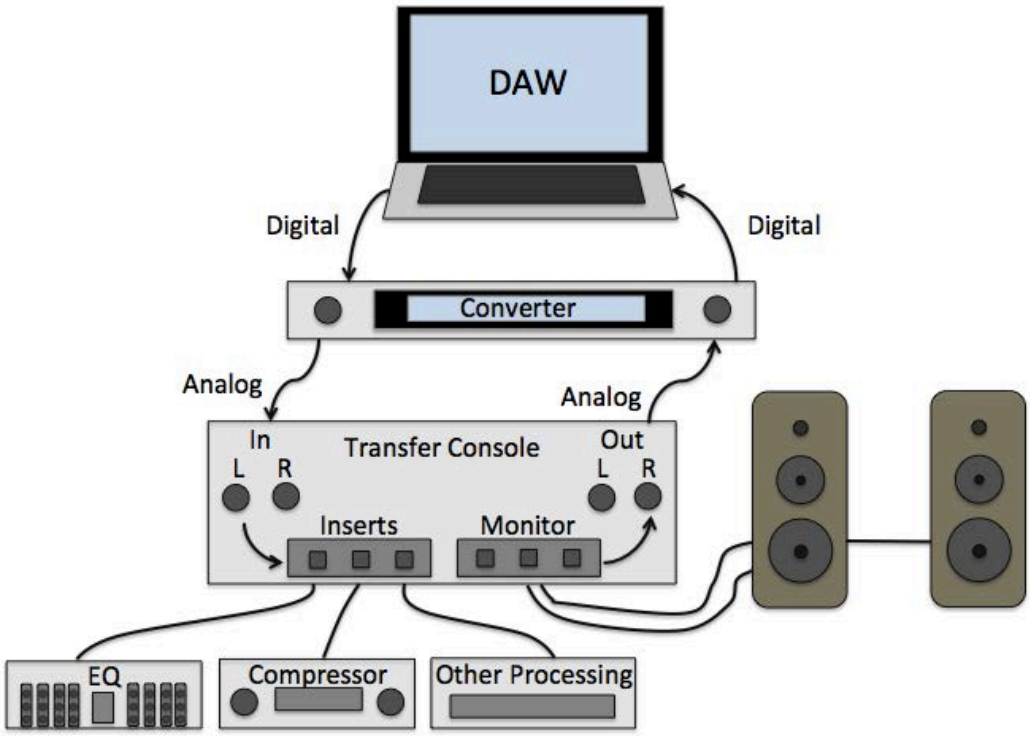


Figure 4

After having any issues resolved with the client or file, the next step is for the mix to be sent through the mastering chain. At this point, the mastering engineer will route the signal from the DAW into his or her analog signal path via a Digital-Analog (D/A) converter. There are a few high-quality converters preferred by mastering engineers. Some of the most common converters are the Prism Dream and Merging Technologies Horus and Hapi converters. **Figure 4** illustrates a basic hybrid signal path. This figure shows the routing as if the limiting is done within the processing chain itself. Sometimes, the mastering engineer will do the analog processing, convert it to digital, apply a digital limiter, then route the signal back out to the monitors.

Once the signal has been converted from digital to analog, it will be routed to a mastering console, also known as a transfer console. These consoles consist of several elements. They typically include several insert sends and returns through which one can insert various pieces of analog gear: EQ's and compressors. The insert sends and returns eliminate the need of a patch bay, which otherwise would introduce unnecessary connectors and cable length.

The transfer console also includes several mastering-specific elements. First, transfer consoles often have an insert to a parallel circuit. This is commonly used for parallel effects such as parallel compression where the mastering engineer wants to mix the signal of the affected parallel processing into the unaffected signal. The transfer console would also have a level control on the parallel circuit so that the mastering engineer can dial in the exact amount of the parallel effect.

Second, another mastering-specific element to a mastering console is an M/S (mid-side) insert. This allows the mastering engineer to use a stereo device not in its usual stereo left and right, but in mid and side (sum and difference). M/S can be used for both timbral and dynamic applications. For example, M/S processing can be used if a vocal is in the center of the stereo image and is abundantly dynamic to the point where it sticks out too much at points relative to the rest of the arrangement. The instruments in the sides, left and right, are adequately controlled dynamically so the mastering engineer can insert a compressor in M/S and be able to compress just the center of the stereo image where the problematic vocal resides. Once the compressor is dialed in, the vocal

in the center of the stereo image will be compressed to match the rest of the arrangement without affecting their dynamics. Similar timbral issues can be solved using an EQ in M/S. This can be a powerful tool, but just like anything else in mastering, the effect is best used cautiously because M/S can potentially cause time-domain phase anomalies and unnaturally distort the stereo image.

The last stage of a transfer console is a final output gain stage. At this point, the mastering engineer can control the amount of signal that leaves the console making it appropriate for whatever medium the signal will be captured. In this case, the output of the console will feed the input of an analog-to-digital (A/D) converter. The gain stage is there so that the signal is at a suitable level to the converter. It is desirable to have a healthy amount of signal going to the A/D, but it is also important not to clip the input of the A/D, unless that is an effect the mastering engineer is attempting to achieve. Some common transfer consoles include the Dangerous Master and Liason, and the Maselec MTC-1X.

Timbral Balance: Application of Equalization

At this point in the signal chain, it is important for the mastering engineer to react to the artistic elements of what he or she hears. In rare events, the mix could be so pristine that there is not anything for the mastering engineer to do. In this situation, there are no timbral issues and the dynamic aspect of the mix is perfect. The only thing left to do is raise the mix to an appropriate level and prepare it for export.

Unfortunately, this is not always the case and more likely there is yet work to be done. Generally the first place the mastering engineer will start is with an EQ where he or she would attempt to resolve any timbral issues before the dynamics are addressed. The tonal results of dynamics processing can be affected afterward if need be. With most mixes, there are often many small timbral issues, or unnatural frequency resonances. Sometimes the mastering engineer will do small EQ adjustments in the digital domain before the signal is sent to the analog chain. He or she may use a precise digital EQ to target any specific problematic frequencies, whereas larger, more general frequency domain issues can be easily resolved with analog EQ. The scale of EQ moves in mastering is in the half-dB range and is thus much more precise than the mixing stage.

Analog Minimum-Phase EQ vs. Digital Linear Phase EQ

It is important for the mastering engineer to realize that any adjustment in EQ has consequences in the time domain of the affected frequency range. This time domain alteration is referred to as ringing, smearing, and phase-shifting. In the analog domain, the phase-shift that occurs is referred to as “minimum phase.” This is a complicated concept, but it essentially means that the ringing caused by the EQ happens after the transient event of the affected frequency range. Conversely, digital “linear-phase” EQ processes the ringing differently. Because computers have the ability to buffer and automatically delay signal processing, this can allow the computer to realign the ringing so that it is centered on the

transient event of the affected frequency. In other words, it takes the ringing that would typically happen after the event and moves it equally both before and after the event.

One of the main differences between minimum-phase and linear-phase EQ is that the linear-phase EQ can potentially "blur" the transient by putting some of the ringing at the front of the transient. The minimum-phase EQ retains the fidelity of the transient and the ringing is oftentimes at a low enough level that the remaining signal will mask the ringing. After some time, the mastering engineer will understand which style of EQ works best in each specific situation (Wyner, 2013).

Flavors of EQ in Mastering

In mastering, as well as mixing, there are many types of EQ. There are bandpass, shelving, bell, proportional Q, and Baxandall filters, to name a few. Each of these styles of EQ is used in mastering in various ways. Typically the bandpass, shelving, and Baxandall filters are used on the extremes of the frequency ranges. They can be used to control and tame any anomalies in low and high frequencies. One specific use of a highpass filter is in dance, electronic, or hip-hop music, sometimes the mastering engineer will cut out some of the extreme low frequencies. This allows the impact of the kick drum and bass sounds to be present without letting the extreme low frequencies muddle the upper harmonic transient. With this technique, it is important to not cut the fundamental

frequency. The aim is to cut just below the fundamental and allow the "punch" of the transient to come through.

Another situation in which an analog EQ would be used is if the mix (for example the vocal) has an abundance of frequencies in the 600Hz range. This frequency area can be described as being "boxy" or "muffled." The mastering engineer would then reach for the EQ, search for the most prominent frequency and then with a bell curve, reduce the level of that frequency. The amount of reduction and width of the Q-value are also elements in play and it is up to the discretion of the mastering engineer to adjust those parameters.

Once the problematic frequency range has been reduced, the mastering engineer will then assess the mix for any further problematic frequencies. Another typical mastering EQ technique is when the mastering engineer is much more likely to use reductive rather than additive EQ. If well trained, he or she would know that a reduction of problematic frequencies is more efficient than raising other frequencies to match the problematic frequency.

With any artistic decision in mastering, there is a balancing act of compromises. With EQ, the balancing act exists in the question, how much is too much? For example, where is the line between "bright" and "harsh"? Both refer to high-mid and high frequencies, but "bright" generally offers a more pleasurable listening experience than "harsh." So, what is the correct amount of EQ? This is where the mastering engineer's experience and intuition comes into play. He or she could also use their reference recordings in the situation. Furthermore, the mastering engineer can trust their room, speakers, and ears to give a reliable

representation of the frequency content of the mix. This combination of techniques and experience will lead to making the correct decision about the amount of EQ. Additionally, it is important to understand the context and become familiar with the genre and the geographic region of where the mix originated; different musical cultures have various expectations on the timbral relationships of the finished master.

Optimizing Loudness with EQ

The topic of perceived loudness is commonly discussed and debated amongst mastering engineers. There are certain EQ techniques which mastering engineers use to optimize loudness. This again brings up our old friend Fletcher and Munson and the Equal Loudness Curves. Remember, the graph plots the threshold of hearing in the frequency and amplitude domains. The graph also shows that humans are most sensitive to frequencies around 3-4kHz and less sensitive at the extremes of the audible frequency spectrum. A clever mastering engineer can use this psychoacoustic phenomenon to optimize the perceived loudness of a master by highlighting the frequencies where human hearing is most sensitive. This can be done by both cutting out some extreme high and low frequencies and also boosting high-mid frequencies. This technique can be helpful especially when a there is a mix with a lack of high-mid presence. It is also critical to not push this technique too far as it can easily create an unbalanced and "harsh" sound. If this technique is used poorly, the master may sound acceptable

for a few seconds of listening, but after a while it will start to become a fatiguing and unpleasant listening experience. For a visualization of this, please refer back to **Figure 2**.

Common EQ Units in Mastering

In mastering there are several analog EQ units that are commonly used. This includes the Dangerous Music Bax EQ, Manley Massive Passive (Mastering Version), and the famed, George Massenburg designed, Sontec MES-432C. These EQ's are known for their quality of sound, ease of use, and precision. They all feature stepped gain and bandwidth controls for ease and accuracy of recall. They also feature boosts and cuts by half-dB increments allowing for ultimate precision in any timbral adjustment.

Compression: The Gains of Gain Reduction

There is a variety of makes and models of compressors on the market today. Some are vintage or vintage-inspired, and others feature new, state-of-the-art technology. Some are used for mixing and some for mastering. So then, what are mastering engineers looking for in a compressor? The one with the most knobs! Seriously, it is the truth. Mastering engineers need compressors with which they can manipulate every parameter with exacting detail and precision. Features of mastering-grade compressors include low ratio settings, sidechain EQ,

various rectifier circuits or transformer options, link and unlink switches, and precise attack and release times. Additionally, compression is often used in parallel applications.

Compression Settings and Options for Mastering

The following section is a brief explanation of the aforementioned parameters manipulated by mastering engineers while using compression.

- **Ratio Settings** Mastering engineers use a compressor on the entire mix and the slightest amount of compression can have dramatic results. Because of this, mastering compressors are equipped with relatively low ratio settings. Common mixing compressors, like a UREI 1176, will have ratio settings of 4:1, 8:1, 12:1, and 20:1. In comparison, mastering compressors, like the Dangerous Music Compressor, features ratio settings of 1:1, 1.4:1, 1.7:1, 2:1, 3:1 and also some higher ratios. The Dangerous Compressor is a great example of a mastering compressor because of the range of ratio options. It has many low ratio options and a few higher ratio settings continuing up to 4:1, 6:1 and 20:1.
- **Sidechain EQ** This is an extremely powerful option in the realm of mastering. Some units have EQ filters built into the detector circuits of the compressor known as sidechain EQ. One example of where sidechain EQ would be very useful is in a rock mix that has too much high-frequency “snap” relative to the lower-frequency fundamental of a snare drum. The mastering engineer could use

a traditional EQ to target and reduce the high frequencies or they could use the sidechain of the compressor to introduce some **additive** EQ to the target frequency range. This boosted signal of the target frequency range would then allow the detection circuit of the compressor to react more dramatically to that frequency range. In other words, the sidechain EQ allows the problematic frequencies to be raised and thus be detected and reduced more intensely.

More commonly, the sidechain EQ is used to reduce the pumping artifacts of compression. An example of this would be in a hip-hop mix that has a lot of low-frequency content in the sustained bass and kick drum sounds. The sidechain EQ can be used to implement a highpass filter to the mix to allow the sustained low-frequency information to pass through the compressor unaffected. This way the compressor will not react to the kick and bass, which otherwise would have caused unwanted pumping of the low frequencies because of ill-timed attack and release.

- **Options in Circuitry** Compressors such as the SPL Iron and Shadow Hills Mastering Compressor have many different options in the electrical components of the circuitry. The SPL Iron is a tube-based compressor that features six different rectifier circuits, each offering a different timbral character. The Shadow Hills compressor features three different output transformers: iron, nickel, and steel, which also offer an array of sonic palates.
- **Linked and Unlinked** The link feature on compressors like the API 2500 allows the compressor to be used in a few different ways. With the left and right channels completely linked, each channel will react identically no matter the difference of

level between the sides. The unlinked option will allow the two channels of the compressor to react independently to each other. This can be used in two situations. The first by unlinking the compressor on a typical left and right stereo track. This can add to the perceived separation of the stereo imaging. Therefore it is important to always listen to the stability of the imaging while unlinking the compressor. For example, if the vocal is in the phantom center of the stereo image, the vocal is equally in both the left and right channels. If there is a high-amplitude event in either the left or right channel and it gets compressed, the stability of the vocal image will be compromised. In other words, the vocal will move according to the amount of left and right amplitude variation.

The other main use of an unlinked compressor is when the mastering engineer uses the compressor in M/S where one channel is comprised of the mid and the other is comprised of the side. In the example mentioned previously, a loud snare drum in a rock mix, most of the time the snare drum is panned to the center of the stereo image so the mastering engineer can use an unlinked compressor in M/S and compress the mids differently than the sides.

- **Attack and Release Times** In general, mastering engineers utilize slow attack and fast release times. The reason for this is that the goal in compression for mastering is to not affect a single peak or fast transient, but rather ebb and flow with the musical phrases. The slow attack times allow the transients to go through the compressor unaffected and the fast release times allow the compressor to get out of the way quickly. This aim is to have an appropriate amount of compression without having any obvious artifacts like pumping.

Other Uses of Compression

Multiband Compression: Mastering engineer and author, Jonathan Wyner, introduces multiband compression in his book *Audio Mastering: Essential Practices.* Wyner discusses this processing technique and how it utilizes compression within a specified frequency range. Like most processing tools in mastering, multiband compression can be dangerous if used heavy-handedly. The most dangerous pitfall with multiband compression is the fact that there are crossover filters built into the boundaries of the specified frequency range. This, similar to EQ's, will cause phase shifting.

However, multiband compression does have its merits; allowing the mastering engineer to tailor each compression parameter to each frequency range. This can be useful because the low frequencies may require different compression processing than high frequencies (Wyner 73).

Parallel Compression: According to mastering engineer, Bob Katz, "[Parallel compression] is probably the single more potent technique to add loudness and power to a master" (Katz 213). Parallel compression is dynamics processing that occurs separate to the main signal. In other words, the two tracks of a stereo signal are routed to the transfer console, split into the primary signal, and then further into a parallel signal. The parallel signal is then processed separately and later mixed back together with the primary signal.

One of the main advantages of parallel compression is the addition of perceived loudness. In audio, loudness is measured in Root Mean Square (RMS), which is a number generated from the average amplitude of the waveform. This is the best way to calculate the overall loudness of program material. The RMS level is read similarly to peak level, in dB FS. For example, an EDM (Electronic Dance Music) track would typically have RMS levels of -4 or -5dB FS, which is extremely loud. On the other end of the spectrum, jazz or classical might have RMS levels around -20dB FS.

Parallel compression can add perceived loudness to a master by raising the RMS level. Typically, parallel compression more aggressive than overall compression. The attack is faster, the threshold lower, and the ratio higher. The aim is to reduce the peak level and concentrate more energy into a condensed amplitude range. This way, the parallel level control can be used to add the concentrated signal back with the unaffected signal, raising the overall peak value. This allows the transients of the unaffected signal to remain true, but raises the average amplitude level of the waveform.

Limiting: The Final Creative Step

Once the program material has been processed and appropriate EQ and compression have been applied (or not), the signal is then routed through an output gain stage of the transfer console where the level can be adjusted as it goes to the analog-to-digital (A/D) converter. The level should be hot enough to make

full use of the bit-depth for the best resolution. At the same time, the level should also not be too hot or else it could distort at the conversion stage. Too much clipping from the converter can be problematic, yet sometimes mastering engineers intentionally clip the converter to create distortion adding a controlled amount "edge" to the master. However, this should only be attempted with only the highest quality converters.

Once the signal has been converted and bounced into the DAW, the final stage of processing is applied. Sometimes mastering engineers use this stage to do some final EQ adjustments. However, this is primarily when the final limiting occurs. At this point, the mastering engineer will apply a digital "brick-wall" limiter to the processing chain. The brick-wall limiter is a compressor that has an ultra-fast attack time and an ultra-high threshold. Basically, it stops the signal from extending beyond the threshold, hence its name brick-wall. For this application, analog limiters are not used because they are not fast enough to deal with the demands of modern mastering. Only digital technology has the ability to do the necessary limiting in a relatively transparent way.

Next, the mastering engineer can then set the threshold of the limiter to where he or she likes. Again, this threshold is where the limiter stops the signal from going any further in amplitude. The makeup gain is commonly correlated to the limiter threshold. For example, if the mastering engineer sets the threshold to -4dB FS and there are no other parameters involved, the limiter will automatically apply a makeup gain of 4dB.

The mastering engineer will also set the final output level. This output level is generally set to -.3dB FS in order to leave ample headroom for inter-sample peaks and allow for the best lossy-format encoding.

Less is More and More is More: A Note on Processing

In mastering the adage, "less is more" is certainly true in terms of the amount of processing. EQ and compression moves are made at scales of tenths of a dB, whereas processing in mixing requires much larger increments. However, another note on processing is that each piece of gear features its own sonic character. For example, the Dangerous Compressor sounds different than an API 2500. Due to this fact, mastering engineers will stack various pieces of gear so that not one sonic signature will become too apparent. Continuing the example, perhaps a mastering engineer wants to apply 4dB of gain reduction through the use of an analog compressor. He or she could implement all 4dB of gain reduction on the Dangerous Compressor, but the outcome might have a bit too much of the Dangerous' artifacts or character. Instead, he or she can apply 2dB of reduction on the Dangerous and the other 2dB on the API. This minimizes the amount of artifacts of any one piece of gear. This is why less is more and more is more. The mastering engineer can use *more* pieces of gear, but have them do *less* work; having two-times the processing doing half of the work. This is true for most aspects of mastering, from EQ, compression, and limiting. As mastering engineer,

Jay Frigoletto, worded it during the interview portion of this project, "this is a way to 'spread out the pain.'"

The Final Step of the Mastering Process: QC, PQ, and DDP, etc.

Once all of the artistic alterations have been completed, including fades and spacing of the material, the last steps in the mastering process involve quality control (QC), metadata (track names, PQ, and IRSC/UPC codes), and the creation of the final DDP file (Disc Description Protocol). There are several elements to the quality control stage of mastering. In large mastering studios, there will be dedicated employees, assistants or production engineers, who are in charge of the quality control. They will listen through all the mastered material, listening for any abnormalities like clicks, dropouts, and anything else that may be unacceptable. The quality controller is last person to listen to the masters before it is sent to out for distribution or to web-based services like Spotify and iTunes.

This person will also enter all the metadata, which would include entering CD text for album, artist and track names. The assistant will also enter IRSC codes. IRSC stands for International Recording Standard Code and is data that is used to identify the owners of the material. An individual code is assigned to each specific track so that it can be identified by radio play, downloads, and streaming.

When all the metadata has been entered, the final step for preparing a master for CD duplication is the creation of the DDP file. The DDP (Disc Description Protocol) is a digital file that contains information about every aspect

of the master including timing, metadata, and the audio files. Creating a DDP gives the mastering engineer total control over every aspect of the album while not subjecting it to the discrimination of the CD duplicators.

So what is the standard format for the creation of CD's? Redbook is a set of guidelines created by Sony and Philips in 1980 in order to standardize the creation of CD's. The following are some of the standards, as outlined in Wyner's previously mentioned book, *Audio Mastering: Essential Practices*:

1. There is at least one start track and one end track marker.
2. Tracks must be longer than four seconds.
3. The entire disc can't be any longer than 74 minutes (650 MB of data).
4. The maximum number of tracks and index points is 99.
5. The contents of the disc are written as a singe volume (without partitions).

Final Thoughts

The following is a summary on some of the common aesthetics of mastering. These include words of advice for people who are interested in becoming competent mastering engineers.

- **Do no Harm** Legendary mastering engineer, Bob Ludwig, explains the "first rule of mastering" found in Phil Ramone's book, *Making Records: The Scenes Behind the Music* (2007):

The first rule of mastering is, 'Do no harm,' explains Ludwig of Gateway Mastering. The secret to being a good mastering engineer is being able to listen to

a mix, imagine how it could sound, and then push the right buttons to achieve the sound you have in your head. For most of the recordings I work on, great mix engineers and producers have spent lots of time trying to get it right in the first place, and I have to honor what they send me.

- **Vocal is King** It can be tempting to pay attention to and highlight an instrumental element of a mix like a "punchy" snare drum. This can be dangerous because it could mask the vocal. As a general rule of thumb, always make sure that the vocal is clear and comprehensible.

- **Balancing Act of Dynamics** One of the most discussed topics in mastering is "The Loudness Wars." This is a phenomenon based on the presumption that louder is better. This is a misguided idea because louder is almost always worse. The louder the master, the less dynamic range there is and the music becomes a homogenous glob of loud noises. It does not lead to a pleasurable listening experience. However, most artists want their music to be of a comparable loudness to what has been previously released. With this in mind, there is an art to making loud masters that seem to have a good amount of dynamic range. This can lead to an investigation into perceived loudness within the frequency domain. It can also lead to investigation of advanced compression techniques, such as parallel compression.

- **Ditch the Ego** It is safe to assume that people find the most success in collaborative work when all the people involved are even-tempered, communicative, and professional. Yes, some artists can be fooled into working

with hotshot engineers who are difficult to work with and have an ego bigger than the galaxy. However, this is not sustainable. The best mastering engineers are those who have the ability lay aside their ego in order to efficiently get the work done.

Literature Cited

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Section 1: Interviews with Mastering Engineers

Introduction and Methodology

Upon investigation into the topic of mastering, it became apparent that there was very little relevant and specific information about the mastering process. Prominent mastering engineers like, Bob Katz and Jonathan Wyner, have written several books. These books offer great overviews of mastering techniques, but generally lack specific details regarding the implementation of those techniques. The goal of this series of interviews is to go in depth into the finer aspects of mastering, to look for common trends, link these trends together to make better sense of mastering, and distinguish what is common practice in this mysterious niche.

The first section of the interview contains seven questions pertaining to various facets of mastering. The second section speaks to the usage of subjective musical language in mastering and how the mastering engineer often acts as a translator from the subjective musical language to a technical application. The following are the seven questions that were asked during the first part of the interview:

1. Could you tell me a bit about how you first got involved in mastering? Did you get into audio first as a musician or were you more interested in the technical side of audio? Did you go to school for audio? Did you intern or have an apprenticeship?

This question asks about the origin of the mastering engineer's career. It allows them to speak about their hobbies, passions, musical and technical experiences, and education. The goal is to learn from other's examples and find how people interested in mastering can successfully navigate their way into a position as a mastering engineer.

2. As your career advanced, how did your philosophy on mastering evolve? Is there anything you know now that you wish you knew when you first started?

The second question attempts to bring out any pitfalls a young mastering engineer might experience and learn how to avoid them. It also tries to shed light onto the mastering engineer's career as they discuss any periods of time when they were mistaken and unhappy with the results they were getting. Simply put, this question is about learning from other's mistakes.

3. Are there any projects that stick out to you as having a particularly challenging aspect or technical difficulty that you had to overcome? What was the nature of the challenge and how did you react to it?

This next question deals with the technical intricacies of mastering. It pertains to some common mistakes found in the mixes they receive and some of the ways they fixed the mistakes. The intent of this question is to try to find concrete, relevant information about some of the common practices in mastering.

4. With the understanding that every mastering project is different and that each piece of gear has its own unique character, how often do you find yourself working in the analog domain? Or do you prefer digital? What are some of the pros and cons that you've found with each format? Do you use a hybrid system? If so, do you do your digital processing before or after the D/A?

This question essentially asks if the mastering engineer prefers to use analog, digital, or both (hybrid). This is a hot topic in mastering because there are very strong arguments for the usage (or not) of the different styles of processing. Both analog and digital processes have their advantages and disadvantages and this question provides the mastering engineers the opportunity to share their thoughts on this subject.

5. How much communication is there usually between you and the artist or producer? Do you master the record as a whole and send it to them for revisions or do you send them a few tracks for review? What is your policy for revisions? Shootouts?

With this question, communication was put under the magnifying glass. The aim was to understand how the client – mastering engineer relationship typically unfolded. This question asked about the methods of file transfers, shootouts, revisions, references, and other such routes of communication and how were are handled.

6. There are a lot of DAWs designed for mastering like Sequoia, Wavelab, and Pyramix (to name a few). Which DAW do you prefer and why? Is it an issue of workflow or does one program actually sound "better" than the other (i.e. does one handle fades better? does one have better dithering, sample rate conversion, etc.?)

This was another question pertaining to personal preference and matters of workflow. It brought forward a discussion of mastering-specific DAWs (Digital Audio Workstation) that were uncommon to a student, tracking engineer, producer, or mixing engineer. This was because DAWs for mastering were highly detailed in aspects that were not so critical in a tracking or mixing situation such

as object editing, fade editing, entering metadata, and creating a final DDP file (Disc Description Protocol).

7. Do you have any advice for somebody who wants to start a career in mastering?

The final question attempts to illuminate the current state of the mastering industry and how somebody could successfully navigate a career in such a small, specified, and competitive industry.

Mastering Subjective Language

The second section of the interview deals with the subjective, non-absolute language used in mastering. There are many subjective terms that are commonly used in the music industry, especially when communicating with clients. When these words are used it is generally assumed that the audience knows what he or she means. However, these terms do not deal in absolutes, but rather provide an idea or guideline for musical interpretation. This can be viewed similarly to the usage of formal musical language, with words such as *piano, forte,* or *crescendo*. The goal of the second part of the interview is to find out how the definitions of these subjective terms translate from person to person and to see if there are commonalities between each person's definitions.

It is also important to mention that the mastering engineer serves as a translator, or in continuing the musical example, a conductor, between the subjective language and the technical. For example, if a client requests more "snap" from the snare drum, the mastering engineer must be able to listen to the material and assess what "snap" really is. It could be a matter of frequency content (timbral detail) or dynamic responsiveness, perhaps a combination of the two. This second part of the interview asks the mastering engineers to define the commonly used terms: ***punch***, ***warm***, ***harsh***, ***clear***, and ***groove***. It also asks the mastering engineers to define the boundaries of the frequency ranges: ***lows***, ***low-mids***, ***mids***, ***high-mids***, and ***highs***.

Results: Interviews in Chronological Order

* N.B. These interviews have been transcribed verbatim.

I. Jay Frigoletto

Jay Frigoletto is a mastering engineer currently working at Mastersuite Studio in Brookline, NH. He is a graduate of Berklee College of Music in Boston, MA and has been operating as a mastering engineer consistently since 1992. His credits include Alice in Chains, Black Eyed Peas, and Arrested Development, to name a few.

This interview was conducted over two conversations. The first was held on November 6th, 2015 at Chrysanthi's Restaurant in Brookline, NH and the second was a conversation held via Skype on Friday, December 4th, 2015.

K: Could you tell me a bit about how you first got involved in mastering? Did you get into audio first as a musician or were you more interested in the technical side of audio? Did you go to school for audio? Did you intern or have an apprenticeship?

J: I was into music from a very very early age. My dad was a Dixieland drummer back in the day. But also played a little piano, played a little guitar. So there was music in the house. He would have us as kids sit at the piano bench. He'd teach us the little tiny things and he'd play the left hand. So, definitely, I was into music

very early on and just showed an aptitude for it. Came home from kindergarten and there was a piano in the back of the class and while we waited for the bus the Nun would go back there an play something and I thought, “Oh, I’ve got one of those piano things.” My mother tells this story and I kind of went in and played Three Blind Mice. She came in and went, “Hey who taught you that song?” and I went like, “Why would anybody need to teach you?” She went, “You know that’s not that way for everybody, right?” And I went, “No, can’t you just do it?” So she was like, OK get this kid some lessons.

In addition to that I was always in to technical stuff. I love music, creativity; love drawing, things like that. But I was also into technical and engineering and I used to like be really into architecture as a young person and do like house plans and had architecture tools and read. I was a junior high kid and I was doing like architectural planning. And I’m like, OK this is getting interesting here. And I used to play with electronics and stuff. You know, I saw what kind of signals you could send over the phone, what you couldn’t and then figured out mayne it was too high of bandwidth and thought, Oh I can hook it up to the radio and I can make my own “on hold” music – alligator clips, undo the phone and hook it onto the leads. And you know, like batteries and LED’s; I’d go to Radioshack and be like, I’m gonna buy a bunch of stuff and wire up your room and make Halloween costumes with like lights.

So I was always into this technical and the creative and the music for sure. So in high school I was like, OK, so what am I going to do with that? I like both of these things. I sent letters to, being a keyboard player I was into synthesizers

and whatnot. So I sent letters to Korg, Roland, and Yamaha. I said, hey, I'm into this technical stuff. I'm trying to figure out how to put these things together, thinking of people who are there. I thought, maybe designing musical instruments and such. And so to their credit all three of them wrote back, sent me nice letters saying, Oh people here kinda did this, they have this background. But I decided, you know what, I don't know, that's cool, but I that's a little to far removed from the creative side. I want to make music with these things. That's when I started the recording thing and you know, got into a band. I did some recording and I said, Ok this is it. This is definitely what I need to do. So I decided I liked that and you know, went to Berklee and did the piano major thing. And then decided, Ok what part of audio do I want to get into? And at the time, I got out of high school in '87, took a year off and just gigged. I got accepted to Berklee and gigged for a year.

So um, I sorta started researching, again being creative and really tweaky-technical, mastering is that little more technically oriented than the mix or recording thing. You know, the attention to detail and some of the tools that are a little bit more esoteric than what would be in the regular studio. That really appealed to me. And nobody knew about it. You know, '92 is when I got out of Berklee and that was really like the Wizard of Oz behind the curtain. Mastering was like, there weren't a bunch of articles about it. We're like, what's that thing? Even the artists we like, I don't know, I finished my record and now it has to go to that guy that has like those bat ears that hears all those sorts of crazy things and it comes back sounding better. You don't know what happens. And artists used to

tell me that. And it's sorta still like that way. There's some misinformation out there, oh just make it loud and bright. But at least people know it's something. It's more of a known quantity now.

But that really appealed to me. The tweaky details and you know. And you almost feel like you're in a little club. It's some small illusive group of people that are like, hey we know a couple of things that the other guys don't know. It's like, a-ha, you don't know how to do this, but we do. You know, you start doing weird things like mid-side processing. People look at you and go, oh my god, you can do that in mastering? And I'm like, yeah totally.

So after Berklee I had decided you know, mastering is what I wanted to do. And of course, they didn't offer any mastering ideas – courses like that at the time. But I knew that was what I wanted to do. So right out of Berklee I went and got an internship at a mastering studio. And from there, ended up moving to Atlanta. I was offered a job in 'post' as a staff engineer. You know wow, I'll take that. I did that for a year but at the same time, I was looking around that scene and there was really no mastering happening down there. So I sorta slowly stared buying equipment and finally just took out an SBA loan. Just went, you know, and went in with the big business plan and went crazy, applied to the bank and got this loan. And so was able to open up a mastering studio. So, that was the way, musician – you know, music and audio school, interned at mastering, staff engineer at the post thing, opened my own place. That was that route.

K: Where did you intern?

J: With Jonathan Wyner actually: M-Works. Yeah when he first left, he was working with Toby Mountain at Digital Recording. He had just left and he wasn't at his big place now but he was in the same building. But he had a room, like downstairs. I had actually worked with him with the band I was in. We had worked at a place called Eastern Sound and Video in Methuen, which is not there anymore. And he had been like part-time engineer there. We liked him so much we were like, hey come with us. So we did some stuff at Bluejay, some mixes – we had him come. And he said, I do this other thing, mastering with Toby Mountain. And the second time we worked with him at Bluejay, he said, I'm opening my own room. And I was just finishing at Berklee at the time and I'm like, Ok that's actually, funny you should say, this is what I'm interested in doing, I want to come check it out. So I spent nine – ten months there, you know, did the internship for school and stayed on afterward. I'd go in once or twice a week, you know, help, assist, watch, whatever. And just was like, yes, this is definitely cool. This is what I want to do. I also learned at the time, he had been working a bit with the No-Noise, doing some of the restoration work, which again, was like cooler. You know, even more esoteric and technical and I'm like I need to do that. So, learned a bit about that, which really served me well because when I went to Atlanta that was one of the things I ended up doing with my loan is bought No-Noise, which at the time was horribly expensive and nobody had the real high-end cleaning tools, was probably close to a $20,000 addition to their $15,000 system. So I ended up spending 35-grand on, you know, with a CD recorder, which was

like, holy shit you can burn CD's? They were like, oh my god you have this No-Noise package and like, you have that Sonic System thing. You know, at the time it was this weird thing to have that. Like, no one had that. It was like huge money I spent – crazy, crazy big. But as a young guy, you know, I was like, Ok, I want to do this. I want to set myself apart and show people I'm serious and have something that other people don't have. And if I don't get enough mastering work, I can get a gig doing this clean-up work that's in town. And like the big places, like the really big places like Crawford didn't have those tools. They were like, you do that? They started referring all that stuff to me. It was a really smart idea to be like, Ok I'm going to bite the bullet and it's scary to take that big loan out, but I need to set myself apart. I need to give myself something unique that doesn't exist here. And that's the way I can get some more clients and that's the way I can make a name for myself, and it totally worked.

K: So you think that's still a smart thing to do now-a-days to invest in something that will set you apart?

J: Yes! Because everybody has a little thing in their bedroom that they can do… Like it's a problem. You don't want to invest so much that you're gonna be like, Ok I need 40 grand a month, you know, just to pay. Because you're gonna have this horrible rate pressure coming up from the bottom. But one way to keep your rates at a certain level is to have something other people don't have – something unique where you go, I know this, but look, this is obviously different. Everyone

has Protools. You can't be like, I have Protools. People don't understand that, you have the big HD rig and he has the little thing on his laptop. He's like, it doesn't matter, Protools is Protools and might not understand that.

But yeah, if you can figure out, OK what's a niche that isn't being served right now. You know, what's and area that isn't happening but there's a demand for it. And, can I set myself up in that niche thing that's gonna set myself apart. Now, I'm gonna be the guy, Oh he's the guy who does that thing. At the time when surround was happening, that was an idea. I got into surround, and of course there's not as much surround happening these days but in concert video, we do a lot in surround. So that's another one of those things where those things came out that people didn't know about. And I'm this technical brain that'll soak that stuff up like a sponge. Ok so I said I'll do that. The 5.1 stuff, I got into that right away. Plus I had post experience anyway, mastering in post, so I understood that. So I think that, find something that's going to set you apart, you know. And don't be afraid to invest a little bit. If you come out of the box just as one of everybody with nothing really special about you, you're going to be fighting that noisefloor to get above it. Where as, if you sorta make that investment, and again, not anything ridiculous, but just enough to get above that level a little bit, then they'll go, Oh he's a professional guy with real stuff. You know what I mean? And then you get that first impression as, Oh yeah, he's actually on this level. You know, the pyramid – he's that step up; the smaller echelon of professional guys, not just the noisefloor.

K: As your career advanced, how did your philosophy on mastering evolve? Is there anything you know now that you wish you knew when you first started?

J: It certainly evolved as I learned things but I had a pretty good feeling for what I wanted to do from the beginning and what I thought really made sense and was lucky enough to have guessed right. So obviously started out in the Boston area they headed down to Atlanta for about 6 years then sorta started feeling the glass ceiling there, like Ok you can only go so far in Atlanta. And I originally thought I'd come back to New York but New York was a pain in the butt: too expensive to live. I ended up calling Sterling which was where my favorite guys were. I like the guys who aren't just super loud and bright. I like the guys who are into that sort of, you know, Ok, you know how to make it loud but it sort of cleaner than the other guys and I'm not willing to go to that last step where it's all fuzzy. Greg Calbi was one of those guys in New York at Sterling whose work I really liked and who had a little bit more of that feel for being picky about that last little 2% of quality, you know, we're gonna go for it. And so I ended up getting through to him and talking to him and he's like, well you know we're moving to our new big space in Chelsea and we're like, you know, we aren't looking for anybody now. We're about to move so we aren't going to do anything. But he said, you know what, a client, good friend of mine, Roy Secala, who's the guy who did, who used to own Record Plant and did like, John Lennon's Imagine and all that stuff. He just moved out to LA and was at this other studio and they're looking to improve

that studio and add things. They might want to add mastering, why don't you call him? So I called up Roy and he's like, yep that would be cool. Why don't you come? And of course, I had all my gear. I had a fully functioning mastering. And they had, it was a place out there called Studio 56 and they had a back room that was like empty and they were like, what can we do in this room to make it do something, like well… So I was like, I can come fully formed. I can come with a truck and you've got that room. I will set my gear up and we'll do something. So that's how I ended up sorta in LA.

Now as far as what had sorta changed in my idea. I sorta found out the loud thing was something you definitely have to deal with. So I had to start learning about, Ok how do I make it loud without just doing that thing. You know at the time Waves had just come out with the L1 which just sounded terrible! I mean, crunchy, fuzzy… but Oh, but it's loud! That whole like, in the first five seconds you go, yay it's loud. And then ten seconds later you're like, Oh. One of the things I say is that in the first five seconds louder and brighter will always catch your attention. Just psycho-acoustically that's the way it works. It's sorta an anthropological thing that as a caveman when the tiger comes and is about to eat you, and it goes "roar" you're going to be drawn to that and go, Holy moly. You know what I mean? So it's like really good at preserving life of a caveman, not as helpful when you're trying to make music because if you make these snap judgments, you're gonna go oh course you're gonna think the louder, brighter version is better. And then you're not going to know why, you know, back in the day I used to listen to this record over-and-over-and-over and now you're like half

way through the song you just want to turn it off. It doesn't mean I don't listen to that record anymore. And a lot of people don't think about it and understand, but that's part of what's going on. Though, Ok, that whole how do I make it loud without falling into that trap, and still having it sound good and sorta maintaining that. But again, with the mastering engineer, and even the engineers that I liked. I mean, George Massenburg was like, holy moly, every freaking record that Massenburg touched sounds so good! That's that sort of three dimensional, realistic, amazing, detail. I need to go after that. So it wasn't so much that I changed, it's more that I learned more about how to get that thing that I already wanted to do from the beginning. And that still is kinda, you know to a certain extent what I'm known for. Some guys are so like, No I won't make it loud, I won't do anything, I don't care what the client wants, this is my… I'm like, no you have to do what the client wants. You have to do what the client wants a certain amount but you know, you have to also educate them and convince them through the force of argument that you know, you have the better idea. And if they don't agree, they don't agree. You have to make it sound the way they want to. But you have to at least try to give them the information and if they go, Ok now I understand, that makes total sense, I'm glad you told me about that, lets not go quite that loud, lets not quite use so much clipping. So, I'm sorta a blend of that, I'm like Ok so I'm gonna make it this commercial thing that you want. I'm not gonna go, don't worry I won't let you fall off the edge before that, you know, you're gonna have buyer's remorse in the morning. And you know, I will get your record to be very very nearly as loud as the other one, but sounding a whole

bunch better at the same time. So that's kinda a bit of what my reputation is and I get calls for that. I'm the guy who's not gonna destroy this or sometimes you get the call and they're like I just went to this other mastering engineer and he just did his usual crunchy, loud, fuzzy thing that hurts my ears, can you fix it? So it's good. And I like that position. I'm not in the position where it's like, oh he's the esoteric guy who only does jazz and classical who's not willing to throw down and make some rock-and-roll. It's kinda like, well how do we get a guy that will make rock-and-roll that's not going to be tweaky-weird like the classical guy but is gonna care more than the big-ol metal just-make-it-fuzzy guy. And there aren't enough of those guys. And I'm like, I'm that guy!

K: I think it's so important to be diverse, especially in mastering. Maybe as the client base shrinks and, I don't know what they're doing now-a-days, but being able to treat each genre of music with respect and being able to know what each genre requires.

J: Yeah exactly. And understanding. There are mastering engineers that might specialize to a certain extent and might do a little more of this than the other thing. Every mastering engineer does a little of everything. You know, mix engineers can be like the rock guy or the hip-hop guy. If you're in mastering you're going to do a little bit of everything. At some point you're gonna get a jazz gig, at some point you're gonna get a classical gig. I mean, I do.

K: Are there any projects that stick out to you as having a particularly challenging aspect or technical difficulty that you had to overcome? What was the nature of the challenge and how did you react to it?

J: There's not one that sticks out but there are definitely things and problems that happen. I mean, heck, 5.1 was a thing for sure where that was a whole big learning curve where we applied that to music and not film, where do I you know, in dynamics processing, how do I pair this? Do I pair each channel individually, do I pair all six, do I pair, five but not the subwoofer, do I pair the left-right pair, the back pair, and like the center-subwoofer, or you know, how do you do that? How do you afford all of that sound. You know, if you like the Manley Vari-Mu sound I'm not gonna buy two more of those. I'm not that rich, you know? What do you do? And then just the whole sample-rate converter stuff. Like, Ok well what was it shot at, do they want a CD of it as well, or is it going to be video, so it's 48k. So it's a much more complex job. And you know, sync'ing to the video. I worked in post so that stuff was great for me. It made more sense to me.

But as far as other stuff, there's not one job that was like, oh wow this is crazy, how am I gonna fix this. There's problems that come in all the time that are unique and other problems that you hear all the time. Like stuff comes in from project studios now and the low end is a mess. They were in a bad little room with tiny speakers so you have to deal with that. Some mixes come in where the guys limit the hell out of it. It's already distorted; it's louder as a mix than I think the master should be. You know, how do you deal with that? That's a problem. And

because I'm into the tweaky-esoteric stuff I got into the multiband stuff before anybody else. Not that I use it a lot, but you know, some people are like, you pull a multiband on everything. I'm like, oh my god no you don't. You pull up a multiband if there's a problem. When it's screwed-up is when I'll pull up a multiband. And they're like use all five bands. I'm like, heck no! You're putting a bunch of crossovers in the path. You know, you don't put anything in the path that you don't need. Everything you do has a downside. And if you don't need it, why pay that price for no gain? Now if it's like, EQ, now there's a problem, if what you gain is so much more than what you lose. If with all that stuff there's gonna be a little noise, there's gonna be a little phase shift, there might be a tiny bit more distortion, but it sounded like hell and it sounds great now. You've definitely gained more. But just to throw extra stuff in the path that isn't really doing anything for you… And multiband's like that. And very often I'm trying to do everything with two bands, maybe a third band. The amound of times that I've used all five bands of multiband? Like, it's been a long time, certainly more than a year. Maybe more than two years since I've had all five bands active on a multiband compressor. Anyways, so a lot of those things I got into sooner than a lot of the other guys. And when they were like, freaky tools. There weren't multiband compression plug-ins. I had to get the crazy TCM-5000 with the MD-2, their mastering package for that, which was like one of the first real mastering-specific pieces of like software editions for this hardware unit you could get anywhere. And that had multiband in it. And at the time that was awesome because there were certain things that I could do, like the hi-hat was crazy, you'd

take that mid-band and just… and some people were like, why don't you just use a de-esser? Well what that is is just like the sidechain high-frequency thing hits it, it compresses mono-band, the whole thing. Whereas the multiband actually only compresses that area. So that was a great tool. And like the mid-side. Now everybody thinks, fun, lets make it wide and stereo and you lose your imaging and it gets all phasey. You should only use this when you need it. Say you've got something where the vocal is like, really dull. And you've got guitars that are super high and piercing. What are you supposed to do? If you add high-end to the vocal, it's gonna destroy the guitar. If you tame down the guitars, it's gonna make a dull vocal even worse. Well if you're talking about a rock thing and the guitars are panned wide and the vocals are in the center… mid-side. I can add brightness to the center and take some of that high-mid pierce down on the sides. And so that's a real challenge when you have two things that the fix for one is the break for the other. And now if the guitar and vocal are both dead-center, now you're screwed. But if you've got that panning thing happening and they're in different places, now you've got a fix for that. Now people will start going, oh my god how did you do that? So those are some of the challenges that sorta can come in. So it's not like, Ok here's one thing and this is what I did. But rather these are some of the challenges that you can run into and here are some of the fixes.

K: With the understanding that every mastering project is different and that each piece of gear has its own unique character, how often do you find yourself working in the analog domain? Or do you prefer digital? What are

some of the pros and cons that you've found with each format? Do you use a hybrid system? If so, do you do your digital processing before or after the D/A?

J: Absolutely hybrid. Digital and analog both do certain things really well and both things do certain things not so well, and can be appropriate for certain sounds and not for others. I love a hybrid approach. My typical path is an analog EQ and an analog compressor, followed by a digital EQ and digital limiting.

K: So, you do digital after the analog?

J: Usually there's certain times where you want to affect a chain before it hits the compressor or something like that. I'll do a little bit of digital you know on the file itself, you know on the source material before it hits the analog chain. But more often, I'm doing digital after the analog. And one of the reasons for that also is that if something comes in at 44.1 or at 48, I'm gonna go through the chain and capture at a higher sample rate and I'd capture at 88.2. And that gives me so much. Now the digital processing, and one of the problems with digital is the aliasing and this and that happen when you hit that brickwall filter. So now I can go back EQ, and do that limiting with a higher bandwidth, and of course spreads the junk out. And then I go through a very simple synchronous sample rate converter, a Lavry, which is why I do 88.2. It's not like you can't do a 96k to 44.1 but that's better at software process offline. I also have a Weiss and iZotope

resamplers that will do when I do it that way. But in the chain, I like to do real time synchronous, so you do the 88.2. And it's the very simplest way to do it. So you move all that, it spreads out all junk, especially when you're limiting and that fuzz happens, it gets spread out. And then you go through that same sample rate converter, which is filtered when you get back down to your desired sample rate that actually contends to remove some junk that was up at… Bottom line is it's cleaner if you do your limiting and that really high frequency EQ boost… if you do that at higher sample rates... It has nothing to do with, Oh it's greater resolution, that's bullshit. We can talk about the myths of how digital audio works, but in processing that higher bandwidth actually helps and you're gonna be able to do your processing cleaner. I'm gonna be able to get a little more level with a little bit little less of the fuzz if I do it at that higher sample rate.

So absolutely, the hybrid way is the way to go. I mean, digital compressors don't really have any mojo; they don't sound good. The digital EQ's sound good. So I very seldom do any sort of digital compression. But digital limiting, on the other hand – there's no way that an analog limiter can react that fast and have the recovery time and all that stuff. Digital limiting is just better. So they both do things… I want both advantages.

K: Ok, so picking and choosing which device works best?

J: Totally.

K: How much communication is there usually between you and the artist or producer? Do you master the record as a whole and send it to them for revisions or do you send them a few tracks for review? What is your policy for revisions? Shootouts?

J: Revisions? If they're small, and they're not on every song, and you haven't gone through a million of em, yeah you do them. If they say, oh now that I hear how it sounds after mastering and they wish they'd mixed it a little differently, then they send a new mix? That's not covered because they're sending me a new mix, that's a new thing, right? That's starting from scratch and that's not anything that… it wasn't like, the normal course of my work to try to understand what the client wants and get them there. That's now, here's something new. And if they're just super tweaky people that just want to try eighteen different things, you just let them know, look there's a limit to this. So basically if it's within reason, or it's and error that I made, or it's just something that, Ok we're still working on getting to where you want to be. We'll always do, even if they're not in attendance, we'll do a couple of songs, they're download and listen, and comment. I sorta do that where they say, Ok now we can go that direction. Now we have to apply that to the rest of them, and not have to worry so much about revisions. That kind of revision, not a problem. But you know, within reason.

K: So is that usually how you work? You do a few songs, send it to them for them to approve, and then you do the rest of the songs?

J: If it's in attendance, then it's a real time process. But if they're not in attendance, I try to, if we have the time, to send them a couple things before I finish the whole record.

And communication's important. I need to know what they're interested in. So you need to know what they're expectations are. What are the records you like, what are the sounds? Sometimes they'll just sorta tell me and I'll listen to it, I'm familiar. Oftentimes they'll send along a reference track and that'll give me an idea. One thing you definitely want to know is sometimes the mix engineer will do, here is the mixes, and here is the car-test where we'll do a little bit of limiting so you can kinda tell a bit more what it's like. You want to know what those car-test mixes sound like because sometimes they are ridiculously loud, crazy, or who knows? You want to know what client expectations are, what they've heard. Like I did this acoustic thing and the guy's like, I dunno it's kinda weird. The master you gave me is quieter than the original and there's not as much bass. And I'm like, I added bass and I made it about 6dB louder, what are you talking about? And I found out, Oh it's an acoustic guitar and voice. And I'm like, yeah I didn't think it needed to be as loud as a heavy-metal record, why would I do that? But now that I found out, Oh ok. Then of course, there's demo-it is where they like the rough mix or they like the demo better. Well there's also this fake mixing engineer mastering. You go, Oh I got used to it being really loud and bassy. But I'm like, acoustic guitar doesn't have like, you know 80Hz does not like move air. It's like having a subwoofer on a guitar and a voice. You know,

so you wanted some more of that because he got used to it. So that was a problem I wish they had let me know ahead of time. Save yourself effort and know what they're thinking, know what they've heard, you know? But at the end of the day, my picture isn't on the front of the record, it's theirs. I will try to save them from themselves if they make a mistake, be a safetynet. But in the end, they're like, this is how I like it to be. And, great, I gave you the information and they said, those two things I agree with you one, this one, I still like it the other way. I don't mind. Sure. As long as you've made a rational decision now, it's when you're like, no it has to be this way because I think it's supposed to be. I'm like, well wait a minute, let me give you the information. They're like, Oh I didn't realize that if you made it really loud the snare drum's going to get quieter and the vocal will distort. I'm like, Well yeah that's how it works. Oh no, I want that to happen.

And you know what? I know how to make a record loud. And you know what, I can probably make it loud, cleaner than the other guy. Even though I'll still have some fuzz happen, and the snare drum starting to smear a little, for the same level as the next guy, my snare isn't gonna smear as much and fuzz isn't going to be as bad.

There are ways. Like I said, clipping, you think is evil, and it is but when you're going for really loud records and you don't want the snare drum to disappear, there's a certain amount of clipping that I will use in certain places in conjunction with other stuff that helps you get that little bit more. There's some of that evil you're not supposed to do that if you use a little bit in the right way and kinda sweep it under the rug it helps you. But you don't want to do that as a

matter of course, you only do it on those records where they go, Ok I heard what you're saying but I still want it to be ridiculously loud.

K: And so is that what people do when you clip the converters?

J: Yeah, clipping converters is one of the things, but there's other places. Like in the TC, you have softclip which you can use a little bit of, but not… you don't want to hit it all the time.

There are also certain spots in digital. Back in the day, people used to like the Sonic bus, this is one of the mastering engineer secrets, is where you'd pull the output bus down by like a few 10ths both for inter-sample peak reasons but also, back in the day, if there were any digital overs, the plant would reject the master. So you would prevent your overs so you would use your channel fader and go just into clipping and clip digitally. One of the things about digital clipping is, in some ways it's worse, but in some ways it's better. Again, it reacts so quickly, the minute it goes over, the digital just takes it down. So it gets into and out of that quicker than analog can, which in certain cases can be better. But in other cases, OK, well it fuzzes up too.

So there's advantages and disadvantages but using a tiny bit of that digital clip can sometimes be a better thing. But then the analog clip, there's a certain amount of, it's almost like dynamics processing, there's a little compression that happens at well. If you've got really great gear with a really ample power supply that's not gonna sag, the distortion products you get from analog clipping is

different from what you get from the digital and can be a bit more musical and a little less harsh and it'll be a little warmer. And there's other elements about how that analog clipping works that also can be a little better.

But yeah, A/D converters. When people start clipping cheap or even medium grade converters, oh my god, no. That sounds really bad. It's just awful. But if you're dealing with, like a $9,000.00 Lavry, yeah that analog stage is like, amazing. You can slip the daylights out of, well you know what I mean. You can clip that and it will still sound pretty good. But when people hear that he clips his A/D converter, I'm clipping a $9,000.00 converter. Don't clip your M-Box. And don't only clip and don't clip a ridiculous amount.

One of the tricks to getting level is to get a little bit in a lot of places. Some guy's like, Oh great, I'll turn it up 'til it's loud enough and clip the A/D converter and you'll get that horrible fuzzy crap. So typically if you need something that's like really loud, an itty bit of clip on the A/D converter, maybe a tiny bit of clip in a digital stage. I'll sometimes let my Weiss EQ sometimes just clip every once-in-a-while. Maybe a little actual compression gives you a little bit. And then sometimes two different limiters. Say you've got a Sony Oxford and then for me, the TC has an AMD-4. It's expensive but it's really good! But that limiter has a certain sound and the artifacts kinda go a certain way, whereas an Oxford also is a pretty good thing. I don't like it as much as the TC, but it's a good one. But it has its problems is a little different area. So I have the analog clip, the digital clip, the compression, and two different limiters, everything doing just a little bit. And each one of those has artifacts are in a kind of different place.

So instead of bunching up all one thing and making it obvious, you've like, spread out the evil. You've got a little bit in a lot of different places. And that's the trick to lots of level clean. That's how to do it. Don't let any one set of artifacts get too objectionable. Spread out the pain.

K: There are a lot of DAW's designed for mastering like Sequoia, Wavelab, and Pyramix (to name a few). Which DAW do you prefer and why? Is it an issue of workflow or does one program actually sound "better" than the other (i.e. does one handle fades better? does one have better dithering, sample rate conversion, etc.?)

J: Back in the day Sonic was like the clear choice, it was the only choice. It sounded better, the fade editor was there, the waveform generation, I mean, everything about it was just head and shoulders above anything out there. Slowly but surely everybody, well, not everybody else, began catching up. And then there was a time that Sonic sort of orphaned their workstation and they started getting into DVD and they tried to do other stuff. And that's when the audio department finally sort of bought out the rights to the workstation and they just became Sonic Studio instead of Sonic Solutions. When Sonic Solutions was the company, Sonic Studio was the product. And then the audio department broke off and became Sonic Studio the company, which Soundblade was the product. So and that's what I'm still using because I've still been using it from the beginning, like since it was pretty darn new. When did I start using it? I think Sonic Solutions was like started

in 1989, it was the people from Lucasfilm, that had done the, what was called Droidworks and they were building workstations of both video and audio before workstations were even thought of and basically that little section of Lucasfilm is what became Sonic Solutions and they came out with the No Noise stuff, the audio clean up stuff. It's like their first thing. And you would send it to them, they didn't have a workstation yet and then they made that available and then they made a workstation. So, early '90's that became available, you know, finally selling that. Late '80's they started doing that. Probably early '90's, like '91 or something they came out with that. By 1992 I was using the damn thing. I've been using that forever. And back then it absolutely sounded better. There's no question about it. There's so many other things. You know, when other DAWs were 16bit, this was full 24, they understood, a guy named Andy Moore is sorta the original guy who put all this stuff together and he really understands digital audio. So everything was properly done, everything was properly coded. You know, it was dithering when it was supposed to. It wasn't truncating stuff or messing stuff up. So that was back then. Now a days, yeah Sequoia, caught up when Sonic Solutions kinda got into the DVD and orphaned the workstation there and everyone kinda got pissed off and needed something. Sequoia more or less said, lets take Samplitude and copy Sonic's waveform generation and copy their fade editor and copy how they put in all the marks and PQ codes in. Ok, so I would probably, if I were not using Sonic, although geez, I'm a Mac guy. I don't really like PC so that's why Sonic's my favorite out of all of them. Pyramix is good, there's nothing wrong with that, it's a really popular one. That one you can

do, some of the classical guys use that as well in addition to that. And of course, SADiE, was probably the second one. That was the first company that really sort of gave Sonic a run for their money. I mean, they were competitive. I don't think they quite matched it but at least they were kind of in the ballpark, they were a good high-end, mastering-centric thing. So those are the big ones. And then, you know there's like Wavelab and it's Ok. Kind of the next level down. The top level stuff is Sonic, Sequoia, Pyramix, and SADiE. But yeah, Sonic, I use just because I've been used to it and I still it does some things a little bit better in the mastering world than other stuff. Now, if you're using a lot of plug-ins, it's still not that stable with plug-ins and that's what everyone complains about. They say, oh yeah, it's buggy bla bla bla. Yeah, when you use plug-ins, it can do some funky weird things. In a higher-end mastering house when you're using pretty much outboard all the time. You know, I barely use plug-ins inside Sonic and when I do, it's one little plug-in. It's like oh, I have this and this and only song number four you do this thing to it after it's in the workstation. It can handle that without crashing. So that's the thing. People have sort of complained, and it's improved, but it's a small group of guys. It's not like, hey they have a huge budget, but I mean they're a small group of people that said they don't want to let Sonic die, it's such a great product. But it's hard for them to compete with people with resources and huge teams of programmers. If you're not using plug-ins, I think it's still the one to beat. If you're on PC, I'd say get Sequoia. Other than that, if you're mixing a lot of tracks of classical music then, sure, get Pyramix. There's nothing wrong with that. But I didn't like that as much. Definitely Sonic and

Sequoia are the two that I like. But again, Sonic if you're not using lots of lots of plug-ins.

K: Well, it's nice if you're used to Apple.

J: Like I said, if you don't use a lot of those plug-ins, and I mean, honestly, it keeps getting a little more stable, a little more stable, it keeps getting better, not worse. And they're doing not quite the object oriented stuff that like Sequoia's doing but it's not bad. They're doing it so that you can drag a plug-in onto just that song. You've got all those little things where drag it and you can slide how long you want it to last. So I mean, it's not bad. It also depends on what plug-in you're using. They don't have all the resources, they're not gonna test two hundred plug-ins to make sure that they're gonna work well. You know, get a plug-in that you know works well with it and use outboard. Just get gear. I think of the people who go, oh the plug-in's not perfect, how am I gonna master? How else are you going to master?

K: The last question may be a little selfish but do you have any advice for somebody who wants to start a career in mastering?

J: I mean, a lot of it is the same old stuff that, I don't know, I say the same old stuff a million times and sometimes people say, no I haven't heard that. So I mean, lets start in general, the 30,000 foot view, audio in general. There aren't a

lot of jobs out there that are like, here's the 9:00 to 5:00 job with the benefits and here's the ladder that you climb and work you way up… There's not a lot of that going on. Even when you get a job at that place, you need to be an entrepreneur, you need to go out there and sort of shake the tree and get your own clients, or the ones that come in, make them love you. To be able to sort of create your own job if you can't get one, even if you're working for someone, within that, you want to become the guy everybody wants to go to. That's how to work your way up in that company and then be able to sort of go out on your own afterward if you want to or start a partnership with someone. Or if you stay at that company but now you've gone out and you're bringing in clients and you're bringing in work and you're the one that they're saying, hey this guy here, he's the go-getter, he's going to benefit our business. I mean, that's really, like anything you don't go in going, Hi I want a job from you, how can you benefit me? You need to go in and be like, how can I benefit you? You know, that's the trick. There's plenty of people showing up with their hand out but when you show up with something in your hand, they go, this is different. You bring things to the table? Ok, great. And sometimes, people who are really good hustlers end up doing better than people who are really good technically. A really good hustler who is good enough often can go a lot further than the super talented guy who's sort of the social misfit. So that's that part. The entrepreneurial thing, definitely figure out how to benefit others, make them like, you gotta have this guy around. Yeah networking and social skills, the ability to sort of fit in to any situation. That's one of the things that some of my interns have said, which is interesting because I hadn't really

thought of it in that way until several of them in a row told me. They said as much as we learned about technical stuff, one of the biggest things we've learned from you is about dealing with people. They would see a lot of client interactions. And it's just that anyone who comes in, you know, making them comfortable, making sure that whatever their vibe is, and you know, the heavy-metal guys come in and you don't want to be wearing an oxford and acting like a nun. Whereas the Christian clients come in, you don't want to be sitting there, you know, smoking a bong and drinking Jack Daniels out of the bottle! This is not something you do on a session. It's about what make the client comfortable. You always want to be professional and business-like to a certain degree but there's certain people that want to joke around, there's certain people who want to be a little bit more loose. There's other people who are like, you're the mad scientist, I want you to act that way. So you have to be able to key into, and it's not like you have to change who you are. No, no, no, there's parts of our personality that you know, I've got in me the rocker, the nerd, the scholar, whatever. I've got all these pieces and it's more like which one of these am I gonna let out more today and which ones am I gonna subdue. So fitting into people. Making them comfortable, making them think it's their idea so to speak. You know what I mean? Always making them feel empowered but that the same time, you're kind of steering the ship.

So, the other part of making it in this business is just, it's a war of attrition. You know, last man standing. There's gonna be a hundred people who go, I wanna do this. Then fifty of them right off the bat are like, oh forget it. This takes work. I didn't know about that. Now the other half that are left, some of them are

gonna be like, oh I really want to do this but I have this full-time job and I have a wife and a kid already. Wow, to break into the business when you already have all of those responsibilities, wow, that's tough. It takes a couple of years when you just eat, sleep, drink this stuff. If you want to make it, don't be like, Oh I'll do it on my off hours. You don't have any off hours. If you need to go into that situation, take a year and work and save up money at home so that you have a nest-egg there, then go spend a couple of years wherever you're gonna go and not have to worry about the, oh well I have to have a job. Things are gonna come up last minute and they're gonna be like, hey can you come over and do this thing? Yes! Sure, I can. You know, you just did a session for eight hours and they're like, hey, there's a thing going on at night, can you come over and do that? Might as well! And trust me, once you're in your 40's with a wife and kid, you're not gonna get it anyway. So that's the thing, be around all the time. Work with people, hang out for every possible session, be at the studio always. Do what ever that takes. Then the people who didn't save up the money or who didn't spend the time or after a year or a year and a half start to go, oh this sucks. I'm not getting paid. No, give yourself at least two years to be used and abused. It's really that simple. And over that two years, the people are gonna widdle down to like, ten if you're lucky. Maybe not even that many. And you want to be one of those ten guys because now you've got a shot. Now people know you, you've done some stuff. You were there when whoever didn't show up and they were like, hey he's gonna be three hours late, can you record these vocals?

Within the mastering context, hey, you're sitting here and observing and they're like, oh you know what, our production guy's not here. Wouldn't it be nice if we had production engineers? But the big guys like Sterling still have production engineers where all of a sudden, they're gonna be like, hey I EQ'd all of this stuff, alright can you edit and do the PQ and do the delivery on the Beyoncé record? And all of a sudden, you've done a couple of those and then it's like, oh hey, at night, whatever engineer you're attached to, when they go home at night and that studio is open, they're gonna go, hey, if you want to go out, shake the tree, and bring your own clients in, you can work that room for half-price. And in a big room like that, half-price is still expensive for your clients, but your clients are gonna be like, I can get mastering done at Sterling and for half-price. The facility wants to do that because they want you to build a clientele and if it really works out well, then bam, we'll give you the night time hours for real in the room. We'll start feeding you work. It's in their interest to have that career. That's sort of the, there it is. I mean it's pretty straight-forward. It's not rocket science. There's a little luck and ok, maybe some rocket science. Go meet bands! Go to clubs, do that. And some of your friends that you're going to school with, some of them are gonna be making records as musicians, some are gonna be making records as engineers and, of course, some are gonna be flipping burgers. You can probably tell the talented guys who are probably gonna do something. Keep in touch with those guys.

And the other thing is to really hone your craft. Work on everything, work on different styles. Once you get it done, listen to it one every system under the

earth, you know what I mean? It's about learning what translates and what doesn't. Keep notes, absolutely take recall notes of your settings. There will be changes that the clients will order but once that's like two years from now, you'll go, I wish I had done that better. Bot I've learned a lot since then about what I do and I can go back and just for practice, just look at what you did and what you'd do differently now. Why did I boost 600hz? Everything sounds so boxy and awful.

Part 2. Mastering Subjective Language

Punch: Punchy stuff to me is usually midbass stuff. I want it to hit you in the chest. If you want punchy kick drum, you're not boosting the super bassy 60hz stuff, you're boosting 90, 100, 110, maybe up to 140. Even on a snare drum, a punchy snare drum could be up to 200, it's a woody thing. But once you go over 200, it's not punch anymore.

Warm: Warmth; that can come from frequency ranges or distortion products. If you're talking frequency ranges, that's what happens sort of above the punch. Sometimes it's 250, it could be as high as 800 but typically, it's gonna be 300 or 400. That's kind of the warm area where you want warmth and body in the guitars. The guitar tracks sometimes are these weird scoopy things where it's all boomy and all bright and you're like, where's the 400 man? It's like so important. There's so much to the warmth and body in an electric guitar in that range. You

know, the other things that people talk about processes being warm is it's really tubey, it's really a distortion product. I can't say it's more second or third order harmonic distortion, you know, because the distortion also gives presence depending on how much of it and what it is. You can get warmth or presence, of course presence is higher up. Well, what about tubes? Well, you know, what about transformers? Transformers are hipper than tubes. You know, for like cool warm bizarre distortion products, things like that, and it gives you a little bit of compression in a weird way. You know, it slows down those transients a touch. There are people who listen to the old Neumann tube mics and then they're like, oh the new Neumann reissues just aren't as good because they have tubes but no transformers. You know, why does a Neve module sound so warm and wonderful and great? Transformers! Why do I love a high-watt guitar amp? Transformers are a really really cool sound.

Clear: Clarity! I mean that can mean so many things. I mean, if you want clarity, so many things go into that. In a mix, you have stuff panned properly, have everything have its own frequency range. Sometimes having a full range or frequencies is not the right thing to do. The big heavy ridiculous guitar thing, a lot of times you take out a lot of that bottom and then it's with the bass.

The musical arrangement absolutely contributes to clarity. And when the bass and the guitar play together, it sounds like this big huge heavy thick guitar sound where if you listen to the guitar sound, you go, actually that's not nearly as heavy as I thought. But in context with everything else, it's amazing. So clarity a

lot of times has to do with that where you have this mix where you feel like it's well represented from the lowest lows to the highest highs but that doesn't mean every situation has the lowest lows to the highest highs. So even and balanced EQ over the whole spectrum, each instrument playing its part, everything in its own space. Even the pan position is hugely important. A lot of times, if you pan something out a little bit wider, you don't have to touch it. You're like, oh, that works!

So from a frequency standpoint, what's gonna give you clarity? It's not so much what you're adding, it's what you're taking away if there's too much of it. If you've got too much of the sort of warmth, boxiness, or if anywhere from 200 to 800 (800 might be a little high on that). But if you've got too much of that thing that muddies it up or boxes it up, or something that puts a blanket over it, it's that. You can't have too much of that.

Harsh: Harsh; anything that if you turn it up, it's gonna make your ears go "eeek". And again, typically it's around 2.5k. That's the one that people sort of think of first. Yeah, that one but 4k can be piercingly awful. That's one of those spots that if you look at Fletcher-Munsen where you're gonna be really sensitive. And that's also the spot that if you look at audiologist data who look at where you loose hearing, you loose the extreme top, that's just natural with age. But a lot of trauma-induced loss happens around 4k. And certainly 1k is plenty painful so you can have harsh anywhere from 1 to 4 and probably a little to either side of that, maybe.

Groove: Mmhmm, groove, my favorite stuff. Here's one thing that people don't realize is that you can actually improve the groove in mastering. People are just like, what are you kidding? They'd be like, it wasn't grooving when we sent it but it's grooving when we got it back. How on earth did you do that, it's just mastering. And there are a couple ways. Of course, one of the obvious ways to do it is to feature the things that contribute to the groove. Whether it's the kick drum was muddy or not being heard, or was it this cool hi-hat part that wasn't there or was it this tambourine or shaker thing that's doing 16ths or was it a guitar riff? Bring out those elements. What are those elements that contribute to this groove and go in and find those and sort of enhance those and sort of highlight those.

The second way you can highlight the groove in mastering is with the compressor. One of the big tricks to buss compression, separate to how I want to control the transients and dynamic range, is to get those attack and release times, and you're still typically going with slower attacks and faster releases, things like that. But even within those parameters, of somewhat slower here, somewhat faster here, you want to get them to bounce with the music. If you can use the compressor to kind of bounce with the music, you can actually create this little hip-hop, create that little bounce and have the music groove with the action of the compressor. And that's a cool trick; it's a tough one to master (no pun intended). And it doesn't work all the time, you can't use it for everything but every once in a while, you can go out and it's like, yeah it's bouncing this cool way.

K: Is that something you can do with parallel compression, or is it just straight buss compression?

J: To a certain extent you can do it with parallel compression but of course, at that point it's bouncing more from the bottom. So instead of just the bottom bouncing, you want the whole thing to bounce. But to a certain extent, yeah, you can do a little bit of that with parallel especially if you're in the situation where you're like, I can't put any more compression on this or it's gonna sound like hell. I'm gonna bounce it with parallel. But it's easier to do it not in parallel if there's not a lot of compression on it already.

Frequency Ranges:

Lows: Lows are everything under 100 and probably everything under 75. I absolutely add midbass into that. It's really an important region. There are times where like, bass guitar is doing this ridiculous awful stuff where it's just taking over, making everything muddy and there's other times when somebody would be like, I want that. Bring back that punch, I want to feel that kick drum, I want to feel that low end and you keep cutting 40 or 50 and it's horrible. And I'm like add a little 100, 110 to it. That's really midbass. Again, it's useful for me to have those two things separate because they function differently. And think about it too, remember those frequency numbers, as you go up, from 20 to 80, you have two octaves, that's a lot of area. And then from like, 300 to 400, that's not even

half an octave. People are like, bass is 20 to 160 and I'm like that's so many octaves. Keep it 20 to 80 then have 80 to 160 be your midbass.

Low Mids: It's almost hard, I think of that lower-midrange, and this is probably the way, you know, one of the teachers I had at Berklee kind of thought of lower-midrange as being a more limited thing. A lot of people are like, oh no, the lower midrange is everything from 100 to 1k. I'm like, by the time I get to 900, I just think that's midrange. I don't think that's lower, you're in the mids! It's the "mud" into the "box". It's muddies and boxies, that's lower midrange.

Mids: So once you get to 800, that's midrange. 600? Ok, I'll buy that as lower midrange, that's still a nasty nasaly-boxy thing. But, I dunno, after that, you're just in the midrange. Maybe even 500. Traditionally, I thought of 250 to 500, that's the lower midrange octave. And then 500 to 2k, 2.5k, is like the regular midrange and then like, 3, 4, 5, 6k, that's upper mid.

High-Mids/Highs: 6 is really starting to be the lower part of treble, you know what I mean? That's kind of trebley, but that could go either way. It depends what it's attached to. It's either the top of the upper midrange or the bottom of the treble. I tend to think of 6k as still part of the midrange. You're still getting presence, and like some snare snap or whatever. 8k, you're definitely into treble at that point. 7.5, you're definitely into treble. 7, yeah, I'm gonna say your high end comes at 7. I'm gonna keep 6's in the upper midrange. And if you're thinking of

the treble, you can think of octaves, 8k to 16k, yeah that's treble. Above 16k, you can sort of thing of as the air kind of treble. But think of that, when you're thinking of frequency ranges like that, think of some octaves. Midrange is the biggest one, that's where we're most sensitive. It covers a lot of ground. But it's the middle, think of the biggest part of the bell curve.

II. Matt Azevedo

Matt Azevedo is a graduate of the Sound Recording Technology program at the University of Massachusetts Lowell. Since his graduation, Matt has spent ten years as a mastering engineer at M-Works mastering in Cambridge, MA. Matt then decided to pursue other interests and has since been working as an acoustical consultant and designer. His multifaceted skill set includes electrician, acoustician, musician, and audio engineer.

This interview was also conducted over two sessions. The first was at the Sweet Touch Café in Cambridge, MA on Wednesday November 18th, 2015. The second was a response to the questions via email.

K: Could you tell me a bit about how you first got involved in mastering? Did you get into audio first as a musician or were you more interested in the technical side of audio? Did you go to school for audio? Did you intern or have an apprenticeship?

M: I was at UML, and fellow UML alum Mark Donahue. I like Mark Donahue. Anyway, he came and did some stuff with us. It was pretty cool.

So I was a senior John Shirley's first year here. And there was this "intro to DAW's" class. Back then we were spending most of our time on 2-inch. So this class looked at Protools and Sonic Solutions. And part of the reason Shirley got hired was because all the little old dudes were there at the time and none of them really knew this DAW stuff. And Shirley's big claim to fame was that he was a hot-shit Protools guy. And the other DAW, the Avid one… ya know, it was weird. It was Protools-y enough for him to be OK with it. And then we got to Sonic. And I had paid a lot of attention when Mark came and demo'ed Sonic and I thought it was the coolest thing. And I'd gone in and like spend some time with Sonic. Sonic does not work like Protools at all. It's a whole different logic to how it works and Shirley just did not know how to use it at that point. So that class ended up being, he'd be like, Ok so what you want to do is… and I'd say, Oh you click on this thing. Like, Oh it's this menu. And by the end of the class, I was sitting in front of the class teaching everyone how to use Sonic Solutions. And I still love that platform. The current version sucks in terrible ways. It breaks my heart because I want to love it. Kinda like if your husband becomes an alcoholic. Like you remember the good times but he's a violent drunk now and it's just really hard to keep the family together. Soundblade, the new version, is a violent drunk. But back in 1998, me and Sonic Solutions 5.4 were BFF's.

So I love that kind of editing. I was super super technically minded. I was really into the guts of how stuff worked. And towards the end of my tenure, I tried

to find a job, tried to find an internship and bounced around a few things. Nothing really panned out and Dr. Moylan called me and said, Matthew, have you found an internship yet? I was like, well, yeah, I'm working on it. I got a few leads hopefully it'll work out. And the manager at M-Works at the time was a woman named Gene Savedi who's awesome. But they needed an intern and she called her alma mater and said, hey we need an intern and Dr. Moylan said, I think this will be a great place for you. I'm putting this fax that I got up on the internship board in one week. What you do in the next week is your business. So I ended up at M-Works. So I had an interview and bullshitted at the right level so I ended up with an internship. And I made a point of hanging out late after sessions and futzin' around with files on the drive and tried to figure out how stuff worked… Smuggled a couple… I had a friend who had recordings from a blues show from WJUL back when it was not UML. We did a little comp CD of live performances. He just gave me the stuff and I stayed out late one night. And I was doing a ton of editing and QC work. And initially there was a band called Dialectric and they wanted their record mastered and didn't have a lot of money. So one of them was roommates with my very dear friend Andy. Andy said, Oh my friend Matt is interning at a mastering studio maybe he could do it for cheap. So the first album I ended up doing was the Dialectric CD. And I said to Jonathan, hey I have some friends who want to work on a record, what should I charge and what would I get? So we worked out a deal. So before I was on the payroll there I was already bringing in… I was already a profit source. I was already an income stream, which I don't think a lot of interns try to get themselves into that position. They

just kind of hang out and wait for something to happen. So I made a point of going out and finding… My wife's a hairstylist. To become a hairstylist, you have to be willing to go out and assault people on the street. Like, she would run Craigslist ads and she would just get people and do haircuts. So I was kinda doing the same thing. I was active in the music scene and I was talking to every band I knew. I was like, oh it's like the nicest studio in town and I can totally master your record. And I screwed up a little bit but not so bad that anybody got angry. Not so bad that anybody noticed, but in my heart, I knew. So when my internship was up, I had become a profit source for the studio. And that was just when they were breaking ground on the current facility. We used to be in the basement. So we were going from a one studio, one, literally, edit-closet facility with two staff engineers and an intern and a manager to the current space with three rooms. So now we're gonna have all this space and we need to be able to operate all these spaces at once; they needed a third guy. I ended up doing a lot of work, like I wired the network. Every Ethernet wire in that place, I terminated. I shouted at the electrician a couple of times, which is kind of a weird thing to be doing, but I did. It's all about being scary in the right way. So it was Colin Decker working in the back room, Jonathan in the big room, and I was in the little room doing edits and QC's. I would come in at like 2:00 and run Jonathan's masters, which is a much bigger pain-in-the-ass those days because he wouldn't print any EQ. He would save everything as presets on the desk. So I'd have a move list and the disc would burn in real time. So it would be edited in Sonic, and it'd be running a loop in real time through the desk then back in to the CD burner. So I had a move sheet. The

song would fade out and I'd be like, "whoop" and I'd change all the settings on everything in real the two-second gap between the songs. And if you screwed up on the last track, you started over. I'm kinda glad that's not actually a practical way to make records anymore. Because in the olden days, like there was no DAW so you had to splice everything together on a reel and you'd play the reel through the desk and you'd have to do all the moves in real time between the tape deck and the lathe. And that style of working persisted longer than it should have. And really, the real way to do it was to have two of everything. You'd have two Sontecs, two compressors, and you'd have two limiters. And your Neve transfer console, there was a button for path A or path B. Put the first song on path A and the second on path B and you'd just push the button. And you'd set up track three on path A then track 4 on path B. We didn't have the switcher. It sucked. You want to learn humility, blow a master in front of the client 3 or 4 times. Boy that sucked, but I did that.

After Jonathan's sessions, I would have clients at night. So I'd come in at 2:00, do Jonathan's stuff. His clients would be out at 5:30 and mine would come in at 6:00 and I'd work until the album was finished. And then after a couple of years, Colin got sick of being broke, and ended up going to grad school. And last time I checked, he was Vice President of online video for Yahoo, making profoundly more money than anyone I know in audio. So, you know, grad school… Master for ten years then go to grad school.

So I took over Colin's room when he left. So then it was me and Jonathan from 2002 to 2010. At some point, Jonathan looked at how much work we were

doing and did finances and decided to rent out my room to a sub-tenant. So I was doing editing, post production, QC'ing, whatever in the back room. And then Jonathan and I would share the big room. And that's how we did things up until 2010 when we were really hurting after the 2008 crash. So there was a real perfect storm of… when I started, if you wanted to burn CD masters, you needed a $20,000 Sonic Solutions computer rig and a $5,000 CD burner. So no one was just gonna buy that. For a long time, consumer CD burners could not write the full set of PQ data. So you could not burn a replication master on a consumer CD burner. So it was by the mid-2000's when you could buy a non-scuzzy CD burner that could burn a valid master. And then it wasn't too long after that, you could get a piece of software that would at least let you open up a list of files and gather them and put in PQ data. So by 2010, everybody was hurt and all the mix studios that used to feed me work had become by enemies because they were all saying, You can go to M-Works and they'd charge you $1,000 or for $200, I got the brand new Waves L2 plug-in and we can put your mixes through that and make em loud and I'll burn you a CD. That'll save you 800 bucks. So work got tight. What it basically came down to was I went from doing 200-plus albums a year in 2006 to 150/120 a year in 2010. And Jonathan had really moved off and began doing more work at Berklee. His wife at the time was a touring concert harpist. They shot a PBS special and whenever she was touring, she would have him on her contract as her manager. So he was on the road a lot with her. So he wasn't doing much. When we were both cranking, we were doing like 250 albums a year. We were doing an album a day, 5 days a week, all year and not taking vacations.

And that was the glory days of 2006. So things got really bad and I evaluated my options and thought what could I do that wouldn't jettison 12 years of audio expertise? So I was hanging out with my friend Andy and we were thinking, Oh what am I gonna do? Andy said, well there's a guy who plays cello in my band and he's an acoustical consultant and he does all this sound stuff. I bet you could get into that; you're smart, you like sound. So I met up with him and talked about acoustical consulting and he said, Yeah you'd probably be pretty good at this but you're gonna have a hard time without any credentials.

So right in 2008 when things started getting bad, I started formulating escape plans and they were all technically oriented. But I still master a few albums a month. I still do some tracking. But, you know, I buy name brand juice now. So that's how I got into it, and that's how I got out of it.

K: As your career advanced, how did your philosophy on mastering evolve? Is there anything you know now that you wish you knew when you first started?

M: When I first started out, the logic, like if you're working on something and it's not working is to throw on another piece of gear. It's like, oh this isn't really working, I'll put on a second EQ or I put on another compressor. I was doing more stuff and the worst sounding records I've done in my career were within two weeks of getting a new piece of gear in the studio. Because there'd be a new thing

and I was really excited about whatever the good-sounding thing it did was and the excitement was masking all the not good-sounding things it has done. There'd be some new toy and I'd overuse it for a week and a half and I'd realize, Nah this just sounds like crap, what am I doing? So there's a bunch of like, new toy showed up and new toy enthusiasm resulted in less than my best work. At this point I don't do anything. People get concerned. They ask me what I did on the track and I tell them. They say, that's all you did? When I'm working in the big room at M-Works, 95% of the tracks I work on are the Sontec, the Requisite, and whatever limiter I'm into that year in the box. That's it. If you're using more than four bands of EQ, you're probably doing the wrong things. If you can see the needles moving on the compressor, you're right on the edge of doing too much. If you can tell the limiter's in, you're probably doing too much. Even when I'm working in the box, there's the EQ I like. Bi-quad filters are bi-quad filters. There's the compressor plug-in that adds a nice distortion, and there's a compressor plug-in that doesn't have a nice distortion. I will sometimes use one of those two and the same limiter I would use at M-Works, which for a long time was the Xenon limiter, the PSP one. I think since Ozone 6, the Ozone is my favorite. 5 wasn't quite there, but 6 and 7, it's really good. So every track I do is the same. Now, that's if the mix is right; that's mastering. Mastering, I'm doing two or three bands of EQ, and a bit of compression. The reason I use the Requisite on every track at M-Works is because it's got big fat transformers, legitly run on proper 300 volt rail tubes. And there's this little bit of euphonic saturation that you get just running through it. And I put things through it because I like a little

bit of euphonic saturation. That feedback control on the newest version, you can even intensify that. I'm pretty much using it for color and I set the threshold usually with the filter turned way up so it's not reacting to bass stuff at all, and the threshold so that if you kinda look at it, you can kinda tell that the needles are alive. And it's just enough to lop off the really peaky-peaks so you save yourself, so your digital limiter's working a little less hard.

Now, there's also fixing mixes. If I'm fixing mixes, the gloves come off. There are times I'm fixing mixes and I'm doing Mid-Side, multiband, parallel compression, or other bizarre terrible things that I'd rather not talk about. But when something's broken, you just do whatever you can to fix it. And I've had some really broken projects come in.

I think the first assumption should be, somebody approved these mixes. Probably many people have approved these mixes before they move to mastering. If the mix engineer liked em, and the producer liked em, and the artist liked em, then if you give them back something that doesn't sound like their mixes, you're disrespecting the fact that all these people said that they like this. So starting from there, sometimes I don't do anything. Some people talk about mastering as framing a picture. I think even that is too dramatic. I have a friend who owns a gallery and I've watched her set up shows. And there's a real beautiful art at what height to hang a piece of art and how directly to light the piece of art. I think that's really what good mastering is. Someone has handed you a painting and you are displaying it. You're saying, Ok there's a strong compositional line so I want that to be at eye-level. And it's got this kind of color so I want like a more

yellowish incandescent light or like a more blue full bandwidth light and I want it to be this bright so the colors pop. It's still the same piece, like the piece has not been changed at all. You haven't turned it sideways and you haven't gone in with a paintbrush and say, Oh this guy needs a mustache. You're just presenting it for the viewer to experience it. I think good mastering is deciding how high to hang it on the wall and how to light it. Everything else is restoration. I have clients that say, I did this in our bedroom and it was really frustrating. We didn't get the sounds we wanted. And it's like, cool, we will change it. But the first question I ask in every session is, how do you feel about the mixes? And the answer's usually like, yeah we're excited about them. Cool, we will maintain your sound. We will give you back your mixes but better.

One group I worked with, a singer-songwriter group came in and I started listening. Everything was fine until the chorus when the kick drum came in, it was like a techno track. They were on the couch, just slackjawed. I was like, is this how you wanted it? Turns out they mixed it on mid-sized midfield monitors with, like and 8-inch driver. They just couldn't reproduce the fundamental so what they were hearing was the first harmonic. So there was a lot of cutting in the bass on this record.

K: So I guess that can segue into the next question. Are there any projects that stick out to you as having a particularly challenging aspect or technical difficulty that you had to overcome? What was the nature of the challenge and how did you react to it?

M: Most mixes just sucked, you know? Bad mixes by inexperienced engineers… I think the more interesting things were, one I thought was really cool fix was an album for my friend, Mike Bulluck, who's a wonderful experimental, traditionally trained upright bass player, who does really wonderful fascinating sound art experimental stuff. And he had gotten, you know, a label was going to put out his record on the vinyl LP. And he had made all his recordings, you know, heavily processed field recordings, heavily processed instrumental things and there was no center to the bass. Like, the information from the left and right were totally uncorrelated and if you hit mono, the bottom will disappear. Due to the physics on an LP, mono information is cut laterally and stereo information is cut vertically. Basically, the cutter head is an M/S matrix. So a mono album is just left/right and the way they made mono and stereo compatible, so you could play a stereo record with a mono needle, the left/right is still a mono signal and the up/down is the stereo information. You take the one magnet that reads the vertical and one that reads the horizontal and you do vertical plus horizontal is your left channel and horizontal minus vertical is your right channel. You know, just straight up M/S matrix. Which means that if you have too loud particularly low-frequency information on your record, the vertical travel of the cutter head gets so big that it either lifts out of the record, the groove disappears, or it punches down through the bottom of the record. But if you have too much stereo information below, say, 200hz you get a record that is physically unplayable.

So Mike had this really cool stuff that sounded great on headphones and was physically impossible to cut to a record. And the normal thing you do when there's too much, when the bass gets too wide for a record cut is to high-pass the side channels. Just put a high-pass filter at 200-ish Hz and you get rid of that out-of-phase bass information. Even for a lot of mixes where you listen and you're like, man that kick drum is muddy… there's just no focus at all in the bass. If you kill the low frequency in the side channel and have a clean mono bass, it does a lot to clean up the low end in general (that's a freebie). But on Mike's record, if you did that, because all the bass was in the side channel, it just, all the bass went away. So what I ended up doing was stacking, high-passing, I think it was the right channel… one of the channels, I cut the bass out… a steep high-pass filter, a third-order high pass filter at like 80hz. And then I took the Ozone multiband stereo width control, cut it down to just two bands, put the crossover on that to where I put my high-pass filter and when you go into the preferences you can change the shape of the filter. So I matched that crossover filter with the crossover of the high-pass filter I used, put it at the same frequency, and then mono-ed the low end. So I basically took all the right channel bass away and then panned the left-channel bass to center and then added 3dB of gain to make up for the loss and it sounded perfect. And like it didn't even sound that different. On speakers it sounded better because listening to out-of-phase information in the air is kinda weird. So it made it just sound better in general and it made it physically cutable. One of the things I like about that particular story is it really ties into the original purpose of mastering, which was to get things ready for pressing. And because I

understood the physical limitations of the release media, I could identify what was a catastrophic problem before it went to press. And it was great. The album came put and everyone likes it.

K: Ok, next question. Do you prefer analog, digital, or do you run a hybrid system?

M: I like tools of both flavors. I have a broad preference for hardware because I can focus on what I'm listening to and turn a knob with hardware. Where with software, I'm really forced to look at a screen while I'm working and I find that critical listening works better when you minimize the visual stimulation you're getting. So I like to have pretty minimal metering in the room. Like, just kinda something that gives an idea of where the levels are at. I like mechanical VU meters for that reason because there's kind of, you know, they don't tell you enough to distract you. Like, yeah you know, it's kinda bouncing around that looks pretty good. So that said, you know how I mentioned that all bi-quad filters sound the same. But they sound really clean. You can very carefully define the phase-response, the frequency-response, you get a ton of control. Really, when you deal with any kind of a DSP process that you're handing to a user, the question isn't what can you do, it's what controls do you actually want to expose to the user?

So digital's great because especially if you get nerdy enough to get into like programming language like Max or MatLab or something, you can do pretty

much anything you want. Something like iZotope RX is totally impossible because you're using FFT algorithms that require time to be able to go backward and forward. So you just can't do that kind of stuff in the analog domain. On the other hand… You know Jeremy Houle? Once upon a time he was at an AES thing and he bumped into Mr. Rupert Neve. And he kinda fanboyed out a bit. He was like, Oh Mr. Neve I loved your stuff. So tell me, what's the secret to Neve's sound? And Mr. Neve intoned, "Distortion". And that's really it. You know like, the magical wonderful thing the Requisite (L2M MKIII compressor) does is a non-linear, non-constant level and frequency-dependent distortion. Big transformers are completely linear except for just a little bit they're not. Tube triodes operated in the linear, middle part of their, you know biased clean at reasonable levels are super linear. But as the peaks just get the tiniest bit out of the linear range, at the very tippy-tops of the wave, there's just this little bit of extra stuff. If you ever get into DSP there's a thing in the world that is called the LTI system. A linear time invariant system. It means that there is a direct relationship between the input and the output and the relationship is always the same. So if you had an amplifier that's set up for 6dB of gain, and it's flat from DC to way higher than we care about. That's a linear time invariant system; you put in a signal and it comes out 6dB hotter until the input signal hits the power rail of the amplifier and then it can't give you 6dB of gain so you get 5dB of gain and a flat-topped wave that gives you all this distortion. So now, linear just screwed the pooch and those distortion characteristics are going to be dependent on the slew rate and the temperature and all kinds of other fiddley non-changing-over-

time aspects of the amplifier. Anything that is linear and time-invariant, you can do in DSP- easy. Rooms, like acoustic systems are, unless you're getting to levels over 170dB where air clips, and you'd be dead so it doesn't matter, but unless you're modeling bomb blasts, sound in a room is a linear time invariant system. So you can do all kinds of great things with acoustical modeling. You can parameterize the room and you can do math and see how the room behaves. Or you can go out with test equipment and play test signals and see how it behaves. Assuming you make adjustments for humidity and temperature you can come back a year later and get the same result.

There are lots of distortions that are musically useful. Like, what comes off a violin string is a near-perfect sawtooth wave. But what comes off the body of the violin is totally not a sawtooth wave. So there's this distortion happening from the input signal of the strings to the output acoustic signal of the body that's profoundly not linear, and it's nice. You know, turning up a guitar amp a little too loud sounds nice. The magic of most distortion is its harmonic. So you get distortion at overtones of the input signal and increased harmonic complexity is often a nice thing. So I think analog gear, I mean there's digital models of saturation and they're OK. It's a thing, whatever. I find digital warmth, which in this case is really just distortion. But digital pleasant distortion processes tend to go from too little to be worth it to too much to be worth it in a very small range. Like, the exciter thing in Ozone by the time that slider's on like, 2 out of 10, it's too much and 1 doesn't really do anything. So you've got this teeny little range of well, 1.1 – 1.3… and that's it. That's the only usable range on the thing. And it's

not dynamic. So the way tubes and transformers react to the signal, you know it's heavily influenced by the recent signal that has gone through it. You know, like if you hit a piece of tube gear really hard with a big transient you suck enough current out of the filter cap that it doesn't have enough to properly run the circuit afterwards, you get this kind of sag in the response after a big hit. I wouldn't do that as a mastering thing but I do it with guitars all the time.

So I think there's this nice free stuff, there's this enjoyable distortion that you get out of analog gear that's still not well emulated. And like I can sit in the studio at M-Works and operate the Sontec without looking at it at all. There's a lot of years of muscle memory. I know where all the knobs are and there's this great tactile clicky feedback when you move them. So I can just sit there with my eyes closed and EQ a record. And I do! I kinda of just sit there and just stare off into the distance and the knobs are different sizes so you can tell what you're doing and I'll just EQ a track. And on the Requisite, I can just take that thing and having that tactile feedback that the control is doing without having to look at the control, I find super valuable in assuring I'm doing my best listening. And I just love not having visual feedback on things. Because everyone's done the thing where you sit there for 5 minutes dialing in a piece of gear and then you realize it's on bypass. The amount that confirmation bias skews our hearing is hilarious. And if you've got a picture sitting in front of you and you think I'm doing this thing, it's really easy to believe that picture. So I like hardware for tactile reasons, for practical reasons. There's a lot of things that I do digitally that I would never want to do otherwise. Brickwall limiting for instance, being able to do look-ahead

limiting, I mean you can't do that analog. So I can't think of a time I do serious CD-type limiting with an analog limiter, it's just stupid. But there's a lot of times when there's pretty distortion I want. I want my EQ changes to not just emphasize that frequency, I want it to create a ripple of the harmonics of that frequency. And digital doesn't do that well.

So my perfect mastering, I'm actually building a mastering studio right now because I just bought a house. But I'm building a Sontec clone, and I'm proably just suck it up and buy a Requisite. I'd like to go and build the all-tube, you know, James Baxandall Baxandall schematic from the '30's. Build a Bax EQ with maybe a sweepable mid in there too. Like the high and low control of a Bax and maybe the mid of a Pultec. I want to stick that in a box. And it's going to be all tubes and transformers because I want to open up the top end and distort it at the same time. And I've got a potentially brilliant alternate way of handling compression that I'll probably initially develop as a plug-in and then potentially do it as a hardware version. But that's it. That's all the hardware I want for my studio. Really, 95% of what I do is the Sontec and the Requisite. If I have just those two and a bunch of plug-ins for the weird stuff, that's all I need. I could buy more crap but I'm not interested. I want a good EQ and a compressor that does what my brain thinks it should do, which is hard to come by. That's all I need. And that's it.

K: How much communication is there usually between you and the artist or producer? Do you master the record as a whole and send it to them for revisions or do you send them a few tracks for review? What is your policy for revisions? Shootouts?

M: Usually a lot, especially in the situation of an attended session. I prefer attended sessions so that there can be a running dialogue throughout the process. If it is an unattended session, usually I will master the entire album and post it for comments and feedback. My official policy on revisions is "revisions within reason are included in the base price". 19/20 times that covers everything, but you need to leave yourself an "out" for that 1/20 client who just can't call it quits. I generally don't like doing shootouts unless it is for a large body of work. For one album, I usually say "Look, pay me for mastering a single, and if you love it we can do the rest and if you don't, well, you have that master to do whatever you want with."

K: There are a lot of DAW's designed for mastering like Sequoia, Wavelab, and Pyramix (to name a few). Which DAW do you prefer and why? Is it an issue of workflow or does one program actually sound "better" than the other (i.e. does one handle fades better? does one have better dithering, sample rate conversion, etc.?)

M: Pyramix and Sequoia are the best things going right now, but both are Windows only. soundBlade and Pyramix are tied for best editing, but sB is a buggy mess in every other way. Sequoia has the cleanest interface and best plugin handling. Wavelab is quite solid, especially given its reasonable price. It is the best option if you want to work in MacOS.

For dither, I use flat TPDF 99% of the time, and nobody does that wrong. SRC I do in iZotope RX. The fade editors in soundBlade and Pyramix are a class apart from all the others.

K: Do you have any advice for somebody who wants to start a career in mastering?

M: Aside from don't? It is a niche industry, and it is getting to be a harder and harder service to do exclusively and make a living. If you decide you have to do it, ideally find a gig as an assistant with an established engineer and learn and watch for a while before really worrying about doing any mastering. If you can't find that gig, or you don't want to wait, start small. Do cheap work and screw it up and it was cheap so what did they expect. Get a real monitoring set up ASAP, then some basic outboard gear. As you and your studio develop, charge what you're worth. If you get lucky and don't overshoot the prices your market can handle, you too can barely get by doing something awesome for a living.

Part 2: Mastering Subjective Language

Punch: Pft. Who knows? I think of it as clear, defined transients.

Warm: Appropriate balance between low mids and upper mids/highs, favoring the low mids.

Clear: Appropriate balance between low mids and upper mids/highs, favoring the upper mids/highs. Ideally a perfect balance is both "warm" and "clear."

Harsh: Excessive upper mids / highs, particularly in the sibilant region.

Groove: What a funk band does.

Frequency Ranges:

Low: 0 to 80 Hz. **Low-mid:** 80 to 320. **Mid:** 320 to 1240. **High-mid:** 1240 to 4880. **Highs:** 4880 + (basically, two octaves each).

III. Adam Ayan

Adam Ayan is a graduate of the Sound Recording Technology program at the University of Massachusetts Lowell. Upon matriculation, he interned at M-Works Mastering in Cambridge, MA. He then moved to Gateway Mastering in Portland, ME where he worked for a few years as a Production Engineer and Assistant to famed mastering engineer, and owner of Gateway Mastering, Bob Ludwig. Since then, Adam was promoted to mastering engineer and developed a highly successful career in his own right, amassing multiple Grammy awards and nominations. A select few of his clients include Rascal Flatts, Carrie Underwood, and Lana Del Rey. This interview was conducted via Skype conversation on Wednesday, November 25th, 2015.

K: Could you tell me a bit about how you first got involved in mastering? Did you get into audio first as a musician or were you more interested in the technical side of audio? Did you go to school for audio? Did you intern or have an apprenticeship?

A: I'd really like to go way back and started out as a fan of music. As a kid I just loved music. I grew up, of course, in the MTV era with the good days of MTV. So MTV and FM radio were like the two places I heard music the most. But as a younger person, I was just really really into music. By my early teenage years, I started playing bass guitar. I like to think of it as I went from being a fan to being a musician and then when I was finishing up high school I knew I wanted to go to

school for music, and I was gonna go to school for just music performance. But one of the bands I was in at the time, I played mostly rock music and in rock bands as a teenager. One of the bands I was in had made a recording at a commercial studio and I think, looking over your questions about the technical side of things, I really didn't at that time know very much at all about the technical side of what we do. I just knew I loved music and I loved playing music, I was a musician. And when I was at the studio, I remember the time going by really fast, really enjoying the process, it was a completely new process to me. And all those positive things about that experience… so when it came time to look for where am I going to go to school, what am I going to go to school for… I knew I wanted to go to school for music performance but I thought, of course my parents were like, you should go to business school. And I thought, what if I go to school for music performance and recording? There's this other thing, you know, that will broaden my horizons in terms of what I could do after school and what I could do professionally. And of course, I pretty quickly found the program at Umass and actually I applied for early acceptance in the fall because I knew at that point I wanted to go there so it was the one school I applied to, I got accepted and I didn't have to do any other applications or essays or anything else. So that was sorta how it started for me but once you get into, what, I guess it's second semester of sophomore year, you get into your first recording classes. Once that happened, my entire perspective on what I wanted to do changed. At that point I realized I wanted to become a professional musician anymore, I wanted to become an audio engineer. And all that technical stuff that I didn't really know or

understand before, I wouldn't consider myself a technical kind of person, I just realized that I had an aptitude for it. But of course, all the way through that, and to present day, I feel as though… the technology is interesting, I love technology, of course, but it definitely takes a back seat to music for me. And, you know, that's why we're all into it. Most of us are into it for the music side of it. I definitely think that there are people in our industry that really are into the technology side of it. And that is maybe more than the musicality side of things. For the music guy, the technology is a means to an end and what's important is making great music and great recordings. That's kinda my perspective. And like I said, I take a little bit for granted that I have the aptitude for the technology side of it and I know that stuff really well and can move forward with learning new things easily. It comes very easily to me, the technology side of it, but that's not what gets me going every day. So that's pretty much it, in a nutshell of how I started out.

K: As your career advanced, how did your philosophy on mastering evolve? Is there anything you know now that you wish you knew when you first started?

A: I think the one thing that I can point out to you that has evolved a little bit over the years, and that I feel like I learned fairly early on, but from project to project, you have a different mindset on is that if you look at what mastering is all about, for lack of a better way to say it, we take recorded music and we make it sound better. We make it sound as great as we possibly can. And that can mean a lot of

different things and it's a little different for each person. But, starting out I felt as though, you know, I'd be given a mix and of course, in theory, I feel like this all the way through today, that the mix has been approved by the client. Somebody liked what they did. There's always something that maybe needs to be fixed, and that's the thing. But we're starting at a point where the client really likes the mix, the mix engineers are happy with it, the producer's happy with it, the artist is happy with it. My job is to make it sound better. And starting out, I guess maybe I didn't take as many liberties as I did within a few years of really feeling comfortable as a mastering engineer and as I would today in terms of, always respecting the mix. And I'm in this great place where I get a lot of really great mixes so it's usually like, why do I make this mix sound better? I'm not going to turn it into something that it shouldn't be or isn't, right?

The one thing that I feel has evolved is, has kind of ebbed and flowed, for me depending on each individual project and the kind of things I get is, I learned fairly early on is to not to be afraid to go for it. You know, if in my mind, I'm like, Ok, well this is the mix that's been approved; everybody loves it and my job is to make it better, if I think I'm making it better, with instead of, you know, going this far with it and going that far with it, I learned fairly quickly to not to be afraid to go that far with it and see how the client reacts. And if it needs changing, we change it, you know like that. So that is something that I think evolved with me and continues to evolve because I think it's different with every project and I think you have to know the client you're dealing with as well as you possibly can to know whether or not if that's acceptable to them. You know, today, I might

work on a project for one client, they want me to go super far with it and for another client, they want me just to make what they gave me better and not take serious liberties with it. So I think that's one thing that is forever evolving and, again, is different on a project-by-project basis. And early on, I probably would have leaned more on the side of not taking super liberties with whatever I was doing… make it sound better, of course, and improve it as much as I can, but after I gained some confidence, I was like, I'm just gonna go for it, you know? And I think what's evolved is I realized you can't go for it with everything, not every client wants that mentality, so it changes on a project-by-project basis.

K: And you can get a sense of what the artist is looking for?

A: Yeah, absolutely. You know, I have some record producers who I've worked with for over a decade and I know what they like and I know some of the liberties I might take if it wasn't their production, it would be the wrong thing to do. So I show some restraint in those situations. And others I know they want me to take every liberty, you know, go as far as I want with it, and I do that. And what's interesting is the flipside is when some clients will be like, Oh I wish you would have taken that a little further, just do whatever you want with it. And in some instances, the flipside instances, I'd be like, well, but you gave me a great mix. You gave me something that is awesome, the artist loves it, I don't need to go that far. Like, you're imagining I could go further but as an experienced mastering

engineer, I'm gonna say, no I don't need to go any further with that because this is where it lives best.

K: Do you think it takes a few years to realize when to stop? Like, having the experience to know how much is enough?

A: Yeah, I think so because, I mean, I hate to say this but I feel like I learned it pretty early on but I think that when you're first starting out you kind of want to try everything. And sometimes it's like you can throw the whole kitchen sink on every project and you start to realize that you find with your ears and your experience that not everything needs that. So I think there was a process for me learning that but maybe the reason I knew that so early on was, having been Bob's assistant, his mentor. And I'll take that one step further, when I was assisting him, when I got to a point when I was assisting him when I knew exactly how he would master something, I felt like I just had a really good bead on how he would do something. And to that point, he would come in the morning after I set up and EQ'd a couple of tracks and most days he'd come in and be like, That's exactly what I wanted. The learning how far to go or not to go happened before that I think. You know, as his assistant. But at some point I felt that I got to this point where, there were a couple of years, of course, where I was mastering my own records and developing my own clientele but I was still assisting him. I was both of those things. I was still developing clientele and I wasn't mastering my own projects every day and the days when I wasn't, I was assisting him. Or on some

days, doing a little bit of both. And I got to a point, and all because of that assistantship experience and his mentorship, where I would be like, Ok I'm assisting Bob on one of his projects and I'm gonna do it this way because I know that's what he would want. I'm doing my own projects now and I'm gonna do them this way because I know that's what I want. There are slight differences.

K: So you're developing your own style.

A: Yeah, exactly. If anything, and I guess my own style, and I've had clients say this in different ways to me, but I feel like my style is similar to what Bob does but it's maybe a little more aggressive, so it's maybe a little more… pushing it a little more and going beyond the boundary Bob might have on something… if there are any differences between the two of us. And I learned where those boundaries were being his assistant. And I learned, I wouldn't cross that boundary on his project because I would respect that's what he would want. He would come in and want me to dial it back or change it if I'd done that. But on my own project, I could cross that boundary line. I knew where my boundary was.

K: Are there any projects that stick out to you as having a particularly challenging aspect or technical difficulty that you had to overcome? What was the nature of the challenge and how did you react to it?

A: Sure, I gave a little thought to this just before we hopped on Skype and there were two that I could think of that were pretty close to each other, and also I think they dovetail well into what I think the next question is. And one of them was this, and I might have played this. You were up here, right? And I played this. It's the Rock and Roll Hall of Fame concert and the other one was this Joe Satriani DVD. So both of those came within a year of each other and what was particularly difficult about both of those projects is they are very large in scope and there was a lot of work that had to be done. So in other words, the Rock Hall of Fame thing was about five and a half hours of music to be mastered in stereo and also mastered in 5.1. Yeah, right! And to further complicate that, it was a bunch of different artists. So there was this thing where when they did those shows, several artists got a set. Like, here was your forty-five minute set, or whatever it was. And it was like, Ok, U2 has a set and they'd bring up all these other performers. Springsteen has a set and he'd bring out these performers, Stevie Wonder has a set and he'd bring out these performers. So, not only was it five and a half hours long, but it was really varied musically. And then to add to that, when it was mixed, a vast majority that was mixed by the same two mixers, but some of the bands brought their own mixers in. So not it's like, it's all these other artist plus it's four or five mixers, something like that… and in stereo and 5.1. So how do you get through all of that work efficiently and get the result creatively that you want best? The Satriani thing was similar, yet different in the sense that it was shorter, it was one artist, one mixer, but it was stereo, 5.1, and 7.1. So there were three different mixes to master. So in the case of both of these

projects, It was like, Ok I have to do all of this, and I'm gonna get through all this work, it's gonna be complex, it's gonna be complicated, and it's gonna take a long time. How do I do this most efficiently? And jumping from, say, stereo to surround sound, how do I make the leap from stereo to surround in the most efficient way as well? And what I decided to try was, I'm gonna get myself a little outside my comfort zone, and my comfort zone at the time was to use a lot of analog. I'm gonna get myself a little outside my comfort zone. I'm gonna do this all digital and I'm gonna try to do as much of it in the box as I possibly can. And, of course, I've done a lot of projects like these and a lot of them I've done all digital but in this case, I was really trying to keep it all in the box, really try to be able to automate as many things as I could, really try to be able to copy-paste automation with the stereo to the 5.1 or at least efficiently get from, you know, stereo automation to the 5.1 automation and do all these things that would make the process as quickly as possible, yet still yield the best results as possible. So I went into it with that mindset knowing that if this isn't going to work out for me creatively, I'll just pull the plug and go back to my comfort zone. But in both cases, this is where I started to discover that I didn't need to stay in the analog domain, or go into the analog domain with every project and yield the results I wanted. So even in the box, I tried some new plug-ins that I hadn't used very often that I thought would be helpful in this process. So adding these things into the signal path is always a scary thing, especially in mastering. But I just went for it and started to realize, wow, this stuff in the box has come so far now for mastering that I'm not missing my analog, and didn't miss my analog in those two

projects at all. And since I've already got myself out of my comfort zone with those projects, I continued to refine my process. Like, do I want to go analog for most, and do I want to do that signal path for most of my projects or do I go digital? And I feel like those projects were sorta like the beginning of me starting to stay all digital and finding tools that worked for me completely in the box that were surpassing what I was doing with the hybrid signal path.

K: Ok, that can lead us into the next question. You obviously use a lot of digital. In a lot of the conversations I have with people, especially at M-Works, is that they can't do what they do without the analog gear. Do you think that's just habit or superstition?

A: I mean it could be a number of things. I'm sure certainly that it's not just the guys at M-Works that feel that way. I would have felt that way five years ago. I would've sworn up and down that my analog was doing better than the digital signal path and that the hybrid was the way to go. But I think it's habit, and I think in mastering, I mentioned that when you add anything or change anything in your signal path, it's not a light thing to do. You do it and you're really careful and you're really cautious. Every mastering engineer knows their signal path well and may have elements of their signal path that change depending on the projects, but the idea is that you know your signal path all the way around really well. So you start changing it up, it can be daunting and you want to feel really comfortable with that. But I've come to this place now that, and with those two

projects as examples of a starting point in trying to do everything digital, where I started to realize that I really do think that digital is better all around and I'm not finding that there are analog tools that are making me want to go back to analog. And in doing this, a number of times I've started off with my analog signal path, and said, well, I'm gonna do my normal thing, maybe I'll try the digital thing. And I started to realize that when I compare apples to apples, I was becoming less and less happy with the results I was getting through my analog signal path. And I'm not sure if I can put my finger on why. I mean phase-shift is always an issue when you use analog. If we had this conversation 5 or 6 years ago, and I would have said to you analog always sounded better to me, or a hybrid always sounded better, some engineers would say, Oh but you need to do a D/A and an A/D conversion and you're doing all these extra conversions, technically that's not a good thing. My answer to that would always be yeah, but I have the best converters that you can find. And I always find the sonic benefit of using my analog signal path far outweighed any issues of conversion and I still believe that's true. But, lately, when I've been trying to go analog, I feel like I'm hearing something that's not jiving for me and staying all digital is feeling better to me in a lot of ways. My other theory, we may have talked about this when we talked at Parsons', but my other theory on that is that a vast majority of what's happening before me now, you know, five or ten years ago, when folks were mainly doing things in the box, their tools weren't providing those colors, not necessarily the sound of just analog, but colors and vibe that you couldn't get out of digital at the time. And I feel as though there are some many really good colored plug-ins now

that are inducing, whether it is harmonic distortion or other things that we've always lost, things that have always worked well for music in the analog domain. I feel like those tools exist now in the digital domain where they really didn't before.

K: So emulations – UAD plug-ins? Stuff like that?

A: Yeah, whether it's UAD plug-ins, emulations of analog devices, or even just other cool new digital devices that sound better than they did ten years ago. Which I think is getting that color that we missed, not being in the analog domain before, those things are now getting colors for us at an earlier part of the process. And I never went out to analog just for the sake of analog. But, the things that I would get out of my analog signal path, I feel like I'm just not getting. And maybe now I am hearing the phase-shift better than I would have before because I'm not getting, the phase-shift was far outweighed by the sonic benefits of the analog path in the past that makes it not the case anymore. That's my theory at least. And really, again, like ninety-nine out of one hundred times now, I'm doing all digital. I'm happier with the results.

And it really truly is not an ease of recall thing for me. Recall in the analog domain for me isn't that hard either. But the sonics are really what's driving that for me. It's like, sonically staying in the digital domain is working better for me. And maybe not for everybody, but for me I feel like it's superior.

K: Ok, that's reassuring. It's like now I don't have to go out and buy thousands of dollars worth of gear.

A: Right, like if I had to outfit this room today, you know, I sit in front of tens-of-thousands of dollars of both analog hardware, digital outboard hardware… You know, it's great I still have them if I feel like it will be useful, I will certainly use them. But if I had to outfit the room again today, it would be a fair amount of plug-ins and I wouldn't need all those hardware devices. It would be a lot less expensive.

K: Right, and still have your pristine monitor path.

A: Exactly, that's the most important thing. Like you had mentioned, if anything came to my mind in this interview, that's one of the things that I come back to, is your monitor path and listening environment. You know, occasionally I get asked, Hey, I'm mastering at home or I'm mixing at home, and I want to spend some money on plug-ins and tools, what should I get? And I'll always say, well what are you using for monitors? And why don't you focus on your monitor path first because that by far is gonna make such bigger difference in experience in your ability to do your work well than any one plug-in or any other device will. So monitor path is super important. And again, yeah, if I had to outfit this room again, that's where most of the money would be is the monitor path.

K: How much communication is there usually between you and the artist or producer? Do you master the record as a whole and send it to them for revisions or do you send them a few tracks for review? What is your policy for revisions? Shootouts?

A: So every project is definitely different. If I were to say… the vast majority of the people that I talk to or email back and forth with… most of the time, the majority of them would be a record producer or a mix engineer. In fact, I would also say that those are the two people in the process that most often would be requesting me to master their record or whoever you're gonna master with. They're usually the folks who chose who the mastering engineer is gonna be. So those are the folks I talk to the most. I certainly talk to some artists, and indie all the way to superstar, it's usually a matter of how hands-on the artist is in this part of the process. In most cases, I would say most artists rely on their record producer to interface with me, and then, of course, they end up listening and the artist is a part of the approval process as well. But most days I come in, I get mixes, I do my job, I may get some notes from the artist, the producer, or the mix engineer. I do my thing and send it back to them at the end of the day. They spend, hopefully some time in a good environment listening, and then maybe come back with some revisions. Come back and say, hey it's all approved or they come back and say we need some adjustments. If they want me to make some adjustments, or now that they've heard it mastered, they want to make some mix tweaks, that sort of thing. And there's a back-and-forth until it's approved. So, the

communication can vary from project to project, what almost never varies is the fact that what I do will go out to a number of people for the process of listening and for approval and that almost always includes the artist, the record producer, and the mix engineer, and may include some other folks as well. So I'm not working in a vacuum. And I was actually just saying this to Gale yesterday and since you're at M-Works, I'd be interested to know if this is still how things go there. But, the way that we work here is we do our thing, we give our clients references, whether they're here, they leave with a CD or if they're in London or Nashville, whatever, we'll send them an upload, which is the most, how we do most of it. There's still an approval process and there's never a, "we master the record today, we cut vinyl masters, it's done" kind of thing. And I remember when I was at M-Works it was very different, it was almost every session then was attended. This is in '97 and '98. So every session I was involved with at M-Works was attended by a client, mostly because they were regional and they would usually leave with a reference and a vinyl master. And we've never operated that way, we've operated very differently. And in fact, sometimes the reason Gale and I had this conversation yesterday was that I worked with a client yesterday that was here, and I think their expectation was that when they left was that they were getting the final master. And not to think that they necessarily needed it yesterday, but when I asked them at the end of the day, what they wanted to leave with for their reference, they started talking final masters. And I said, oh no no no. I mean, what you're getting today is a reference that you should listen to, make sure you love, and then you should let us know and we'll make

final masters. That's sorta the next step in the process. And so at any rate, are they still doing it like that at M-Works? Where it's like end of the day, final mastering and it's done?

K: You know, I honestly don't know. I've only seen one client come in and actually do an attended session.

A: I'm sure things have changed. I mean, I was there eighteen years ago.

K: Yeah, and I think there is a pretty hefty revision process. Alex (Psaradukais) does a lot of stuff like Dropbox uploads for the single and he'll get approval to do the rest of the record.

A: Sure, so I was just curious. But in terms of your original question, that is usually how it works is we'll send a reference off to whoever's involved in the approval process, which again, always includes the producer, and the artist, and usually the mix engineer. And then they'll get… somebody in that camp will get back to us with revisions, if there are revisions, or an approval if there is an approval. Most projects, there is a little bit of a back-and-forth. You know, there might be a, Hey we love the EQ but we want to change the sequence or we want to send you a new mix. Or there may be a, We love it overall but we want to make some overall EQ changes. There's usually a back-and-forth until the point at which they approve it. And then we can start getting in to making final masters.

K: OK cool, moving on. So I know you guys are working on Pyramix whereas a lot of people use Sequoia or other DAW's. Why did you choose the DAW that you did? Did you A/B different ones? Was it a matter of workflow, stuff like that? Could you talk a bit about that?

A: Yeah for sure, absolutely. We definitely did that. At the time that we moved to the Pyramix, our entire world had changed around that time. So take one step back in this, which I think is a valuable part of this conversation. We worked on Sonic Solutions for a long time. And, of course, most mastering houses did up until the early 2000's, almost everybody was on Sonic Solutions. It was the best sounding DAW and at a time where there were some really bad sounding DAW's for whatever reason under the hood, things didn't sound good. Sonic always sounded great. Sonic's editing model and workflow was really tailored to mastering. But what Sonic couldn't do by the late '90's was it couldn't do hi-res audio, or it didn't do it well. And it didn't do multi-channel audio very well. And it certainly did not to at all do Direct Stream Digital, or DSD. And we were doing all those things. So it was almost like, within a course of a year, we went from everybody knowing Sonic Solutions to, at least me, being Bob's assistant at the time, knowing about four or five different workstations. Protools was just starting to come on for us and we were getting most of our digital mixes as .wav files or as Protools files, which at the time might have been Sound Designer 2 files or .aiff files. We were doing multi-channel into SADiE because we found that it was

the best for us at hi-res and multi-channel. We had Sonic's new workstation, Sonic HD, which really was just crap. It was meant to do hi-res for us and it didn't really work. It was really a nightmare. We had Sonoma, which was Sony's proprietary DAW for DSD. And we had one or two others that we used a little bit and we realized pretty quickly that this really sucked going from one workstation to another depending on the project. If we were doing DSD, I'd have to sit down at Sonoma, which was a really bad editing model. And I'd remember all my quick-keys for Sonoma, I'd remember all my tricks for getting past its bad editing model, and I'd go over to Sonic and have to reconform my thinking to that. And then I'd go over to SADiE… So at any rate, when we came upon Pyramix, we were looking for an application that did it all. At the time it was either going to be Pyramix, SADiE, or Sonic HD. And Sonic HD pretty quickly was out of the mix because it just didn't function very well. It's hardware didn't interface with its software. It was really clumsy. We just decided to ditch it. So it came down to Pyramix and SADiE, which SADiE was their latest and greatest at the time would do DSD as well as hi-res PCM. And they both sounded really good. They were both superior to other workstations sonically and they both had tools for DSD, which at the time we thought it was going to be a bigger deal for us than it turned out to be. Pyramix had better tools for DSD. And Pyramix also, so sonically there were those reasons. It sounded great. It sounded as good as SADiE, if not slightly better and it sounded better in the DSD domain, at least in terms of processing in the DSD domain and using actual tools like equalizers and limiters. So it had that up on SADiE and everything else. And workflow-wise, it allowed us to continue

the way we worked in Sonic Solutions. So the one complaint I hear from Pyramix users, especially relitavely new Pyramix users, is that when they sit down at Pyramix and they're like, how the hell do I use this thing? It's so complicated, there are so many ways to do things, how do I do this? And what I tell them is that you gotta narrow your field of vision in Pyramix to something that works for you. And knowing that about Pyramix before we got onto it, we realized very quickly that we could take that Sonic editing model that we all knew so well and we could port it over to Pyramix. In other words, we could put our Sonic hat on and, how would I do all these things in Sonic? We could make Pyramix quick keys, lets make Pyramix templates that look like Sonic Solutions. So source and destination, one, two, three, and four point editing and similar quick-keys. Because we knew all these other workstations, we also said, well you know there's this quick key here, this function in Protools that we really like. This quick key or function in Sonoma that we really like. Can we emulate that on Pyramix? And in a lot of instances, we could. So it was both about sonics and about workflow. And Pyramix felt superior to us in both of those ways. And then the workflow side, Pyramix just does things… I shouldn't speak about Sequoia or other workstations because I haven't had any experience with them. But in terms of everything else I've ever experienced, nothing else would conform to our workflow like Pyramix would, or did. And lastly, in terms of workflow, being PC based, it allowed us to network in a way that you just can't do on the Apple side of things, which is huge for us. I mean, you've seen our place. We have two mastering rooms, we have edit rooms, and everything is networked together. Yeah, so I can be mastering a

record today and one of our production engineers can pull up my record from last week that's now approved, has now gone through that process that we spoke about a few minutes ago, and they can start making masters. I don't have to stop what I'm doing to do final masters. I can continue on mastering a new project. So all of those things were big for us, and you know, that was a nutshell or a longwinded way of telling you why we chose Pyramix over everything else.

K: Do you find that you're still getting many DSD projects? They just come along now-and-then?

A: Now and then is probably the best way to say it. They're few and far between. It seems like every couple of years, I feel like over the past few years there have been a bit of a push for DSD again. And I think what's probably driving that is a lot of people, certainly on the coast at least, have enough internet bandwidth that if they want to buy hi-res audio, they could buy and download DSD audio. I think the major record labels view hi-res audio as a means of moving forward in terms of, I don't know if I necessarily agree with it, but as a business model… they probably love the fact that they can sell a DSD file for about three times as much as they can sell iTunes files. You know, because there's an audience for them. There are people who'll buy them because they're DSD files. And that's not to say DSD doesn't sound superior, it certainly does. But I think that's what's driving it now. With that said, they're still few and far between. This past year, I've done a handful of them and they've all been live Bruce Springsteen records.

So for the past year, and hopefully moving forward, Springsteen's camp has been going back to his archives in terms of recorded concerts and either working from analog, in terms of the early ones, working from two-track analog, or remixing… most of them are actually re-mixed and they're being mixed to DSD and then I am mastering them DSD. And then they're either for sale at livespringsteen.net or nugs.net, which is a company that does a lot of live music for a lot of artists. They're being sold as DSD files or hi-res PCM files, audio CD's and the whole gambit. That's where most of my DSD work has been. There really hasn't been a lot of it.

K: Do you find that it sounds better than a 192-24 bit file? How is it different?

A: It has been a while since I've done a comparison. But we had done those comparisons ad nauseam like ten or fifteen years ago and it sounded slightly better in different environments. It sounded slightly closer, if you actually took, say, an analog tape, and you digitize it 192-24 and you digitize it DSD, you know, essentially from the same exact signal path, you know a "Y" to go each direction. And in an apples-to-apples comparison, the DSD always sounded a little closer to the analog, so slightly better. I would counter this, though, with the notion that in the DSD world, the tools are extremely limited. It's almost like that discussion we had about analog and doing analog conversion, D/A – A/D conversions, at some point now, we always found that whatever ding you might get when doing multiple conversions, was far outweighed by the overall sonic benefit of using the

analog signal path. Analogous to that, I sorta feel as though DSD versus PCM, even say, 96-24 PCM, the amount of "better" the DSD sounds, it was usually far outweighed by the fact that there are so many more tools in the PCM world than the DSD world. Once you commit to DSD, at least, beyond just committing yourself to using it just as a recording format, that's where it thrives. But if you commit to it beyond that, as an overall production format, immediately your hands are tied by the tools that are available and that means there are very few tools in the DSD world. That's why DSD never took off. At least from a production standpoint is that the tools didn't really exist early on. Here we are fifteen years or so later from where we first started doing DSD and they haven't improved very much at all. In the PCM world, there is just a plethora of really amazing sounding tools to use. So that's what made DSD a hard sell all these years.

K: Ok, I guess the last question is a bit selfish. Do you have any advice or words of wisdom for somebody who wants to start a career in mastering?

A: I think the best advice I could give you is to try to find a good mentor. I think that in 1997 or '98, for me was a little easier than now it is for somebody in 2015, but it still exists, there are still opportunities out there. And I think that that is really important because, you know, to get back to one of your first questions about the things that have evolved or changed in terms of my process and what I learned early on in terms of how far to go with something or whether I've gone

far enough, I learned a lot of those things by being an assistant, my being Bob's assistant. I learned those things a lot faster that I ever would have learned them if I just decided, I'm going to be a mastering engineer, I'm gonna buy some plug-ins and a couple of the workstations and start offering that service to clients and sorta learning on my own. There's a huge benefit to having a mentor. There's also a huge benefit to being around your peers. And we're a smaller, in terms of mastering engineers, it's just Bob and I, and we have other engineers here. But we all learn something from each other, especially Bob and I learn things from each other every day, like oh hey, I tried this new plug-in, you should try it out. Or hey, I tried this new plug-in, you should watch out for this or that that's not super obvious. You know, I think that even kind of before my time in some ways when there were larger recording and mastering facilities, you could learn from a number of mentors and you could learn a lot from your peers. So, those would be my biggest two pieces of advice. Find a good mentor, and really spend a lot of time with your peers and try to learn a lot that way because I think that in 2015, for several years now, folks are more apt to be in their own little bubble, like you miss… there's a lot of experience and a lot of things you miss out from sorta knowledge sharing, and even just, off the cuff, decisions, whether or not it's creatively or technically that you get from those interactions.

K: Yeah, because I've thought about that. Especially as a mastering engineer, you are able to just sit in your cave by yourself and do mastering and not have the interactions with other people. That's one reason why I like UMass Lowell so

much is, you know, with our friends, we're in the studio every week, talking about audio… Yeah, it's just a good community to like, bounce ideas off of each other and to learn from each other.

A: Yeah, and I think that that's invaluable. I think that if you can continue that community in the professional world and in your professional life, you'll gain an awful lot from that. I think that stuff is super important. Those are some of the things that, unfortunately, the way the record business is now, it changed a little bit and it's harder to find those situations, but they exist. I mean that's why I got to things like Parsons' and AES and do those things because we are on a bit of an island up here in Portland. Bob and I have each other, I certainly have some friends that are in the business that are around here that do really great work and I'll get beers with and chat with but, you know, we're not like in New York City where I can just be in one facility and walk down the street and visit friends, that sort of thing. So, I think there's a lot of value to that. And you weren't able to make it to the educational summit at the end of Parsons', but I think the very first point that came up, or the very first question that Paul Leerman posed to the panel is, I'll paraphrase, but it's something to the affect of, should we be teaching professional audio, especially in 2015 with limited job opportunities, especially in music production. And through the conversation that came out of that question, one experience I remember having at UMass Lowell that always stuck with me was… Well let me tell you the experience and I'll tell you what I mean, but did

you guys have to take a class called Recording Industry, well were you undergrad at UMass?

K: No, I wasn't.

A: No, that's right, you went somewhere else. But, as an undergrad, there's a class called Recording Industry when I was there. I think it was junior year and it was basically like a means of explaining all these areas and options for your professional livelihood when you finish up at UMass Lowell. Like, don't just think about being a record producer, or whatever you're thinking, you should know that all these other things exist, which is a really powerful thing. But I remember in the first class, first day, first class, the guy that was teaching it, Marty Pulland I think was his name, one of the first things he said, basically was like shitting on the record business! Saying, how many of you guys want to own studios when you graduate? Only at this point two or three people raised their hand. He was like, you're not going to own a studio, you're not going to open a recording studio, right? But out of that conversation, one thing that came up was that he said, if you think you're gonna be Prince's engineer at Paisley Park, that's probably not gonna happen. And I remember sitting there being like, uh, that was at the top of my list of places to work. That was before I knew about mastering, before I knew that's what I wanted to do. I was like, I love Prince. He has this studio called Paisley Park, I wanna go be his engineer. But what I got out of it was, you know, I said that and another gentleman that was on the panel said, well,

my friend I went to Berklee with ended up being an engineer at Paisley Park. Anyway, my whole point of this story and the reason why I'm bringing it up to you now is that Marty Pulland would have also said, you're not going to become a mastering engineer at Gateway Mastering Studios and have a room build for you at Gateway Mastering Studios. If I had heard that then, maybe I wouldn't have come down this road. I think that he would have said the same kind of thing about that. And my whole point is, find a good mentor, get out there and do it, and especially when you first get out of school, it's like, go for what you want to go for. Even though it's infinitely more difficult as it ever has been, but the opportunities exist and if you're the right person to do it, with the right mindset and the right work ethic, you'll find an opportunity similar to the one that I have. Very few people do find them but they still do have them. Don't shy away from trying because you won't ever know if you don't try. You know, your career is probably gonna zigzag a bit and maybe not be as linear as mine has been because I think that is kinda unique. But just go out and do it and don't let the Marty Pullands of the world tell you that you can't. He was like, don't expect to be an engineer at Paisley Park. Ok, I crossed that off my list.

K: Yeah, when I first started at UML, I was like, OK these guys send people to Avatar, all these big studios in New York. I was like, I wanna do that. But then I heard the reality of the situation and I was like, Ok so maybe not. That's not for me.

A: Right, well, the thing I love about the UMass Lowell program, a number of things I love about that program is that they arm their students with all the stuff that they need to be good as professionals, but they also arm them with reality. And I also don't think that they really completely discourage students by any means from doing what they want to do even if it's going to be a difficult path. I think that they encourage them to do those things and they also arm them with the reality. You know, this isn't a glamorous business. I mean to that point, I was at the Latin Grammys last week. I won album of the year, but I felt a little slighted by the recording academy because they didn't give me really good seats and I felt like they don't always take great care of their nominees, aside from maybe the artists themselves. But that's the way this business is. The Grammys are glamorous and winning one is certainly glamorous, but it's not the glamor that most people might think it is. And I feel like at UMass Lowell, they do a really good job of keeping you grounded in a positive way, giving you the reality of what our industry is in a positive way while opening up all these other frames of reference of other things you could be doing in this industry that might be just as interesting to you as being a record producer or being a mastering engineer, or whatever. I'm definitely one who's like, go for your dream, go do it and try to make it happen when you first get out of school and it's a good time of your life to do that. And it may very well happen. And if you don't try, it's certainly not gonna happen.

K: Ok, I have a sorta unrelated question. But it kinda came up when you were talking about how you and Bob share ideas and stuff. But I've always been wondering, if I may ask, I know there's inevitable hearing loss that happens, is there anything that you guys do to protect your ears? Like do you take a break after an hour just to rest your ears? Is there anything you can do to prolong the inevitable as much as possible?

A: Yeah, no, I think protecting your ears is the number one thing you have to do. And for both Bob and I, I mean neither one of us go to a lot of shows. And part of it is we're super busy. And I can speak for myself in the sense that I have a super busy studio and I have a young family so going out isn't something I do a lot anyways. But when I do it's always with earplugs. It just doesn't exist going to a rock show and any show for that matter, without earplugs. At the Latin Grammys I had my earplugs. On the plane, earplugs are something that I had with me. Earplugs are something that I carry around with me and I use them very very often. And I'm walking down the street with my dog or if I'm out running, I live in the city, if a fire engine comes by, I'm not too proud to do this (plugs his ears with his fingers) as it's going by. So it's really protecting your ears in all instances that I think is the most important thing that will help stave that off. And then in the studio, for me, I mean I listen 85, sometimes 90dB SPL while EQ'ing. I do take brakes. They're not regimented, like on the hour, but you know, if I start like I need to walk away for a few minutes, go get a glass of water, get a cup of coffee, stop and do an email, that sort of thing, those breaks I think are really important

for my ear and for my mental fatigue. I find that, maybe because I don't listen super duper loud, but I find that mental fatigue sets in much faster than ear fatigue. In fact, ear fatigue is almost not a thing for me. I don't listen super loud for long periods of time and I do take those breaks. I'm also fortunate enough to be listening in a really well balanced environment. So it's not like eighty-five to ninety where like, two to five kHz is just tearing my head off the whole time, or something. I listen on a very well balanced and very neutral playback and monitoring situation, so that helps. I think everything that you possibly could think of doing to protect your ears, in the studio, meaning don't listen very loud, take those breaks. And everything you can do to protect them outside of the studio, meaning wearing earplugs in any loud situation, you've gotta do and that's the way to do it. Interestingly enough, you were at the Platinum Mastering panel, right? At AES? That question came up, and I was the first one to say anything and I was like, well that's a loaded question. And Tom Coyne's response was I don't want to know. He's like, I don't want to know. I feel good, I feel like my ears are fine, and as long as my clients continue to love everything that I do and keep coming back, and the people who are working with me feel like I'm in a good place. And I mentioned that to Jonathan Wyner, you know, a mutual friend of ours, and he actually takes the opposite tact, which I thought was interesting. He annually goes to get his hearing checked so that he feels like he can at least have a baseline as to where his hearing is. I sorta lean more on the Tom Coyne side of things, maybe not as brashly, but I do everything I can do to protect my ears. I don't get my hearing checked often. I can't remember the last time I did. Maybe

as I get older I might start falling more… if you think of those as two extremes, Tom's take on it and Jonathan's take on it, maybe as I get older, maybe I'll start leaning more toward Jonathan's take on it. But it is a really important consideration, to make sure that you protect your ears everywhere that you go.

K: Yeah it's funny, my wife always makes fun of me. Even when I'm just walking to campus, I have my earplugs in because, yeah, loud trucks go by and it's just annoying.

A: Annoying too, and I think just as a person in general, being a human being in general, society's pretty loud in general and it can be annoying. And I really do think it grates on your whole body and mind in some ways. So I applaud you for doing that. You never know, you live in Lowell and a fire engine comes by that's super super loud and you happen to be really close to it, it could hurt your ears and I think it's wise to do that. Even at movie theaters, pretty much anywhere I know it might be loud, my earplugs are with me. It really makes a difference.

K: I know, we were watching the new Mad Max when it came out and the whole time I was like, ah!

A: I know, any, like action, shoot-em-up movie. I went to see The Dark Knight Rises when it came out a few years ago at our quasi-IMAX here in Maine and I had my earplugs in and I was like, it's just unbelievable how loud these situations

are. I mean, we'll go to shows and it's almost criminal just how loud shows are. I mean, the Grammys, it was super loud in there. It's like you are putting on a show for, in that case, the entire Latin music industry, but a lot of people like myself in a lot of genres of music, or the regular Grammys for that matter where the entire U.S. record business is there and you're making them all deaf… It's not cool!

K: I know, it would be a terrible tragedy if everybody goes deaf.

A: Right! It'd be good for you guys getting out of school (laughs).

K: Yeah! But we do need their expertise and experience too because we're a bunch of dumb kids.

A: Right, well I don't think that's the case. You're probably a lot smarter than you give yourself credit for. Also, and with the ears things. I think that in general most people have better ears than they would give themselves credit for. And then somebody like yourself who is exposed to so much critical listening, whether it be academically or what you do on your own out of interest and knowing that this is your profession, you probably have better ears than you think. But yes, having a mentor to help you with the experience and their even more refined ears is really important as well.

Part 2: Mastering Subjective Language

Punch: Punch to me is good low-end response, though good and tight low end response, a good glue between the kick drum and the bass guitar in the world of pop and rock. And also to me is the main driver of the rhythm section of any recording. That's how I would define punch.

Warm: Warm I think is similar, yet different than punch. It's about the low end. It's still about… I was gonna say, It's about warmth. No, it's still about the low end. It's still about the bass guitar and the kick drum in most pop and rock music, but I also feel as though it's not about the tightness or that glue as much between the kick drum and the bass guitar or the low end in general. To me, if you talk about the difference between warm and punch, warm is a little more open, a little more subdued, and less tight, less glued. And also, unlike punch, to me, warmth not only is about the low-end response, but is also about a duller top end. I don't think you have warmth and you have brightness in quite the same way. You would never say something that's warm is bright. Not only is it a focus on the low end, and again, one that's not tight or as glued as punch is, but it also is taking away from the top-end in some way.

Clear: Ah, clear. I don't usually use the word clear a lot, but I use clarity quite a bit. It's usually about detail, and often it's about detail in terms of individual instruments. If I'm describing a recording that sounds clear to me in some way,

maybe that means I can pick out individual instruments in their own clarity. Frequency-wise, it usually means upper midrange to overall top air. Things that open up clarity and open up upper harmonics on an individual instrument.

Harsh: Harsh to me usually means upper midrange, too bright, unmusical, and a listening experience that is not pleasant at all. And the kind of thing where that part of our hearing range, that 2-4k range is sorta being overwhelmed with that kind of frequency response. And it also means lack of low-end warmth.

Groove: Groove is similar to punch for me. And maybe groove is somewhere between punch and warmth. And again, a focus on the low-end, a focus on the lower part of the kick and the bass guitar. Maybe it's more glued and tight than warmth is, but not quite as much as punch.

Frequency Ranges:

Lows: Low to me is all the way up to 80Hz, maybe 100 at the high end of it.

Low-mids: Whereas low mids, I would say 100 to about 500Hz. And, of course, there are a lot of ways to differentiate between those two things but, in terms of just overall perception, I really like low but often don't like low-mid. I can tolerate accentuated low a little bit but I cannot tolerate too much low-mid. Perceptually, low-mid to me means cloudy. It also can mean sort of a fundamental

of a bass guitar, depending on where the instrument is being played, but often as you get further away from low and further into low-mid, I get cloudy, that's how I would describe it.

Mids: I would call mid like 500 to 1,500 or 1,600Hz.

High-mids: Whereas high-mid I would call say, 1,500 and maybe on the higher side, 2k, upward to 6 to 8k. High-mid to me is a double-edged sword. It's very important in terms of transient response on most instruments. It can be important in the clarity, the clear range. You know, if you don't have enough crack on the snare drum, because you don't have enough high-mid, then there's a problem. It's a double-edged sword to me in that you can very quickly get into the harsh zone that we talked about a minute ago with the high-mids. It's really important that they're well represented and that transient response is good, but you can very quickly get into a danger zone. And I think our ears get tricked by that danger zone sometimes. Maybe it's because of the sensitivity of our ears or maybe it's because that transient response that lives in there is so important to music, that sometimes we can cross the line in terms of OK high-mids into bad high-mids pretty easily and we can fool ourselves into thinking that the bad high-mids are OK. And then that cold light of day, listen the next day can be like, Oh now we've got into harsh.

Highs: So highs for me mean 8k and up. I can find a lot of pleasing things perceptually in the highs. It's usually to me about upper harmonics. Again, you can, maybe not quite as bad as you can in the upper mid range, but you can go too far easily and still be pleased by it and trick yourself a little bit pretty easily. And for the entire overall spectrum, like, if I think about a part of the spectrum that makes me the happiest, it's usually the low. Yeah, and I guess that's why I'm a bass player.

K: Yeah, and I totally agree with you about the low-mids, like, 200-300Hz. I usually refer to it as being muddy.

A: Yeah, and that stuff is usually, I've found over the years, having done mix evaluations for people, and working with a lot of really really good engineers too, that that range is the range that is most commonly problematic for recording and mix engineers. Maybe for mastering engineers as well in lesser environments than my own, but that is the most problematic range because I think it's a difficult range number one, and I think, in terms of monitor path, it can be the range that lies to you the most. I feel that that range is the hardest to get just right in a room. And for most recording and mix engineers, it's the range that they struggle with the most. And they might not hear that they have too much of that range in their mix until it gets to me or until they bring it into another environment that's not lying to them about that range. So they end up with muddier mixes because of

that. Until they learn that about their space, and or fix that problem, which is a hard problem to fix I think.

IV: Paul Angelli is a graduate of the Sound Recording Technology program at the University of Massachusetts Lowell. After leaving UMass Lowell, Paul lived in New York City for many years working in the indie-rock scene as a producer, manager, and audio engineer. He eventually found his way to the famous mastering house, Sterling Sound and worked there for nearly a decade. Paul maintained his own client base of predominantly indie-rock artists and also was the main assistant to Ted Jensen. Paul has since left Sterling Sound, currently works for Bose, and has returned to UMass Lowell to pursue a masters' degree in Sound Recording Technology. This interview was conducted via phone conversation on Saturday December 5th, 2015.

K: Could you tell me a bit about how you first got involved in mastering? Did you get into audio first as a musician or were you more interested in the technical side of audio? Did you go to school for audio? Did you intern or have an apprenticeship?

P: You know, I stopped working in mastering in 2006. So I'll fill ya in on my story. I started working as a mastering engineer probably in about 1993. I was at Sterling Sound for years – for eight years I guess. One of the reasons they were

interested in hiring me is I had been working in the indie rock community for years as a producer, engineer, live mixer. I was known as a bit of a computer geek, you know? I was the audio engineer that was the "geek" at any studio that I worked at. Which basically meant that I'm no super genius at computers but I knew more about computers than just to push the buttons on the applications. You know what I'm saying? So when they brought me in, they were interested in getting someone out of the New York indie community because up until that time, this is the old music business now. Like, I'm pre-dating the '90's like when they wouldn't even take a client in to do a mastering session unless you had a P.O. from a major label. By the early '90's, you know, IRS records, FFT, the indie rock thing, for you, this day of the web and Sound Cloud and all this stuff, which is great. It's hard to understand really what it was like when the major labels had total hegemony over how music got distributed. Indie rock was really indie rock. They really comped it on radio stations at colleges. But that was even a very different thing. You know, you would subscribe to fanzines that you would get in the mail, stuff with your favorite bands. You know, really word of mouth. It's crazy, right? You learned about stuff at the local CD store. It was probably CD's and all that. There was vinyl but this in not the old-old time

K: So we're into the CD age now?

P: Yes, firmly into the CD age. So I'm just tellin' ya, when I came into Sterling, now they wanted someone who was an indie guy and I think a lot of the bigger

facilities, the way to get your foot in the door was, you know, great you have your client base, you start building it, and you'll also do production work. Which meant, you know, editing, QC, compiling, master generation. That kind of stuff. Because then, the tools back then, Sonic Solutions was the only tool back then, which got eaten by Roxio. So everything had to be done in real time. So one of two workflows would occur. So I was Ted Jensen's guy for like three years and I maintained my own indie rock client base. So Ted, on his records, would start EQ'ing and then once he had, if he needed editing or anything during the day. Depending on the record, the budget, the timeline… Sometimes the client would kind of walk between his studio and my studio where Ted would EQ pieces and then edit with me. Sometimes my involvement would just be at 4:00 Ted would come in and say the record's ready, you know, make the masters, do the QC. We were sometimes mastering actual PM CD's to the CD format, but back then, the big robust format for replication was ¾-inch video deck with the Sony 1630 processor on the front-end to get the digital audio off of it. Just to paint the picture to you, there were still lathes in every room, but there were only two people actively cutting vinyl at that time. One of them was, of course, probably the most legendary vinyl cutter in the history of music, George Marino, who I was lucky enough to be there when we did the entire Hendrix catalog to CD and to vinyl. You know, the Zeppelin stuff. Actually Zeppelin, predated my time there. I did the Cat Stevens catalog with Ted… Anyway, just to fill you in. The other guy cutting vinyl was Tom Coyne, who was unquestionably the top R+B and Hip-Hop and pop music guy. So I was lucky enough to sit at the knees of three of these

guys for years. It was a really cool experience. Each and every one of them, particularly George Marino, he has unfortunately passed away a few years ago. But, that man was an angel on this earth. You could never meet a lower-key person. To know that everyone from Metallica… you know what I mean, to Yoko Ono would absolutely not do a record with anyone other than him. You just sit in that room thinking this guy's amazing. So anyway, just to give you a little, because this is going to frame a lot of my answers here. So, you know Adam (Ayan) and I can connect you with Ue Nastasi down at Sterling as well.

K: Yeah, that'd be great!

P: Yeah, really great engineer. Really more into the, you know it'd be good probably. He's really more into the metal scene… to give you up-to-date stuff. You know, the technology changes so much faster than it used to. So, alright, that's sorta how I got involved. Yeah bla bla bla… (reading from the questionnaire) audio as a musician or are you more interested in the technical? Yeah, I did both of them. From the minute I was a kid, I was really really into it. So yeah, you know, I went to the UMass program. I'm like graduate number ten. I came in either the first or second year when you could actually enroll as a freshman in the program. Like a guy like Bill Carman… Bill was an E.E. major and then Moylan started the program and Bill was able to migrate over half way. Just to give you a little background.

K: As your career advanced, how did your philosophy on mastering evolve? Is there anything you know now that you wish you knew when you first started?

P: You know, that… how has it evolved? That's a good question. This one I'll answer from something I actually learned from Ted that I think is actually one of his key strengths that I saw. So, Ted, what I learned from him, that was definitely an evolution for me was, listening to the raw tracks when you hear them and understanding whether your approach is to do absolutely as little as possible or whether it needs major surgery or somewhere in between. And bringing that open mind to the table with every project. And, of course, then (this is where I'll sound like Alex Case), having done your homework and having the skillset to take any one of those approaches. I've never seen anyone like Ted just be able to listen to a sampling of the tracks at the start of the day and just know the way to go. Does that make sense?

K: So it comes with experience…

P: Yeah it comes with experience and then part of it, for me, is talking to your client or the artist and understand… and you know, this is gonna sound weird, not necessarily listening to what they tell you, but what they're really saying. I had artist who, they were working in their studios or whatever, they were listening to

their record the whole time but they were hearing it kinda in their head. And you know, they would come in and be like, yeah man these tracks… and I'm just throwing out an iconic record, but they'd be like, yeah it sounds like Houses of the Holy. And then they'd hear it on another system in a neutral environment for the first time in your mastering room, they would be like, huh. That doesn't sound like what it sounded like. You know what I mean? It wasn't just that it was different speakers, but that now they were putting on different ears. Does that make sense? So a lot of times, talking to people to really understand what they were thinking about the sound of their own music and what they meant by words, which your second page is kind of funny about that. So it was that kind of client thing is the other part that evolved and I think it does for everyone. Like so when a client would say, Can you give it more air, and then turning and saying, Can you describe what you mean by air? Inevitably it would be like, what they really meant was, could you get more out of the reverb tail on the hi-hat during the breakdown. You know what I mean? It was like, hey we're not really talking about air, could you give me more high-end? Well what do you mean by high-end? Well I mean shimmer on the guitar. Ok, that's not necessarily high-end. Great, we'll bring that character of the guitar out, no problem. So the other big thing was really learning to… the language that I needed to really understand really what my clients were saying, what they were hearing, and what they were after.

K: Yeah, so that's what the second page of the interview is about.

P: Yeah, so that answer was basically, that was my second page answer for ya. You know what I mean? I think it's great that you are after it. You know, funny enough, when I was in the grad program, in some of my classes with Alex, we had very deep discussions particularly about the term, punch, and what it meant to a snare drum and a kick drum in particular. But my big take away from that is you already get what the challenge is. What really matters is, with every artist you work with, what do they mean by these words? So you're…

K: You really have to be the interpreter.

P: You're already ten years ahead ahead of where I was at your age, lets put it that way (laughs). So you know what I mean? So it's good. Alright, (reading from the questionnaire) are **there any projects that stick out to you as having a particularly challenging aspect or technical difficulty that I had to overcome?** You know, nothing really sticks out now, but again, it's about communication. I had these rock clients I did a lot of records for. Really really good, like downtown indie rockers at the time. And in the late '90's after everything had moved to digital, they decided to make a record on analog. This is another communication story. So they came in, actually, who's the artist? Anyway, it'll come to me. I think it might have been Patty Larkin, was a big woman rock artist at the time. Ok, so, anyway, she came in and we were working on the tracks all day and she was like, yeah Paul, we recorded to 24-track analog, it's awesome… bla bla bla.

And we got it and it was good, but it had analog hiss, which people had kinda forgotten. And we mastered the whole thing. It had an appropriate amount of analog noise. It was very well recorded. So we got it all together, they did a great job. It was a really great record. For me, it was a straight-forward day. I assembled the whole record, edited it, had all the PQ points in and we were going to take the master and the producer, I remember this guy's name, Freddy Catz, and he said, alright, fantastic Paul. So when do you remove the hiss? And of course, we had Cedar, which I think now is still a software package, I think they're still around. They were ahead of the curb with noise-removal at the time. Now iZotope is clearly the winner now. And fortunately for me, I worked with them enough that I looked at Freddy and I said, you know, this thing sounds great Freddy. You remove the hiss when you record it. And he looked at me and he was like, right it's an analog process. Ok, good answer, thank you. And you know, with clients that I didn't know, that could've become a two-hour you've gotta be kidding me lets go get the Cedar hardware archival tape-hiss reduction and see if we can make it work. So, just to me, it's funny that technology is the technology. It's all about, especially in mastering, it's all about communication. You're the midwife. They've been working on their baby and you're the one who's gonna birth it for them. It's another story where, really, ultimately where like matching the communication style and getting that message across to the client. Trust me, I've said the wrong things plenty of times as well. Fortunately on that day, it was a client I knew well and I knew he would understand. And the other big one, which I still think is a challenge for people is learning about digital distortion, like

the really subtle things about digital distortion. When you're working to get the last tenth of a dB out of the program material. The tools now are way way more advanced. In the early years I did it, we had all the good look-ahead program limiters, that kind of thing that are standard now. But the first four or five years of mastering for CD there was none of that. And man, did you really have to learn how to get it a tenth of a dB over, then bring it back. The concept of figuring it out what we're absolutely gonna be the loudest passages of the entire record, and making that to the tenth of a dB as loud as you could and then letting the rest of the dynamic range of the record fall where it should around that. So that was the other challenge. And as I said, I still think it's a challenge now, the guy, the engineer behind Meridian Audio in the UK. What was his name, Bob? Anyway, he's the guy who invented Meridian Lossless Tracking, which is now Dolby's lossless… Dolby then licensed it. It was the DVD audio format… Bob Stewart. He has written… you may want to look him up. In my research in audio, I was like obsessed with this guy, I think he's great. He's really pushing the envelope of what do we need in digital to really capture audio, manipulate it, and then play it back as humans can actually hear it. He's a big proponent of, you don't need too much but you need enough. The last time I checked in, and every couple of years, he does a survey of digital audio strategies for human perception paper for AES and the one five or six years ago was awesome. He was like, really what we need is 52k 20bit: here's the reason why. And he's not saying that should be the format, but he's always trying to hone in. He's like, 192, 24bit: overkill for a delivery format for the following reasons, appropriate for an archival format, you

know. So you may want to look up some of these Bob Stewart papers because he has some super fascinating, really great stuff about digital distortion, particularly with respect to bit depth. And if you look on Meridian Audio's site, some of the papers he has just published there so you don't even have to go like, through the AES. It's really amazing. He shows how manipulation of digital audio at various bit depths really induces even how the processing induces distortion if you don't dither it correctly and really understand your chain. So I think that's a really important thing that honestly I saw a big separation between the understanding and the way we treated that at Sterling and how this kind of next, how some of the mastering facilities that I dealt with here in Boston when I came back didn't have a (subtle) in engineering. Now places like Sound Mirror: world class. I mean there are people who are doing it right. I'm talking more about people who are doing the five hundred dollar a day home mastering, which I personally think the technology is there to do. And it's a great way to go and I think that's where you see a really wide discrepancy between people who totally get it and people who totally do not get it, alright? So that's the other big thing. Ok next question… bla bla bla.

K: Yeah, it's basically digital, analog, or both.

P: Yeah yeah, and I'd be curious what Adam has to say. These days, he'd probably pick, you know, the tools in the digital domain probably exist where you could do most of your stuff in it. Although, you know, I can say the analog stuff

I've had in my room at Sterling, you know, it was ridiculous because I had like, original parametric EQ's handwired by George Massenburg.

K: The Sontecs?

P: What's that? Yeah, you know, I had the original Sontecs and the big old Focusrite, I think it's the 200 series, that they don't make anymore. So even then though, it was totally hybrid. And in a sense that, so the mastering, the format was people who had money when I was mastering, at least the first couple years, still mixed to half-inch because it was better. Your only mix formats were, you know, DAT tapes and all that. You know, no one wanted to go to 16bit as a delivery format if they could avoid it, right? Then 24bit proprietary came in in the mid '90's. Tascam came out with a DAT machine, Alesis made actually and amazing 24bit hard-disc/disc format. And I gotta say, even the DAC's in this Alesis were mindblowing for the money. They just sounded great. Anyway, so my sources came kinda half-and-half. But at that point it was a hybrid deal because I had such good stuff. So, whether the source was analog or digital, I brought it in digital… let me think. I did the analog manipulation part of the chain first, then converted it back from the analog domain to 24bit, did my digital processing… At that time, you know, it's kinda standard issue with, what was then to bring it into Sonic. My thing was to bring it in four or five dB softer than I needed it, still at 24bit. Then have a loop back with, you know, there was no mastering-level digital EQ at the time in software, absolutely not. Zsys did do that great. Daniel Weiss, I had a

couple of really high end digital EQ's that I then had in a loopback situation in Sonic. So I would keep that handy for the finishing-touch part like, once I'd get the record, it would be close, and once I would lay in the sequence, I would be like, eh track five to match will take a half a dB at 6.3k. So I would keep that in and then I would, once I had the whole record where I'd wanted it, then I knew, back to the earlier comment, then I would know where the loudest passage is gonna be. Then I could globally get the gain up to where I wanted it by whatever method was appropriate for their audio and for the tools that I had at my disposal, and then I would do my pass. So it was a hybrid system kind of in that way.

T**he next bullet point, "how much communication"**? Back when sessions were attended, 99.5% of my sessions were attended. Because there was no digital. It wasn't like, hey I'm putting the files on your FTP. Also, again, due to my relationship with the indie rock community, it was really… I'm not going to say it was a family affair but these were my people, you know? Part of why they wanted to work with me was because they wanted to spend the day at Sterling Sound to be quite honest, you know what I mean? So they were there, that was that. They would take a reference, back then it had to be a disc. So few people could even afford… back then, Protools was way more expensive than analog, think about that. So then they would go back and they would hit you with their notes and you would determine was it adjusting the gain of a couple of tracks, was it a global, hey the whole record needs 2/10^{ths} more around 8k. You know what I mean? Was there some global thing that they wanted? So it could be anything from there's

one track that sticks out gain-wise or EQ-wise or, hey, the whole record, yeah it's good but I'd love just a little more of, you know, the bottom two octaves. Alright?

So DAW's you know, you talked, this I would take from other people, but I can tell ya. So, you know, again, I used Sonic just because that was the thing. Basically Sonic's fade editing no one has ever come up with a better way to do it. Sequoia's editing, their fade editing is basically how Sonic worked. So, people that I've talked to, if they have access, they still seem to prefer Sequoia's editing, but really all the heavy hitters ultimately go to Pyramix. I'll tell you though at Sterling, they all use Nuendo for processing and then they dump it into Pyramix to do little stuff and then actually make disc masters. When they're going to iTunes delivery, that kind of stuff, I don't know if they even bother with Pyramix anymore. I don't know, what did Adam have to say about that? I'm curious.

K: Adam, they used Sonic Solutions for a long time at Gateway kind of fell out. So they ended up using Pyramix because they could adapt it to like Sonic Solutions did in terms of workflow, hot-key shortcuts, all the fade editing. That sort of stuff.

P: Yeah, that's right. Definitely for final editing and master generation, Pyramix is the closest, there's no doubt about it. The other thing is, most places that I know, Gateway and Sterling, they just made their own quick-key template and dumped it into Pyramix so it would just behave like Sonic and they didn't have to

re-learn all the quick-keys to be able to move fast. So I mean it just sounds like a dumb workflow thing, unless you work on an App all day, then you know, that's worth the time and investment. I gotta say that the iZotope tools have brought a whole new dimension, particularly for the non-heavy hitter studios. Even, I know one of the engineers at Sterling, I don't think even considered it. I mentioned it to them about a year ago and actually, Ue Nastasi, I talked to him a few months later and he was like, holy crap, I checked that thing out. You can make a record with that. So even guys like that are shocked to see what Izotope has been able to bring to the table with modern processing power and computers, you know what I mean? So that's it, that's kinda what I have to offer you. I hope that's enough but not too much.

K: You may be looking at an old version of the questions, but I have one last question. Do you have any advice for somebody who wants to start a career in mastering?

P: You know, this is where I don't know the business well enough right now. Of course, my advice would be to try to get in the door in some capacity at one of the major places that are left. For the same reason that I still think it's worth it for people who want to make records to do what they can to try to get in the door at one of the few remaining big studios. And a lot of it is just, in the beginning, put yourself in a situation where you're gonna be talking about techniques, technology, and communication with the artists all the time with people who are

doing it every day. You know, then you can take that and start your own place and I think these days with technology and digital delivery, what's changed is a person could, with minimal investment, you know, but I mean minimal investment... build a room in a house or in a space and really run a one or two person show. The trick is finding a dedicated client base. I think that can be done, but I think coming into it after some time, maybe a year, two years, five years, of being somewhere that you can just soak in it with other people, I think that's just really important because you could just learn so much more so much quicker. You know what I mean?

K: Yeah, that's how I feel about my time at UMass Lowell. We have a great community here where we can bounce ideas off each other and help on each other's sessions. Yeah, and I'm also doing an internship at M-Works so…

P: So J-W (Jonathan Wyner), I heard M-Works is closing or something?

K: Oh, that's kind of a taboo subject right now.

P: Ok, so anyway I was the chief engineer at a boutique multimedia mastering place when I moved back to Massachusetts called DVD Lab. We shared space with J-W and then ultimately the front space in 232 was a brick wall and we were in there for years. Anyway, Matt Azevedo is one of my closest friends on this planet. So did Matt teach the Advanced Acoustics class for you guys?

K: That wasn't offered.

P: Oh, I feel so bad for you!

K: Yeah, I know. But I was talking with Pat Drumm and he was telling me about it.

P: Yeah, it was funny for me because Matt and I worked together for years and then he was my teacher. I'm, you know, much older than he is. But getting to be a student of Matt Azevedo's for me was a highlight of my audio life. So, no one has more respect for… you know, anyway, I'm in the Matt Azevedo fan club so I'll shut up. Yeah so that's interesting, I don't know if it's news there, but I also know J-W really well. I did a couple records a month for five or six years in Jonathan's room. But just to tell you how this works, when I did my records, I would come in and run my own meters, I would re-wire the way the gear was set up, just like Matt would. You know, everybody has their own style. So his room is phenomenal, I mean it's a great place. My clients were really really happy with anything I ever did for them in that room. Alright, so there's more gossip, more depth to the gossip than what I heard. Ok, I should probably just call Jonathan.

K: I asked him about it and he said, I'll let you know more when I know more. You know, the intern is low man on the totem pole.

P: We all did our time man. So, you know, in the multi-track world, I quite honestly answered the phone and cleaned toilets for the first six months for minimum wage so I know exactly what you're talking about.

K: Well you're lucky you got minimum wage.

P: I was lucky at the time. Yeah it was good when the big rooms in New York were booming and it was right at the start of both when as colleges, at that point it was only UMass Lowell and the University of Miami, but the big studios were beginning to get that having kids who actually went through a program and came in armed with a little bit of knowledge of how microphones and consoles and signal flow worked, was a big deal. It definitely gave us at that time the edge of getting into the big rooms in New York. The thing is, in Yew York City at the time, they were starved for good second engineers because, this is gonna sound kinda funny but you know, no surprise, a lot of guys who wanted to do it thought the gig meant sitting around smoking pot and playing guitar, which… it was exactly the opposite of that. It was, you know, don't do drugs, be really into audio, and live here eighty five hours a week, and survive on the free fruit left by clients. That's what me and my friends did for the first five years we worked in New York City.

K: Yeah, I did an internship for this place called Brooklyn Recording and it was very similar. Like, I would eat the leftovers from the sessions and that was my dinner for most of the time.

P: Yeah, most of my closest friends at the time founded Studio G. I actually did the wiring, the design, and the install of the original Studio G. Yeah, Tony Mamoni and I were on tour together for years. I lived about three blocks from there in Williamsburg for billion years. Yeah, Ok, Karl, this was good. Don't hesitate to get in touch with me. Unless you have more OK. Ok, well thanks Karl, it's good to meet you.

K: Yeah good to meet you too! Thanks Paul, I really appreciate it.

P: Yeah, it's my pleasure. I hope I didn't talk too much. I'm famous for the sidebar, so…

K: No, no, I'm happy to get all your thoughts.

P: Yeah, absolutely, have a great weekend!

***** Paul promised to write out his thoughts and comments about the subjective language used in mastering. The following is a response recived via email on Saturday, March 25, 2016.

Punch: Funny enough I never really use this term; it's just not one that I ever picked up. However I would probably say, for instance, that a "punchy" snare has good lower-mid presence and solid attack. I'll add more mysterious descriptive words - punchy is something you feel in your chest/gut.

Warm: Very present but not bloated lower mid content. Maybe 160-180 Hz up to somewhere between 250-350. This is an area where many elements in a mix have spectral content - bass, sn, vox, gtr, horns, kbd, etc. For me warm means they all have that content well represented but don't step on each other. The total energy does not swamp the mix but is right there for the listener's ears to revel in.

Clear: Everything in a mix can speak. Even elements that are low in level within the mix can be easily focused on by the listener if she chooses and that element's unique details are all there. Achieved through the whole range of audio processes--mic placement, EQ, dynamics, etc.--during the record/mix/master process.

Harsh: Too much upper-mid content, typically can be anywhere between 1.6-1.8 KHz up to 2.7 KHz-ish. Might be the source element that had it, could be a recording/mix artifact, could be unwanted or badly chosen distortion artifacts.

Dynamic: Has a large dynamic range. Use of the term is program-dependent. Dynamic in a classical recording means a very different thing than dynamic in a Green Day record, for instance. But have no doubt that they can both be dynamic. I think of this as describing not just section-to-section behavior but also individual elements. It's most commonly thought of as being about the relative gain levels of

the material. But it can also very much involve how well performance intensity of the player was executed by the musician and handled by the audio team. Timbral differences can tell the listener so much about dynamics aside from the actual mix balances within a piece.

Musical/Groove: Ha - I had the good fortune to assist on recording sessions with many of the top session players in NYC back in the nineties. I also fought with many bad indie rock drummers about the difference between playing with bad time (which they thought was the same as having a groove) and actually playing with a groove when I was a producer/engineer in the same period <sigh>. Groove to me means that the players are all locked to the tempo, but then can choose to push or pull against that as they want. It's the feel of the rhythm on top of (within?) the actual meter of the tempo. Wow, that all really means something to me but reads very slippery, doesn't it? Can't wait to see what your findings are on this one!

Frequency Ranges: OK, here is where I am sure I do not line up with the academic lists. I know this from my years of teaching and pursuing my master's, ha ha. I know I was always consistent with it, though, because I used the same terms with my clients for 20 years and we always talked about it in the context of their music to make sure we agreed what we were discussing. I always put the "what do you mean by these words" discussion at the top of the list when kicking off a project with new artists.

- Lows - up to some where between 85-120 depending on the genre, the elements and the mix approach. Low mids - from there up to maybe

400-500 Hz. Mids - now up to 1.2 - 1.5 K, probably not as high as 1.5 K but again program-dependent for me. Upper Mids - up to 2.5K but could go slightly higher depending on genre, element, etc. Highs, everything above that.

V: Jonathan Wyner

Jonathan Wyner is a graduate of Vassar College and has since become a highly respected mastering engineer. Jonathan's career has spanned nearly three decades and continues at his mastering facility, M-Works Mastering. Jonathan also works for audio software company iZotope as the Director of Education and also is an instructor of Audio Engineering at Berklee College of Music in Boston, MA. A selection of Jonathan's credits includes Frank Zappa, Aimee Mann, and David Bowie. This interview was conducted at Atwood's Tavern in Cambridge, MA on Wednesday December 9th, 2015.

K: Could you tell me a bit about how you first got involved in mastering? Did you get into audio first as a musician or were you more interested in the technical side of audio? Did you go to school for audio? Did you intern or have an apprenticeship?

J: Well, I definitely didn't start out looking to be a mastering engineer. Nor, if I were at that point, knowing what I know now, if I went back, I probably wouldn't think I wanted to be a mastering engineer either. There was an opportunity that

came up at a particular point in time to work in a mastering room. And there were two things that made it very attractive to me, three things. One is the hours, I mean that wasn't really… having a job that whatever the hours the job were. The fact that being involved with digital recording at all at that time was enormously expensive and so it really was pretty much the province of a very high-end recording studio. This was even before DAT or PCMF-1 or any of those initial attempts at kind of, consumer recording formats that were pro-sumer that relied on consumer technology to make it more affordable. There was no such thing back then. So you know, a multi-track studio would have to invest, 200-250,000 dollars in a multi-track machine, which took it out of the ballpark for most studios. And then there were mastering facilities and the CD was suddenly a format, was a viable consumer format, so mastering facilities had to make that investment in two channels of digital audio, at least capture. It was a very new and kind of fragile technology, but it was inevitable in a mastering context. So, I got a job at a mastering studio and suddenly there I was dealing with this sort of newfangled machines figuring out signal flow, understanding clocking, coming into contact with concepts that most people weren't aware of. You know, it was very exciting and new and interesting. And digital editing was all new. And so I was intrigued. I thought, Oh Ok, I'll get to learn new stuff and exciting stuff. And there were benefits that came along with that, which was if you affiliate with a new popular technology, there are some artists that are interested in aligning themselves with it, and some of whom had fancy and famous names. So I came in contact with a catalog of work, or a body of work that I probably wouldn't have if

I was just doing a multi-track analog studio. The other thing that was advantageous that working at a mastering studio was at that time it was not trivial to release a CD. And so it was much cheaper to make vinyl. And so anything that I was mastering that was going to be released had to be release-quality in a sense that the artist had to be able to justify the investment that they were making. So the end result was the quality of the products I was working on went from doing a bunch of producing and recording and mixing on my own, and interspersed with the occasional high-quality project was something that was, you know, there was a lot of stuff that was not that good. At the mastering studio, the quality just suddenly went to like release-quality. Everything was better. It was much more satisfying artistically to come in contact with better records all the time. So, anyway, that's sort of what happened at the point of entry for me. And, you know, the seduction was the name of the artists I got to work with.

K: That was the Bowie catalog you did?

J: Bowie, Zappa, Paul McCartney's brother, Russ and Roland Kirk, you know, every genre… Ringo Starr, and Nirvana. Sub Pop records was an early client and they would just send me boxes of records to look at and it was kind of awesome. So that definitely kept me in the game. It was great ear training to have to listen to records that I was working for a guy called Toby Mountain, at the time. I'd listen to the records he was doing over and over, and listening to my own work over and over. I learned an immense amount in a very short period of time, I think. I mean,

it still took me a couple of years before I was ready to master my first record, and it took me another, honestly, it probably took me another ten to fifteen before I felt confident in what I was doing as a mastering engineer. I mean, I mastered hundreds or thousands of during that period but, in terms of feeling the facility and ease with the craft, it took me a very long time. And I don't know if that's true of everyone. I don't know how many mastering engineers would admit to it taking them that long. I remember the first session I did and I sat down in front of the console. I'd seen it done, I've heard it done, I've mixed records, whatever. I finally had my first attended session and I sat down in front of the console and I was totally panicked. Like, oh my god, what do I do? I knew how everything worked, but attaching the act of changing something to result and the impotence, I was really at sea for a while. And it went well enough. That client ended up doing five or six records with me over the next twenty years so I didn't end up screwing it up too bad but, it was kind of terrifying. You know, it was weird. I think anybody who remembers what the first mix was, or whatever, even though you understand the tools, the first attempt to do something, until you've integrated the act of observation and listening and assessment and sort of have that internal benchmark of what you hear and what you want to hear and how to get there, there's a lot of confusion and bewilderment. And that doesn't ever completely go away. But at least those periods last a much shorter time when you have experience.

K: One thing I've come to realize with anything like this is that it's sort of like learning a new language. For me, for example, Alex (Psaradukais) is letting me sit in on sessions and Nick is really helpful with showing me how the signal is routed, but until you actually immerse yourself in doing it…

J: It's context and what I call integration. It just takes time. I mean, it's that five thousand hours or ten thousand… how many years?

K: Ok, and did you grow up in a musical family? Is that how you got into the position of being hired at a mastering studio? You said you did mixing and producing before?

J: So yes, my grandfather's a composer, or was, and a conductor. My father was a pianist. My uncle was a prize-winning composer and pianist and yeah, I was completely surrounded… I mean, I probably spoke music better than I did English when I was three. And I grew up around Mozart and Bach and Yiddish art songs and Stravinsky, Michael Jackson and Jimi Hendrix and Sly Stone and this sort of incredible stew of genres. And actually that among other things is one of the things that I think set me up to be a mastering engineer. Because I see music as the language and the style is like an overlay. And so I was prepared for that for being able to understand how to connect style to intent or whatever. So I played music when I was a youngster. When I got into college, I went to Vassar. I ended up in a liberal arts program. I don't know all the reasons why I didn't go to a

conservatory, but I didn't. Probably because I was smarter than that. But liberal arts schools tend to be much stronger at fine arts than performing arts. Art history as opposed to painting. I meant there wasn't really a strong… there were just a few of us who were interested in performance and because there were so few of us, we had free reign of the school. And I had already, going into college, been interested in music and technology and pulling apart guitar amplifiers, mistreating instruments… And so I was immediately attracted to the radio station, became the program director. My campus job was recording all the faculty and student recitals. I had no idea which direction to point the microphone in, but I figured it out after a while. I knew how to thread the tape machine. They had an electronic music studio for composition that had an Electrocomp modular synth and a keyboarding synth and two 4-track machines. And I sort of put together this little production chain where I'd record stuff in the recital hall, mistreat it, and play it over the air. And I was in bliss but at that time there was no such thing as electronic production or music production as a discipline. You know, I was just kind of making it up. And I didn't realize until a few years after graduating that it was a thing you could do. And once I did, I started investigating it, went to a program in Ohio called the Recording Workshop.

K: Where in Ohio?

J: Chillicothe.

K: Ok, cool, I'm from Sandusky, up on Lake Erie.

J: Ok, yeah, I spent six weeks there (at the Recording Workshop), got totally turned on, they couldn't tear me away, and they offered me a job to stay and teach. And that's where I cut my teeth on understanding concepts, practices, did lots of sessions, produced my first album for a label out there. It was an eight-track half-inch recording for a band called, The Royal Crescent Mob and it came out on Celluloid Records and it was very charming. Not a great sounding record, but there are a couple of tracks on that record that are actually pretty awesome. I remember sending it out to mastering, and it came back and I was very disappointed because it didn't improve anything and I was surprised. The record just sounded just like my mixes! Ok, anyway, I worked for about four years professionally as a recording engineer and producer before I was offered the mastering job and that experience was invaluable in fostering my understanding about, and also my experience as a live performer. Understanding what went into the thing that I'm in charge with doing something about.

K: That's funny; I did a very similar thing in my undergrad. I recorded all the recitals and concerts and was in charge of making CD's and archives. Ok, anyway, the next question deals a bit with the technical aspect of mastering. **Are there any projects that stick out to you as having a particularly challenging aspect or technical difficulty that you had to overcome? What was the nature of the challenge and how did you react to it?**

J: Well… where do I start? The two that immediately come to mind, I worked on a record by Amy Mann called I'm With Stupid. I think it may have had eleven or twelve tracks on the record, eight of them were mixed by Jack Puig and produced by Jon Brion. To this day I think, by the way, that record is a study in guitar distortion, it's amazing. Anyway, he mixed the record in a studio where he had never mixed before and for whatever reason, he wasn't really referencing outside of the studio and he had set up the control room with all of his favorite outboard gear right behind him. And it was a considerable amount of outboard gear and so there was an edifice of outboard gear behind him on the producer's desk of his favorite compressors and EQ's, which served really well as an early-reflections generator. And so the mixes had a real lack of presence. They were really kind of murky. I got a call from Aimee and her manager, Michael Housman, and she said I need your help. She said, I don't care how much time it takes. So I spent three days investigating how to reverse engineer this early-reflection generator and pretty successfully rescued the record. The stakes were really high. She was doing really well in her career, one song had already been a hit on the record. So that's one that sorta sticks out to me. Another is the mastering of Madame Butterfly that was recorded by the BBC in 1912. It was recorded via gramophone and the way that they would record it was the transcription platters would only hold seven minutes of audio and they had two of them. Singers in the room, they would start one and about ninety seconds before the one stopped, they would start the other. And the frequency response across the platter changed dramatically. You would

loose a lot of low-frequency information as you got closer to the center and they were read from the outside in. They wanted me to stitch it together, do a lot of noise removal, which in and of itself was challenging because we didn't have the electronic transfers, an acoustic transfer, which meant that all the transients were rounded off and it was harder to do noise reduction.

K: So were you working from the originals?

J: They had done a capture. They lived in the national archive in New Zealand and they sent me DAT, no, I can't remember what the format was. It might have been DAT. And so I had to put it together so it sounded like a continuous performance, which meant doing a lot of clever processing in increments as you went across so you wouldn't detect a change in tone during the seven minutes, or five and a half minutes or whatever it was that you were listening to. Then on the transition of one to the beginning of another, that had to be seamless. That was enormously time-consuming.

K: So what kind of stuff did you have to do? I don't know if there was automation at the time.

J: Not good automation. This was 1989 or '90. So, a lot of real-time gradual changes and a lot of clever editing using these luxurious hundred-millisecond buffers that we had, which was pretty much all we had. I mean, we couldn't do

long crossfades. So, it was a lot of slight-of-hand with crossfading, and sometimes manual, sort of, crossmixing between synced sources and then editing the transitions.

K: Yeah, it sounds tough.

J: It was painful. It was tedious. You know, it was a challenge. It had its advantages. You know, back then, you got paid for every moment. And this was a project of historic importance and the client was willing to make the investment. Noise reduction was a time consuming and expensive undertaking.

K: Was that the Sonic Solutions No Noise system?

J: Yup. So I don't even remember what the invoice was, but they were willing to pay it. And they were willing to keep asking me for more, even beyond the point where I was interested. That's what it was like. I was like, lets stop now and they said, No no lets do more! Anyway, so those are two… The Russ and Roland Kirk compilation that I had to do was an interesting challenge because it was basically a re-issue of three recordings. One of which had been done on digital quarter-inch reel-to-reel. And there was an analog quarter-inch master and a vinyl that had was the only existing copy. They had lost all the other masters. And I had to take all three and master then onto a single compilation and that was another interesting challenge. I dunno, I could probably come up with a million others.

K: I'm curious, what was your solution to the early-reflection issue with the Aimee Mann stuff?

J: Well, it all came in on half-inch tape. I played around with the alignment on my Studer and also discovered that there was something about the sound of my Neve 2254 compressors without compressing, just the audio path. It really complimented what I was starting with. So that was the foundation of my solution. I kinda tried running the audio through everything I had. Using signal path changes for the color is something that mastering engineers do routinely, rather than relying on EQ or compression. It's about choosing the right path. Sometimes that has as much as, if not more to do with the result of the sound than what you do with the gear. So in this case, that was absolutely true. I didn't turn on the compression on the 2254. It's too slow and the noisefloor is right on the edge of being acceptable. For that record it was OK.

K: Ok, so next thing **I wanted to touch on is the topic of analog and digital**. I know you use both and I'm wondering if you could talk about the pros and cons of using a hybrid system rather than just working in the box.

J: Well, forget about in the box for a minute. I think there are really three variables here. One is hardwarde vs. software, another is analog signal path vs. digital. And back in the day when you did anything analog, it required an extra

A/D and D/A conversion and it's not until fairly recently that the penalty for doing that is acceptable. In most cases, I wouldn't take an orchestral recording and take it out into the analog domain unless I really wanted to fix something. So DSP-based processing has this obvious advantage of avoiding this conversion step. Hardware has the obvious advantage that you can reach out and touch a knob and turn away from a screen. Visual acuity, in my mind, doesn't play a large role in our aural perception. It can influence it for better or worse, but I would much rather… If I'm thinking about what I'm hearing I want to focus on what I'm hearing, not what I'm seeing. So the reason for my investment in hardware is partly because of that. And also, video displays are famous polluters of stereo field. The advantage of analog over digital? Until recently, digital EQ has gotten pretty good and we've gotten better and better at modeling harmonic distortion and the vagaries, time domain distortions of analog typologies and so on. I've been willing to accept digital EQ as a substitute or whatever for some number of years. Compression's a different story. There's such a complex interaction between the output of the device and the side-chain. And my subjective impression is that most DSP-based compressors did not do a very good job dealing with that complexity. So most digital compressors felt to me as if they caused the sound to get small. It would be bandwidth-limited, the image would shrink. I've also been given to understand that most digital compressors, either in the box or out, to date have been feed-forward and not really feedback. And a feedback compressor has a different kind of smoothing effect on the overall level because of the feedback loop. I think we're starting to turn the corner there but I'll

still investigate hardware compression. There have been some instances where I've done everything in the box lately. You know, and there are obvious advantages to that: recall, economy, etc. The early-reflection generators that hardware make, you know, they're not so cool. So if you can get all of the gear out of the room, you know, in somebody's world, the perfect mixing and mastering or anything studio is a room where you just have one knob and a pair of speakers and some 3-D glasses. So hardware gets in the way of that.

K: So is that a new product in development at iZotope? The 3-D glasses?

J: No, I wish it was. The other good thing about hardware is that if you invest wisely, it holds its value. Buy a compressor for $3,000.00 and you can at least sell it for $2,000.00. In some cases, you can sell it for $8,000.00. Plug-ins, you know, one of the biggest problems our world faces is, with certain kinds of things, given with the absence of scarcity, where a world where it's easy to copy and duplicate something, then how do we assess its value? You know, people used to go to the record store and try to buy the new Talking Heads record, maybe it was sold out. You can't sell out of a record now. It's always available on the Internet. If your server goes down, people go on to the next artist.

K: Ok, cool. **The next question is how much communication is there usually between you and the artist or producer in a typical mastering project**?

J: As much as possible. And I mean, obviously, the possible is an important variable. That doesn't really say anything but it's my way of saying that there needs to be some communication especially before and during the session. And without that, it's almost certain that you end up with a less-than satisfactory result and with it, you can surmount almost any obstacle because if you align your expectations and you're all working toward the same goal, with the same sort of filter, or some way of determining success. So, it's really important. While I have to trouble and no problem offering my assessment… Any mastering engineer that thinks they know better than the artist, at the end of the day is doing the artist and themselves a disservice. Not to say that artists don't make mistakes, they do. But If you come into a project with that approach, you're setting yourself up for a difficult situation, probably for an unhappy artist.

K: Right, they're the ones in charge, they're the ones paying the bills.

J: Well, they may have ideas that are different and who's to say whose are right? I mean, there are people who will send records from around the world and just say, please master my record. And do we do it, sure. Is it my favorite way of doing things or is it the thing that's going to lead to the most satisfying result?

K: Yeah, one of the things I was talking with Alex about was that he was working on a record with Greg (Calbi) and it was from Japan. He was saying it was like, super harsh. The high frequencies were really grating and Alex said that Greg

said, yep, that's how the people in Japan like it. They want it super harsh and aggressive. I would do something to make it less harsh but they want it that way so it's super important to know your client and their expectations. I also wonder how other cultures prefer their music to sound.

Ok, next question. **There are a lot of DAW's designed for mastering like Sequoia, Wavelab, and Pyramix. You obviously use Sequoia.**

J: At home I sometimes still use soundBlade because I don't have a PC.

K: So my question would be why Sequoia?

J: Familiarity, ergonomics, the feature set is complete enough to do most of the stuff that I need to do. It has superior editing… those are the primary reasons. And there are certain things Soundblade does that are better than almost every other program. Maybe with the exception of SADiE. I think Pyramix is kinda configurable, kinda the same way. There are certain things you can't do. And their crossfades and their fade shapes and the playback engine are just kind of awesome. So yeah, I guess those are the reasons.

K: Did you test out several different DAWs before you decided on Sequoia?

J: Until recently, there were not so many DAWs that would directly output DDP's, or that would allow you to do four-point editing. In the case of Sequoia,

when I first started using it, there were occasions where I would output surround test discs for clients and you can do it directly from the DAW. The real win with that DAW is that when we started using plug-ins, having the ability to put plug-ins on clips and crossfades between plug-in states, is just like a total win. You no longer have to automate anything. It's a time-saver and a right-brain saver. I'm intrigued by Studio 1. It seems like an interesting production tool that has some decent mastering features.

K: I'm not familiar with Studio 1.

J: It's the Presonus DAW. There are many more mastering DAW's. Some of which implemented pretty good feature sets except it couldn't address external hardware. I guess I'm not willing to invest in a totally closed system. So that's why I gravitated to those. And once you buy into something and you start to generate a body of work and archives in that platform, you're married to it. And as long as the manufacturers don't do anything totally stupid, which Sonic Solutions did, but you're gonna stick to it. It's kinda like marriage. You don't get divorced unless somebody does something really stupid.

K: Yeah, I'm still running Protools 9. I have an old computer and haven't updated. I'm sticking with it.

J: You'll change. At some point you will.

K: Yeah, at some point but it works just fine for me now.

J: Yeah but if you're doing a lot of MIDI, it's not 64-bit and you'll run out of virtual instruments. You know, you can only address four gigs of RAM. Even if you have thirty two, you can only address four.

K: But I'm not doing much on my own rig anymore. I'm mostly using school's computers and it's working just fine. Ok, cool. **My next question is do you have any advice for somebody who wants to start a career in mastering?**

J: Well, if you're hell-bent into getting into mastering, then I would suggest you do one of two things. Get a real assistant position at a real facility, working for somebody who has a name, and spend at least a couple of years hanging out and learning and absorbing what you can. Or the other is spend a stupid amount of money equipping a room that satisfies the minimum requirements and get really really good at social media and executing a business plan and networking and have some kind of buffer for a few years between yourself and the bill collector and give it a go. If you wanted to pursue it as a specialty, a singular discipline, I don't think you can mess around with doing a lot of things at once. You have to completely steep yourself in whatever that specialty is. I'm not sure I'd advise it, but I think that that is true. I think that if you're going to incorporate it into a collection of services you're looking to do, you don't necessarily be that good at

it. You know, you can mix projects and offer to compile them for artists. You don't usually have to rise to the same level. You know, then it's sort of like, the mix engineers who say to me, I would've loved to have sent you the record but my client insisted and they pulled out $250.00 and said, master this now, and so I did. So they got what they got and I got $250.00 and off we go. There are certainly those shortcuts. There's a real big question in my mind whether anybody, the way things look right now, can make a long-term career as a mastering engineer in the way that we used to think about somebody being a mastering engineer. You know, you can apply some polish and you can output files and do all of that but, you know, there aren't enough record labels, there aren't enough repeat clients that can really support working at a really high level. I mean I could go on but I guess those would be my pieces of advice. Yeah, so being detail oriented, being very humble, and I don't mean humble in the Christian sense. I mean humble in the same way that, you know, interns will succeed if they don't offer opinions if they're not asked for. That's doubly important in the context of mastering. If you walk into a room with Bob Ludwig or Greg Calbi or anybody and start telling them about how you once mastered a record like they did, sorry you're done.

K: Yeah, I mean that's just rude.

J: It's not rude, it exhibits a lack of understanding and also a closed mind. Just being in the room, just checking out what's going on, listening, seeing if you

notice tendencies and patterns and habits. And then when the invitation comes to ask questions, ask questions. But to offer assessment or opinions is, in that situation where you're learning, you're acquiring context and knowledge, I think it just gets in the way for everybody. I just see that, people make that mistake over and over again where they'll go in and sort of want to seem like they know what's going on and be everybody's best friend.

K: They probably want to sound like they know what they're talking about.

J: Yeah, it's not born of ill will, they just want to seem like they're part of the club but they want to sort of play that game. But really, that ultimately doesn't contribute to anything, to the relationship or to the learning.

Mastering Subjective Language

Punch:

J: Well I think what they mean is reducing the peak-to-average ratio to the point where you get enough thrust from the transients speaking more in the midrange and less in the top end.

Warm: Yeah, it depends on what instrument they play.

K: Lets say it's a pop or rock record.

J: Who's asking for it?

K: The producer.

J: Well, then forget it. Usually they mean not too much top end, probably not too much deep bass, I don't know. Really, from a bass player, they're usually talking about early harmonics, vocalists they're usually talking about not too much midrange. Some people may think it means depth of field, maintaining ambience or emphasizing ambiance. But I think that it doesn't have any, there's no intrinsic singular meaning to the word, except to say an emphasis on the low-midrange and less on the high end and less on the presence. But that's, you know, just hard to distill a single meaning.

Clear:

J: Clear means where you have, you don't have collisions.

K: Collisions in terms of…

J: Yes. In terms of arrangement, in terms of energy buildup in the spectrum. You know, it says everything about proportion, and a lack of masking. That's probably the closest I can get.

Harsh:

J: The opposite! No, harsh is too much emphasis on either odd order or higher harmonics and a lack of support in the fundamentals. But again, when some people say thin, some people mean harsh. And, you know, maybe there are some alt-punk records that want to be harsh. You brought up the Japanese artists, so yeah, is harsh good or bad? I don't know. Well, we say it and we think it's undesirable. That's really culturally chauvinistic.

Groove:

J: Oh yeah. Groove is where you get enough of the 2 and the 4 and not just the 1 and the 3.

K: Yeah, I'm having trouble understanding that one myself.

J: Well, it's the thing that the stylus sits in when you plan an LP. You know, groovy, the expression came from when they were cutting live to vinyl. If the got a good take, they would say, well, it's groovy, it's all in the grooves.

K: Oh, cool, I didn't know that.

Frequency Ranges: Lows, Low Mids, Mids, High Mids, Highs.

J: Again, it depends who I'm talking to. If I'm left to my own devices I think, you know, low end is 100hz and down. Low mid is 180 to 350, maybe 400. Midrange is 500 to 1.8k, maybe 2k. High-mid: 2k up to about 3.8. Upper mid, that sort of presence where you get up into the high end where there's a sort of strata in the high end. One is like 4 to 6k, another is 6 to 9k, and then you have a little overlap, 8 to 10k and then up into air.

K: Are there certain characteristics that you would attribute to each frequency range?

J: I don't think a frequency has a characteristic, I think it's the relationship and proportion of frequencies and one band to the other depending on the particular instrument. Where proportionality and character comes into play. And I almost have this religious superstitious belief that there is something, such a thing as as the proper proportion and when something feels like it's in the proper proportion, there's an ease to the experience of listening to it that just kind of happens and until that happens, I feel uneasy. So sometimes I feel that what we do is audio-chiropractic, it's kind of an alignment of the frequencies, which is one of the reasons why I think that time domain is as interesting and important to consider in what we do as any kind of frequency adjustment. So I think whether we're talking about groove, or psychoacoustic impression of the results of our signal processing chain, changes in timing relationships defines a lot of what we hear and why we like what we like.

Discussion: A Summary of the Responses

Part 1: Specific Questions about Mastering

Question 1: Background and Finding a Career in Mastering

Jay: Jay was interested in music and electronics as a child. He went to Berklee College of Music for piano performance, but was also interested in the technical side of music recording. He then interned with Jonathan Wyner at M-Works Mastering. After the internship, Jay moved to Atlanta, GA and worked at a post-production facility where he began to build his own mastering studio, investing in Sonic Solution's unique and, at the time, rather exclusive No-Noise System. After a few years in Atlanta, Jay moved his mastering business to Los Angeles, CA and later relocated to his current facility in Brookline, NH.

Matt: Matt graduated from the University of Massachusetts Lowell and during his time there he honed his skills in the use of DAWs and other pertinent audio disciplines. He then interned at M-Works Mastering where he did editing, PQ, QC, and eventually became a full-time mastering engineer. He then described the state of mastering during the early and mid-2000's. He also talked about how and why he left mastering to pursue acoustical consulting.

Adam: Adam grew up playing electric bass and is also a graduate of UMass Lowell's SRT program and interned at M-Works. He was at M-Works for a time and then moved to Gateway Mastering in Portland, ME. He soon found himself assisting famed mastering engineer, Bob Ludwig and eventually was promoted to full-time mastering engineer.

Paul: Paul is one of the first graduates of the SRT program at UMass Lowell. He established himself as a prominent figure in New York City's budding indie-rock scene in the early 1990's. After working at recording studios for a time, he began working at Sterling Sound and had his own client-base among the indie rockers as a mastering engineer. He also was the lead assistant to Ted Jensen. He currently works at Bose Corporation.

Jonathan: Jonathan grew up in a musical family and his interest in music recording and technology was spurred when he attended Vassar College and began recording the concerts and recitals there. He then traveled to Chillicothe, OH and attended the Recording Workshop where he was quickly hired on as an instructor. Jonathan was then offered a job at a mastering studio called, Northeastern Digital, where he was one of the first mastering engineers to embrace digital technology. It was because of this that he was able to work with high-profile artists. Jonathan eventually founded his own mastering studio, M-Works Mastering, where he continues to work today.

Question 2: Evolution of Mastering

Jay: Jay had a good sense of what he wanted to do in mastering, probably from a musical upbringing and critical listening experience. He mentioned that it's also important to be diverse, being able to master any genre.

Matt: At first it was easy for him to put more gear in the signal path if there was a lot wrong with it. He then realized that the saying, "less is more" applies to mastering. Now he uses three pieces of gear for 95% of his work: the Sontec EQ, Requisite compressor, and a digital limiter.

Adam: Adam pointed out that each mix the mastering engineer receives has been approved by the client, most likely the artist, producer, and mix engineer. His job is to take that mix and make it sound "better," if he can without destroying the integrity of the mix. He then goes on to say that when he first started, he didn't take as many liberties in the mastering process as he does now. He says, "Don't be afraid to go for it."

Paul: Something Paul learned from Ted Jensen, was to be able to listen to the raw mix and then know what exactly what needed to be done in the mastering process.

Question 3: Technical Challenges

Jay: Jay discussed dealing with 5.1 surround sound and how to process each channel. Also, low-frequency problems that come in mixes from small studios. He also discussed the problems with multiband compression.

Matt: Matt discussed one project with an avant-garde musician who wanted to cut his record to vinyl. The problem was that the recording had a lot of stereo bass information and it was impossible to cut to vinyl without complication. Matt described what he did to fix the problem, which was to highpass one of the stereo channels and then use a frequency-specific stereo control plug-in to move the bass frequencies of the other channel to the center. One tip Matt includes is if there's a generally "muddy" low end in a mix, put a highpass filter on the side channel using an M/S control.

Adam: There were two very large DVD projects, a Joe Satriani project and a video for the Rock and Roll Hall of Fame. Both were many hours in length and diverse in scope. These projects forced him to find the most efficient and effective means to master the music. He discovered the effectiveness of current digital technology via plug-ins. This allowed him to efficiently change settings and automation to best fit the diverse musical material.

Paul: The biggest challenge he mentioned was about communication with the artist.

Jonathan: There were two projects that were challenging. The first was an Aimee Mann record called, *I'm With Stupid*. The mixing engineer worked with a reflective surface behind him so there were strange effects in the mixes because of the early reflections. The second was the compilation and mastering of a 1912 BBC recording of Madame Butterfly, which was recorded via gramophone.

Question 4: Analog, Digital, or Hybrid?

Jay: "Absolutely hybrid." Jay typically uses analog EQ and compression and then digital EQ and limiting. Sometimes he EQ's the file before going to analog if there are timbral problems with the material. Jay also provided the tip, to avoid too much of a strange artifact of a piece of gear or plug-in, use a few different pieces of gear or plug-ins to do the same amount of work. "Spread out the pain."

Matt: Hybrid: "I like tools of both flavors." He prefers using knobs and other manual controls to master because it allows him to look away from a computer screen and focus more on the music. He also discussed the benefits of the effects of distortion through incorporating tubes. It can add a pleasing character to the music.

Adam: Almost always uses digital and he discovered its effectiveness while working on those two large concert videos. He "became less and less happy with the results [he] was getting from [his] analog signal path." He also pointed out that there is inherent phase-shift while using analog gear and also that plug-in developers are becoming better at modeling the distortion characteristics of analog gear.

Paul: He used a hybrid system, primarily using a Sontec EQ, Focusrite 200 series, and Sonic Solutions.

Jonathan: Jonathan uses a hybrid signal chain. Recently digital EQ has gotten much better than it has been in the recent past. However, he feels that digital compressors haven't matched analog yet. He also stresses the importance of a high-quality converter.

Question 5: Communication

Jay: Sometimes there are revisions and if they're small, he will do them with no charge. However, it is best to be straightforward with the client and let them know that they can't have too many revisions just for the sake of revisions. Also, sometimes he will send the client a reference master to see if they like the direction in which he's going.

Matt: "Usually a lot." He usually masters the record and then does revisions. His revision policy is "revisions within reason." He also said that he does not like doing shootouts.

Adam: It's different from project to project and he mostly communicates with the producer or mix engineer. He also usually provides the clients with a reference master.

Paul: 99.5% of his sessions at Sterling were attended by the clients so the communication was immediate.

Jonathan: "As much as possible." He also says to make sure that everybody's expectations are in alignment.

Question 6: Choice of DAWs

Jay: Gives a brief history of mastering-specific workstations like Sonic Solutions. Jay's personal preference is soundBlade because it is a derivative of Sonic Solutions and it is also Mac compatible. He also mentions that Sequoia, Pyramix, and SADiE are the other top DAWs for mastering.

Matt: Pyramix and Sequoia are the best DAWs for mastering, but both are only Windows.

Adam: He discusses some of the history and development of mastering-specific DAWs, which evolved from the use of Sonic Solutions to the use of many other DAWs, each of which was good at one aspect of mastering and not necessarily anything else. They eventually settled with Pyramix because they were able to model a lot of the same editing techniques and hot-keys as Sonic Solutions. Pyramix is also able to do all the necessary mastering-related tasks such as hi-res audio, DSD, surround, and outputting final masters.

Paul: While at Sterling, he used Sonic Solutions, but now he says that the guys at Sterling use Nuendo for effects processing and Sequoia for the creation of the final master.

Jonathan: M-Works is set up to use Sequoia because of "familiarity, ergonomics, the feature set is complete enough to do most of the stuff that [he] need[s] to do. It has superior editing…" He also uses soundBlade at home sometimes because it works on Mac.

Question 7: Advice for Young Mastering Engineers

Jay: Become the guy everybody wants to work with. You can do this by working hard, staying late, and being attentive. Also, try to be an asset to the company by providing your own clients, another source of income for them.

Matt: The best way is to find an assistant job with an established mastering engineer. Other than that, start small and work your way up. The most important element in mastering is the monitor path.

Adam: "I think the best advice I could give you is to try to find a good mentor." Also he says to be prepared for whatever opportunities come your way. He also points out the importance of being in a community where audio and mastering is discussed on a daily basis.

Paul: Try to get your foot in the door at an established mastering studio and be in a situation where mastering and audio are discussed daily.

Jonathan: There are two ways to get into mastering. The first is to get an assistant job at an established mastering studio and gain experience and contacts. The second is to set up your own studio and become adept at social media and self-promotion.

Part 2: Mastering Subjective Language

Punch:

Jay: Midbass frequencies between 90hz and 140Hz.

Matt: "Clear and well defined transients."

Adam: Good and tight low end response, specifically with the kick drum and bass guitar in a pop or rock situation.

Jonathan: "Well I think what they mean is reducing the peak-to-average ratio to the point where you get enough thrust from the transients speaking more in the midrange and less in the top end."

Warm:

Jay: Could be frequency ranges or a product of distortion. Usually exists around 400Hz.

Matt: "Appropriate balance between low-mids and upper mids/highs, favoring the low mids."

Adam: Similar to punch in that there is good low-end response and relationship between the kick drum and bass. But the instruments might not be as "glued"

together. Warm also means that there is not much brightness in the recording. You cannot have a warm and bright recording.

Jonathan: In a pop or rock setting: “Usually they mean not too much top end, probably not too much deep bass.”

Clear:

Jay: There are a lot of elements to clarity a few being, panning (making the best use of the sound stage), musical arrangement, and even and balanced frequency response throughout the entire spectrum.

Matt: “Appropriate balance between low-mids and upper mids/highs, favoring the upper mids/highs. Ideally a perfect balance is both "warm" and "clear".”

Adam: Deals with the detail among individual instruments in terms of frequency. Recordings with clarity have a good amount of upper harmonic presence, upper midrange to “air.”

Jonathan: A lack of collisions in terms of musical arrangement, spectrum, masking, and proportion.

Harsh:

Jay: Usually around 2.5kHz, but anything in the sensitive spots of the Fletcher-Munsen curves.

Matt: "Excessive upper-mids / highs, particularly in the sibilant region."

Adam: "Harsh usually means upper midrange, too bright, unmusical, and a listening experience that is not pleasant at all." This involves especially the 2-4kHz range and a lack of lows and low-mids.

Jonathan: "Harsh is too much emphasis on either odd order or higher harmonics and a lack of support in the fundamentals."

Groove:

Jay: Creating groove in mastering through the use of compression to create a bit of ebb and flow in the dynamics.

Matt: "What a funk band does."

Adam: Similar to warmth and punch in that they are about the low-end response and "glue" between the kick and bass. "Maybe it's more glued and tight than warmth is, but not quite as much as punch."

Jonathan: "Groove is where you get enough of the 2 and the 4 and not just the 1 and the 3."

Frequency Ranges:

Jay: Lows: anything below 100Hz. **Low-mids**: 100Hz to 600Hz. **Mids**: 600Hz to 2.5kHz. **High-mids:** 2.5kHz to 7kHz. **Highs:** 6kHz and above.

Matt: Lows: 0 to 80 Hz. **Low-mids:** 80Hz to 320Hz. **Mids:** 320Hz to 1240Hz. **High-mid:** 1240Hz to 4880Hz. **Highs:** 4880Hz and above, basically, two octaves each.

Adam: Lows: everything below 80 or 100Hz. **Low-mids:** 100-500Hz. **Mids:** 500Hz- 1.5 or 1.6kHz. **High-mids:** 1.5 to 8kHz. **Highs:** above 8kHz.

Jonathan: Lows: 100Hz and below. **Low-mids:** 180-400Hz. **Mids:** 500Hz-3.8kHz. **High-mids:** 4-9kHz. **Highs:** 9kHz and above.

Conclusions

These interviews are excellent insights into the world of mastering. Each mastering engineer offered a diverse array of responses by providing real-world insight and intelligent responses to the questions. From these responses, we are able to pick out some trends that emerge as common practices in mastering. However, it is essential to note that there are pros and cons for every response and it is equally important to remember that whichever method the mastering engineer prefers, there is always a reason behind their choice.

Part 1: Specific Questions About Mastering

Question 1: Background and Finding a Career in Mastering

Several interesting trends emerge from the responses of the mastering engineers to this question. The first is that they all went to school for audio in some regard. Everybody except Jonathan Wyner and Jay Frigoletto went to school for specifically audio, and even more specifically, to UMass Lowell's Sound Recording Technology program, though this is a matter of coincidence due to regional proximity.

It is also noteworthy that all of the mastering engineers began their careers either interning or working at a mastering studio. This coincides with the final

question because most of the engineers said that the best way to start a career in mastering is by trying to work at an established mastering studio.

One last thing to mention is the importance of being an asset to the studio where one is working. It is great to be an assistant, but it is better to be an assistant and bring in your own clients that can be another income source for the studio.

Question 2: Evolution of Mastering

This question offered a diverse array of responses. Some of which were bits of practical wisdom like being able to listen and assess a mix for the things the mastering engineer needs to do. Other responses discussed musical background and knowing how much was enough in terms of processing, offering the idiom, “less is more.”

Question 3: Technical Challenges

Question 3 also offered a variety of responses from the mastering engineers. These challenges ranged from processing historical and primitive recordings; the challenges of cutting vinyl; communication with the artist; and the intricacies of mastering 5.1 surround sound. Perhaps this range of responses is because the mastering engineer must be prepared to encounter any challenge. It is also crucial for the mastering engineer to have all the tools necessary for fixing

problems. These tools can include common devices like EQ and compression, but might need more specific and detailed tools like, noise reduction, automation, and tape machines, to name a few.

Question 4: Analog, Digital, or Hybrid?

When it comes to signal path, there was a very clear preference of a hybrid system for mastering. One of the engineers, Adam Ayan, mostly preferred a digital signal path and none of the engineers said that they use an entirely analog signal path. Though one can imagine that an entirely analog signal path could happen from time to time under the right circumstances.

The engineers who preferred the hybrid signal path said that they could pick and choose of the best equipment of each flavor (digital or analog). One thing they stressed was the purity of the signal path, especially quality of the Digital/Analog and Analog/Digital conversion. Their reason was that if they go out to an analog signal path, they want the conversion to be the best, most precise conversion in order to maintain the integrity of the signal.

Question 5: Communication

It was apparent that there is a correlation between the amount of communication and the satisfaction of the client; the more communication, the better the result, the happier the client. This would include finding out how the client feels about the mixes. Should the mastering engineer do a lot to change the mixes or are they generally happy with the mixes? Are there any reference recordings that the client suggests so that the engineer can try to match the references in terms of loudness and overall timbre and vibe?

Sometimes the mastering engineer will master a few songs and send it to the client for their impressions. Once the client approves the mastering, the engineer will create a final master, which will be checked for flaws like inappropriate distortion and then sent out for distribution.

Question 6: Choice of DAWs

Within the world of Digital Audio Workstations for mastering, there seems to be a common preference for Sequoia and Pyramix. These are the two most widely accepted DAWs for mastering. Though if there is a mastering engineer that prefers to use Apple equipment, soundBlade is the most preferred for that operating system.

The mastering engineers also discuss the use of Sonic Solutions as the first operational digital system for mastering. Unfortunately it died out after the

company decided to change its focus from mastering to the creation of DVDs. This provided a chance for other companies to make DAWs for mastering and this is when Sequoia and Pyramix emerged as the best mastering DAWs available.

Question 7: Advice for Young Mastering Engineers

The most common advice given by the engineers is to find an assistant position at an established mastering studio and also try to get into a mentor/apprentice relationship. We see this especially in the career path of Adam Ayan, who assisted Bob Ludwig for years and later became a full-fledged mastering engineer.

Some other advice offered by the mastering engineers was to be a reliable person. This best achieved by staying late, offering assistance whenever needed, and being generally knowledgeable and helpful. Becoming a successful mastering engineer requires a combination of preparedness and luck: being ready for any opportunity that arises.

Part 2: Mastering Subjective Language

Punch: Most of the mastering engineers stated that punch is about the transient response of low and low-mid frequencies, generally referring to the kick drum and bass guitar in a pop or rock setting.

Warm: The general response to warmth is similar to punch in that it is referring to the frequency response of low-mids. This is often in conjunction with a lack of high frequencies.

Clear: This refers to a combination of a few elements in the mix: evenness of frequency response across the spectrum; thoughtful execution of the musical arrangement; the clever use of panning within the mix; and an appropriate amount of upper harmonic presence in the "air" region of the high frequencies.

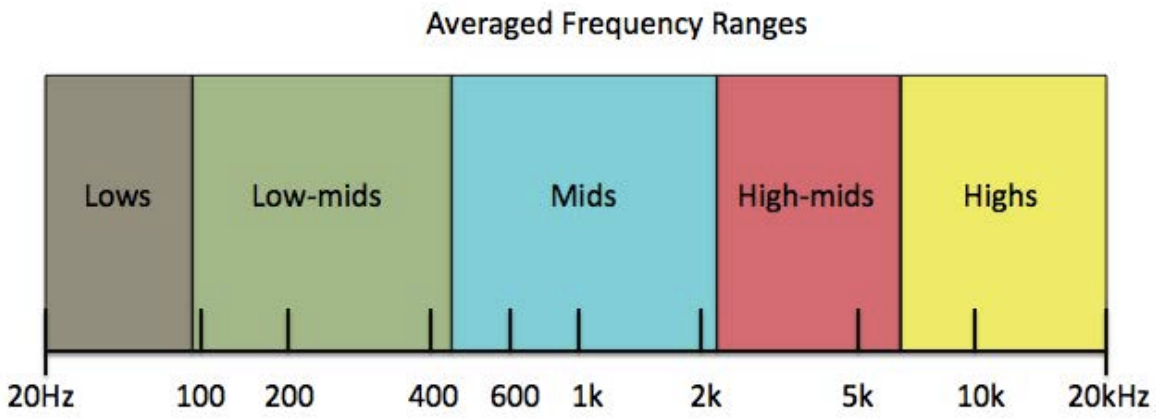
Harsh: Harsh involves an abundance of upper-mid frequencies, which includes the 2-4kHz region where the ear is most sensitive. This also means that there is a lack of low-frequency fundamentals supporting the upper harmonics.

Groove: This term offers the most varied response because it is quite abstract in the scope of terms used by musicians. It is generally accepted that groove is about the cyclic dynamic response of a song found mostly in the drums and bass guitar, the rhythm section of a band.

Frequency Ranges: Some of the mastering engineers offered responses of octave ranges. This is important to remember while assessing spectral characteristics in order to fairly account for the frequency ranges. The responses to frequency

ranges were quite varied. The following is an average of the responses from the mastering engineers.

Lows: anything below 92.5Hz. **Low-mids:** 92.5 to 455Hz. **Mids:** 455 to 2.285kHz. **High-mids:** 2.285 to 7.22kHz. **Highs:** 7.22kHz and above.



Recommendations

This section of the project examined several aspects of mastering. The interviews provided invaluable insight into contemporary mastering techniques from people who have established themselves as authorities in mastering. The following is a discussion of some areas of further potential study.

- **Sample Size:** The mastering engineers that were interviewed in this section were all located within the greater Boston area. More specifically, the majority of the interviewees had a connection to UMass Lowell's Sound Recording Technology program. This study would benefit from a larger sample size in order to confirm the common trends in mastering.
- **More Questions:** The questions asked were designed to be rather general and to allow the mastering engineer to go into any specifics if they desired. Some additional, specific, questions could have been asked regarding the details of mastering. For example, it would have been interesting to get their thoughts about mastering for surround sound.

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SECTION 2: PRODUCTION OF "ALL THE SADNESS"

I. Introduction and II. Methodology

This second portion of the thesis focuses on the recording, mixing, and mastering of the song, "All the Sadness", by Alan Williams' group, Birdsong at morning. The elements of this section are twofold: the mastering of four different mixes of "All the Sadness" and an EP match exercise involving this song and two previously recorded Birdsong at Morning songs. In this part of the study, "All the Sadness" was recorded, and sent to four different mixing engineers, Alan Williams, Nick Dragoni, Brandon Vaccaro, and Bradford Swanson. The four mixes were then mastered as if they were each a "single": one song that stnds alone as a complete musical work.

The second part of this study was an EP match exercise. In this section, "All the Sadness" was incorporated with two other Birdsong at Morning songs: "Murderous Friend" and "Down in the Hole." These two additional songs were a part of Birdsong at Morning's 2015 release, *A Slight Departure*, which was mixed by Alan Williams and Mastered by Adam Ayan. This exercise attempted to match the overall aesthetic of "All the Sadness" with that of "Murderous Friend" and "Down in the Hole" in order to create a cohesive short, three song EP.

The Recording Process

The recording project of this thesis was divided into four sessions. All of the sessions were conducted at the University of Massachusetts Lowell's Sound Recording Technology studios, rooms 114 and 213. The song was recorded using Protools with a sample rate of 96kHz and a word length of 24 bits via Avid HDIO converters.

Our goal for the first session was to record the acoustic guitar parts, played by Alan Williams. These were essential to record first as the guitar parts were the steady elements that continued throughout the entirety of the song; the foundation for this particular piece of music. We also used this time to record a rough vocal track.

The guitar used in the recording was a carbon fiber Emerald Guitars X20 and two microphone techniques were employed to record the guitar. The first was a pair of AKG "The Tube" microphones. We set up this Blumlein pair about ten inches in front of the guitar with the center pointing toward the joint of the neck and the body of the guitar. This technique was used because these microphones are known for their "warm" characteristic, placing emphasis on low-mid frequencies and featuring slight harmonic distortion.

The second microphone technique was a spaced pair of Schoeps CMC6's. One of the microphones was about six inches away from the guitar pointing at the bridge and the other pointing at the 12th fret of the guitar. Alan suggested this arrangement, being one of his favorite microphone techniques.

Following the recording of the main acoustic guitar part, we recorded with a pair of acoustic guitars, a Taylor and a Martin 5-15 parlor guitar, as auxiliary overdubs. We recorded these two guitars using both a Sure KSM-44 and an Audio-Technica 4060.

Unfortunately, this first session was marred by technical difficulties. Even with the quality assistantship of fellow SRT masters student, Pat Drumm, the overdub process was difficult. We were accustomed to using the cue sends in a full-band tracking session where one instrument occupies one send the entire time, however; overdubs were difficult to manage using this system because we needed to layer tracks and organize those layers differently as the session progressed. The solution to this problem, of which we employed for the rest of the sessions, was to matrix the sends within Protools. We placed an aux send on individual audio tracks. We then routed the aux send to an aux track. Finally, we assigned the aux track to an I/O send which went to the headphone mixer. With this method, the cue sends can be routed quickly and easily within Protools. For illustrations and more details of Session 1's set-up, please see the following **Figures 6 and 7**.

1

Microphone Input/Assignment Sheet

Sound Recording Technology
University of Massachusetts Lowell
35 WILDER STREET SUITE 3
LOWELL, MASS 01854
(978) 934-3850

Date: 11-16-15 Page: 1 of 1
Studio/Machine: 114, 213
Course/Lab: Masters Rec. Project
Eng/Assist: Karl Fleck, Pat Drumm
Artist: Birdsong at Morning

Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note	Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note
1	Handy 1	1	Acu GTR 1	The Tube	Blumlein	25					
2	Handy 2	2	ACUGTR 1	The Tube	Blumlein	26					
3	API 3	3	ACUGTR2	Schops CMC6	Bridge	27					
4	API 4	4	ACUGTR2	Schops CMC6	Neck Joint	28					
5	Handy 3	5	ACUGTR Dbs	KSM44	Neck Joint	29					
6	Handy 4	6	ACUGTR Dbs	AT4080	Neck Joint	30					
7	API 5	7	Scratch Vox	Beck 195	Scratch Vox	31					
8						32					
9						33					
10						34					
11						35					
12						36					
13						37					
14						38					
15						39					
16						40					
17						41					
18						42					
19						43					
20						44					
21						45					
22						46					
23						47					
24						48					

Notes: _____
Crosses: Main Room Ch _____ Hallway Ch _____
Iso Booth Ch _____ Amp Closet Ch _____

Headphone Mix Notes: _____
1 - _____ 3 - _____ ST - _____
2 - _____ 4 - _____

Figure 6. Session 1 Input List

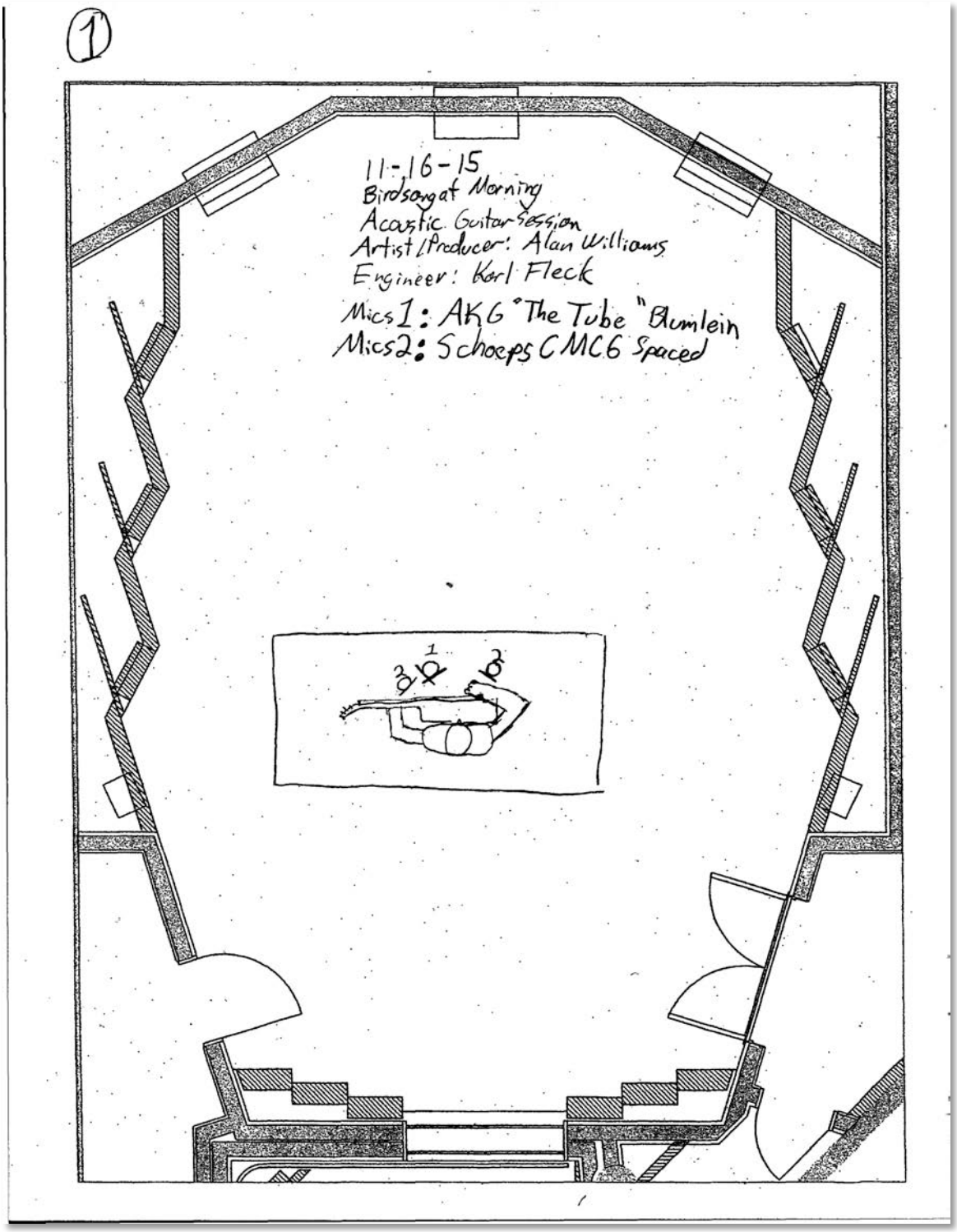

Figure 7. Session 1 Floor Plan and Mic Placement

During the second recording session, assisted by Ethan King, we focused on the vocals and electric bass. We recorded the vocals with two microphones, an AKG “The Tube” and a Bock 195. Both microphones offered different “flavors” as “The Tube” features slight distortion while the Bock is not as distorted and provides a slight bump in frequency response at around 10kHz. The reason for recording with two microphones is that the mixing engineer more flexibility with the character of the vocals.

We recorded the bass primarily with a Radial J-48 DI. One fun experiment we conducted was acoustically recording the bass guitar. This instrument was a 1960’s Harmony hollow-body bass and featured F-holes. This provided the instrument with a very unique acoustic characteristic, which was then recorded with a Neumann 140.

At the end of the second session, there was just enough time to add a piano part. The piano was recorded with a pair of DPA 4006 omni-directional microphones. These mics offered a clean and precise sound, working perfectly for our purposes. Again, for illustrations and more details of Session 2’s set-up, please see the following **Figures 8 and 9**.

(2)

Microphone Input/Assignment Sheet

Sound Recording Technology
University of Massachusetts Lowell
35 WILDER STREET SUITE 3
LOWELL, MASS 01854
(978) 934-3850

Date: 11-20-15 Page: 1 of 1
Studio/Machine: 114, 213
Course/Lab: Masters Rec. Project
Eng/Assist: Karl Fleck, Ethan King
Artist: Birdsong at Morning

Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note	Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note
1	Hardy 1	1	Vox	The Tube		25					
2	Hardy 2	2	Vox	Beck 145		26					
3	API 1	3	Bass DI	Rod. J48	DI	27					
4	API 2	4	Bass	140	Mic at F-hole	28					
5	Millenia 1	5	Piano L	DPA 4006		29					
6	Millenia 2	6	Piano R	DPA 4006		30					
7						31					
8						32					
9						33					
10						34					
11						35					
12						36					
13						37					
14						38					
15						39					
16						40					
17						41					
18						42					
19						43					
20						44					
21						45					
22						46					
23						47					
24						48					

Notes: _____

Crosses: Main Room Ch _____ Hallway Ch _____
Iso Booth Ch _____ Amp Closet Ch _____

Headphone Mix Notes: _____
1 - _____ 3 - _____ ST - _____
2 - _____ 4 - _____

Figure 8. Session 2 Input List

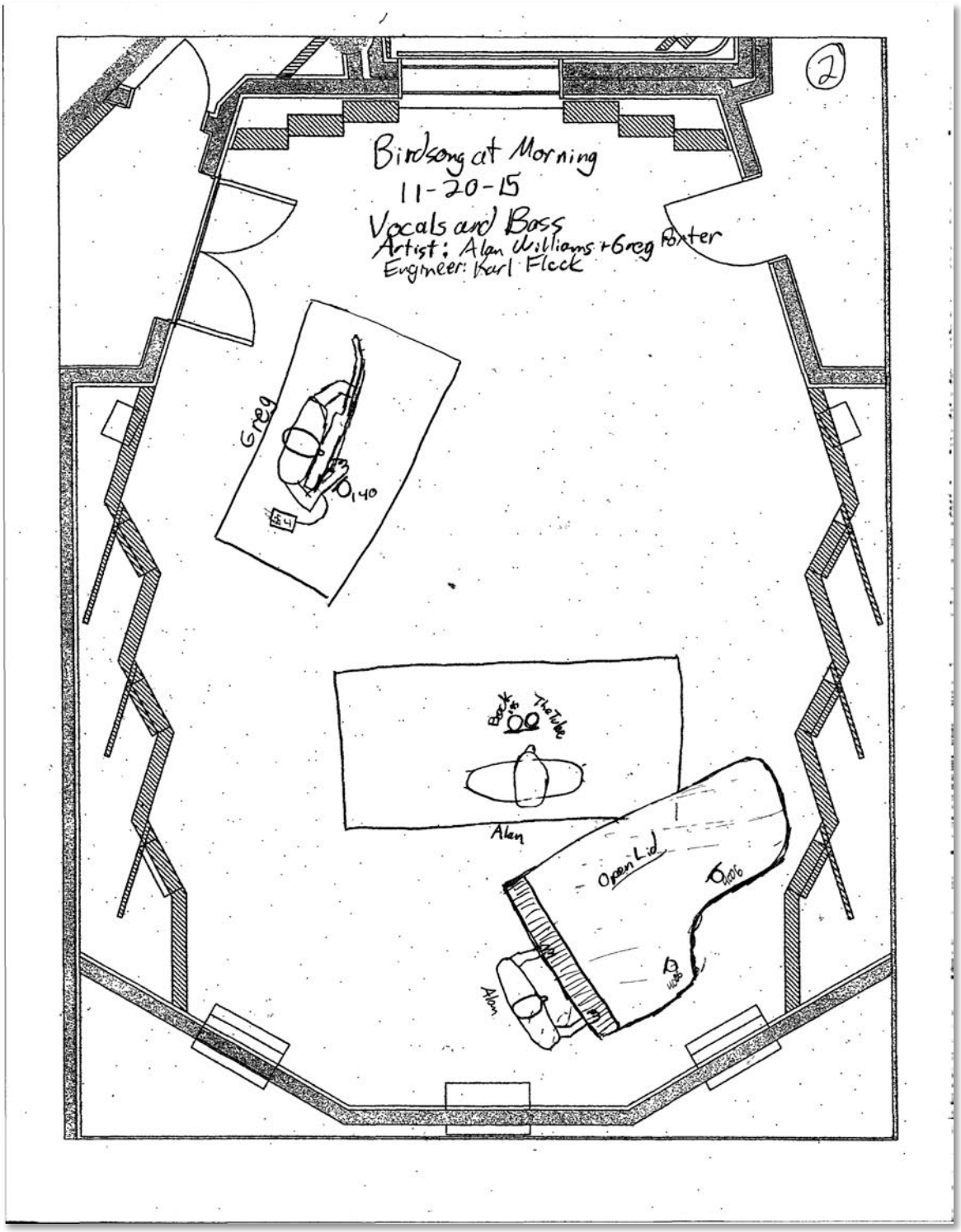

Figure 9. Session 2 Floor Plan and Mic Placement

The third session was dedicated to recording the strings. Alan wrote a string arrangement and hired his favorite string quartet for the recording. Alan has recorded strings many times for his previous albums and has created a method to simulate a full orchestra sound while only recording a quartet. Firstly, his method includes using three stereo pairs of microphones and also uses spot mic's on each instrument. The three stereo pairs are: XY, ORTF, and a Decca Tree. The spot mic's used were Neumann 140's on Violin 1 and 2, and AKG "The Tube" mic's on the Cello and Viola.

The second aspect to Alan's string recording method is a layering multiple takes. The main layer consists of three comped and edited takes of the entire quartet, using all stereo pairs and spot mic's. On top of that, there is a take of just the first Violin part and another take of just the second Violin part. We recorded these two takes using only the stereo pairs. This combination and layering of tracks simulates the sound of a full orchestra. See **Figures 10 and 11** for Session 3's input list and illustrations of the mic placement and floor plan.

3

Microphone Input/Assignment Sheet

Sound Recording Technology
University of Massachusetts Lowell
35 WILDER STREET SUITE 3
LOWELL, MASS 01854
(978) 934-3850

Date: 11-21-15 Page: 1 of 1
Studio/Machine: 114, 213
Course/Lab: Masters Rec. Project
Eng/Assist: Karl Fleck, Ethan King
Artist: Birdsong at Morning

Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note	Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note
1	Hardy 1	1	Violin 1 Spot	140		25					
2	Hardy 2	2	Violin 2 Spot	140		26					
3	Hardy 3	3	Cello Spot	The Tube		27					
4	Hardy 4	4	Viola Spot	The Tube		28					
5	API 5	5	Room 1	Schops CMC	X Y	29					
6	API 6	6	Room 1	Schops CMC		30					
7	API 7	7	Room 2	Schops CMC	ORTF	31					
8	API 8	8	Room 2	Schops CMC		32					
9	Millenia 1	9	Room 3	DPA 4006	Decca	33					
10	Millenia 2	10	Room 3	DPA 4006	Tree	34					
11	Millenia 3	11	Room 3	DPA 4006	"	35					
12						36					
13						37					
14						38					
15						39					
16						40					
17						41					
18						42					
19						43					
20						44					
21						45					
22						46					
23						47					
24						48					

Notes: _____

Crosses: Main Room Ch _____ Hallway Ch _____
Iso Booth Ch _____ Amp Closet Ch _____

Headphone Mix Notes: _____
1 - _____ 3 - _____ ST - _____
2 - _____ 4 - _____

Figure 10. Session 3 Input List

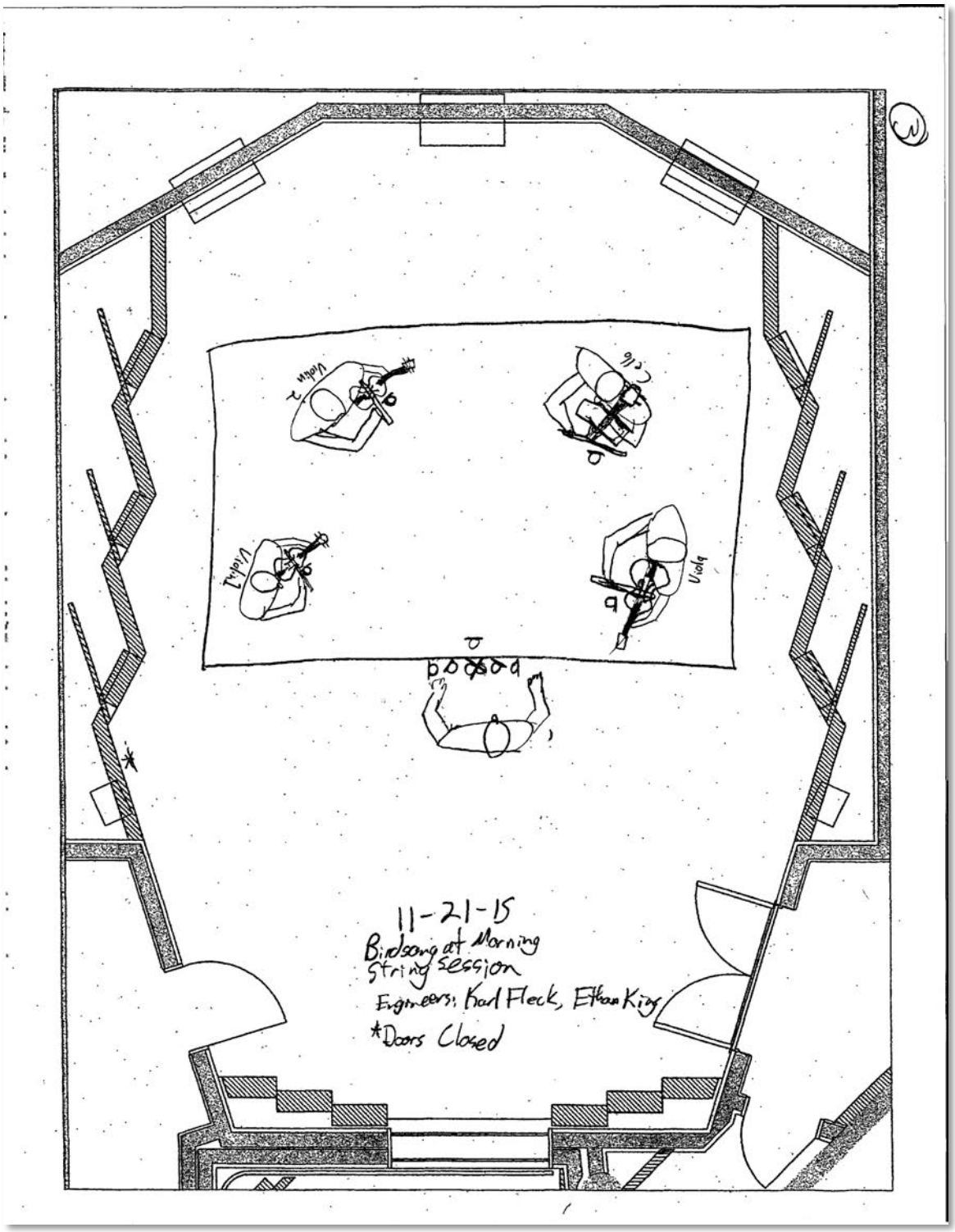

Figure 11. Session 3 Floor Plan and Mic Placement

The fourth and final session was for recording the electric guitar. Band member, Darleen Wilson played a Fender Strat style guitar with single-coil pickups. She played through a modern solid-state Fender Princeton amplifier. We recorded the guitar using a Radial J-48 DI (direct input); a Sure SM-57; and a Sure KSM-44. We placed both microphones about four inches from the speaker and the KSM-44 was placed on the center of the speaker cone while the SM-57 was placed about an inch off center.

There was an issue of noise coming through the amp and several steps were taken to reduce that problem. First, we lifted the ground on the DI and then we had Darleen physically move into the hallway to give the amp space. In the end, the solution we found to be most helpful was when Darleen turned the face of the guitar away from the amp. For one reason or another, there was feedback between the guitar and the amp. This could be because of the single-coil pickups, which are notoriously noisy. See **Figures 12 and 13** for Session 4's input list and illustrations of the mic placement and floor plan.

(4)

Microphone Input/Assignment Sheet

Sound Recording Technology
University of Massachusetts Lowell
35 WILDER STREET SUITE 3
LOWELL, MASS 01854
(978) 934-3850

Date: 12-8-15 Page: 1 of 1
Studio/Machine: 114, 213
Course/Lab: Masters Recording Project
Eng/Assist: Karl Fleck
Artist: Birdsong at Morning

Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note	Tie Line	Preamp	Channel/ Track	Instrument	Mic	Note
1	API 1	1	Elec GTR	J 48	DI	25					
2	API 2	2	Elec GTR	57		26					
3	API 3	3	Elec GTR	KSM44		27					
4	API	4	Elec GTR	DPA 4006	Room	28					
5						29					
6						30					
7						31					
8						32					
9						33					
10						34					
11						35					
12						36					
13						37					
14						38					
15						39					
16						40					
17						41					
18						42					
19						43					
20						44					
21						45					
22						46					
23						47					
24						48					

Notes: _____
Crosses: Main Room Ch _____ Hallway Ch _____
Iso Booth Ch _____ Amp Closet Ch _____

Headphone Mix Notes: _____
1 - _____ 3 - _____ ST - _____
2 - _____ 4 - _____

Figure 12. Session 4 Input List

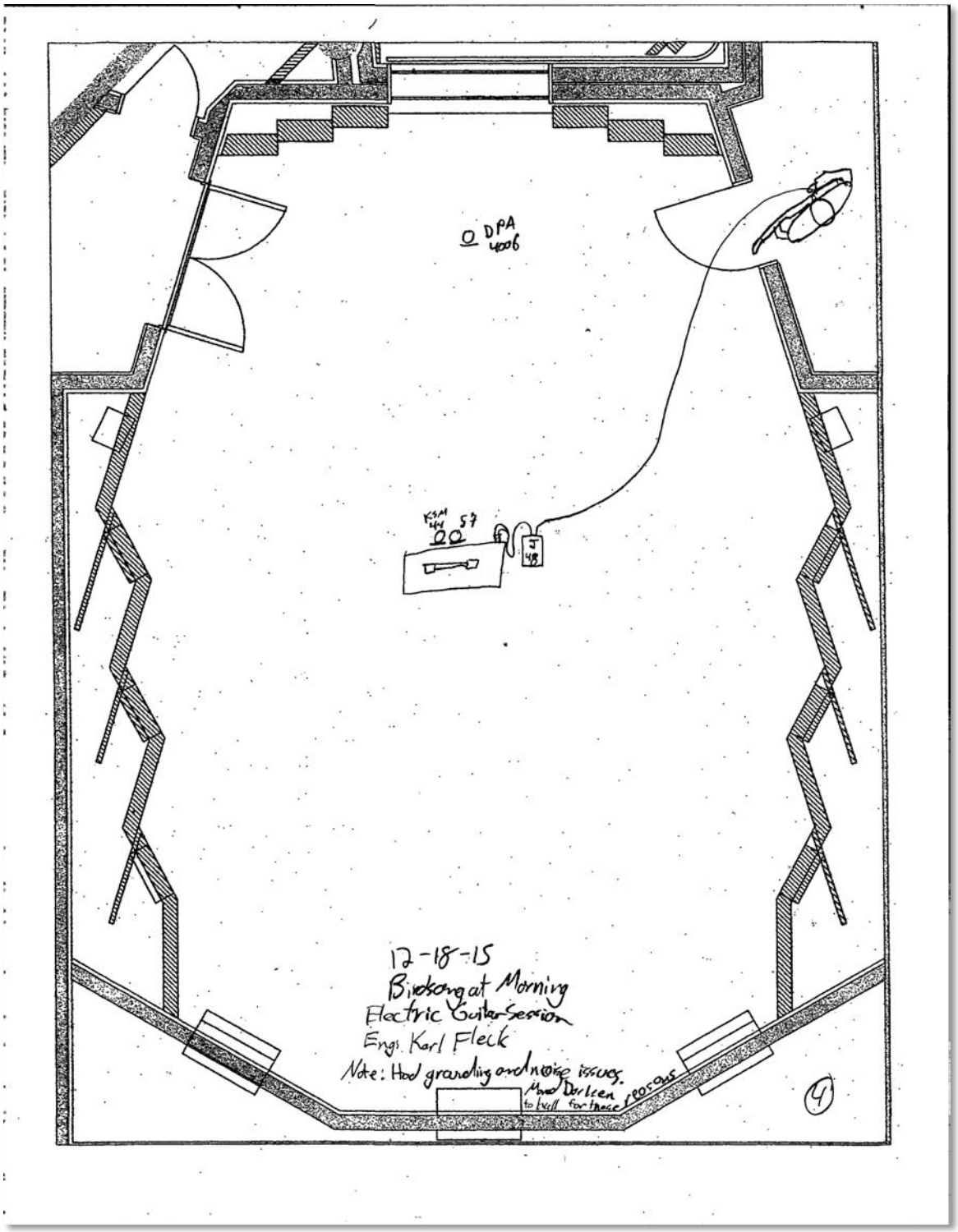

Figure 13. Session 4 Floorplan and Mic Placement

The Mixing Stage

Once the recording was completed, I edited and prepped the Protools session for the four mixers. This involved choosing the best takes of each take for each instrument and putting in all the appropriate fades on the audio clips. This process also involved using Melodyne to tune the vocal tracks.

All of the mix engineers involved in this project used Protools for all of their processing, except for Alan Williams. Because of this, a separate bounce was made of each track starting all from the same point. I put these files into a folder from which Alan could simply import all the files and they would all be lined up and start from the same point. An issue arose with the rendering of one of the acoustic guitar tracks; it appeared to have been rendered with a separate, low-level, guitar track. Due to the timeline of this project, Brandon Vaccaro solved the problem by re-rendering the file and double-checking its integrity.

After editing the session and finalizing all the components, I sent the following the instructional details to the mix engineers, Nick Dragoni, Bradford Swanson, Brandon Vaccaro, and Alan Williams. The following was sent to each of the mix engineers as an outline of the instructions and expectations:

Mix Engineer Instructions

The time has come to hand the session off to you trusted mix engineers. Before I get into the instructions, allow me to provide a little background into my goal for this exercise. As you know, I have been working with the band Birdsong at Morning, and over the past few weeks, we recorded one song. The focus of my project is mastering so I'm now handing the recording off to you four mixers. Once I get the mixes back, I will master them and analyze their similarities and differences. I will also discuss the various aspects of each mix with regard to the mastering process.

Here are your instructions:

1: Get a copy of the session from 213.

The session is located on the Media 0 drive is in the folder called "Karl Fleck_Thesis Project Files". Inside that folder there is another folder with the Protools session. There is also a folder of files for Alan, who can simply import them all into Nuendo. All the files start from the same point so it should be pretty straightforward.

Also, for those working from the Protools session, the session as it is now is about 40 gigs. I tried deleting the unused playlists and files but it's still a large session. If you can find a way to trim it down a bit, please feel free to do so. I didn't dig into it too much because I'm paranoid about losing something

important. All this to say, if you're taking the session with you, make sure you have a large enough hard-drive.

2: Mix away!

You are free to mix the session any way you please. If you want to mix completely in the box, feel free to do so; if you want to bounce it to tape and mix completely analog, be my guest. This mixing exercise is supposed to simulate real-world situations where the mastering engineer has little to no control over the mixing process.

As far as editing goes, all the tracks are as they ought to be. You should be able to open the session and mix. However, if there is something obviously wrong (i.e. missing fade) that I missed, please feel free to edit as much as you feel comfortable. But all the tracks present are of the correct take, etc.

3: Documentation

For the sake of the formal project, I would like full documentation of your mixes. This includes all effects and settings used. Also, please include the pan and fader information. If you're working in the box, screenshots of settings etc. will suffice. If analog, please fill out recall sheets.

4: Delivery

The timeline of the project is such that you should have your mix and documentation completed and ready for delivery by Tuesday, February 2nd. If you

have it completed sooner, that'd be great. This delivery date gives you all of winter break plus a few weeks to refine your mix. I will have a computer and a flash drive and we can meet and I will get your mix and documentation.

The session is 24-bit 96k so I would like to see your final mix as a 24-bit 96k .wav file. It can be either interleaved of multiple mono. Please refrain adding any large fade-in or outs to your mix. If you would like a fade, please let me know, and I will do it in mastering.

I would like to offer my sincere gratitude and thanks to all of you for partaking in this exercise. I can't communicate how much it means to me that you are taking time to work on a mix. I asked you all to do a mix because of your reputation as authorities in audio and I am honored to have you all involved in the project.

Receiving the Mixes

Alan Williams's mix was the first that I received. Alan worked entirely in Nuendo and did a few alterations to the arrangement of the song. I did not have a problem with this as first because, after all, it was his song and he was the producer. Second, it made sense musically.

Upon further review of the mix, I realized that there was already some limiting on the mix so I asked Alan for a revision. I asked him to reduce the level of the mix and also remove the limiter on his mix. Alan used the limiter to catch just the loudest peaks of his mix, but I asked for the revision because, ideally, the mastering engineer should be the one to use the limiter. After Alan made the revision, the mix was the ideal level and file type and was ready for mastering.

The next mix that was received was Nick Dragoni's. Nick came to UMass Lowell's room 213 and mixed on the API Vision console. His mix was immediately acceptable for mastering; the mix was at an appropriate level, with peaks landing at least -3dBFS and at the original sample rate and bit depth.

The following mix that I received was from Brandon Vaccaro. His mix was the most complicated and detailed of all the mixes. There are multiple changes in the entire mix throughout the song. He even took the liberty to add some parts to the song. Vaccaro re-amped the electric guitar parts and also added several percussion tracks. This addition of additional instruments was not part of the original song, but with mastering, you never know what the nature of the mixes you receive will be. Therefore, I decided to keep the mix the way Vaccaro submitted it.

SRT alumnus, Bradford Swanson, was the final mix engineer. With the time restraint and the commitment needed, it was a bit difficult trying to nail down a fourth mix engineer. First, I asked Alex Case, but quickly realized that he had many other obligations and responsibilities. Second, I asked Bill Carman, the SRT studio manager. But like Case, he was unsure of being able to complete the mix on time. Bill advised that a search for another mix engineer should ensue. This is when I decided to contact Bradford Swanson. My fellow SRT graduate students highly recommended Swanson as a mixer. Fortunately, he was able to complete a mix within the timeframe and sent in his mix the night before the scheduled mastering session.

Finding a Mastering Room

One of the practical aspects of mastering is minimizing undesirable processing, weeding out the "weakest link" in the signal path. This could include elements such as cables, D/A-A/D conversion, effects processing, the choice of DAW, and final sample-rate conversion and dithering. Another goal in mastering is to have an acoustically neutral space with speakers that can accurately represent the program material across the entire frequency spectrum. This is very difficult to come by, especially if one does not have a dedicated room of one's own. The venue for the mastering aspect of this project has changed and evolved as the mastering session date approached. The following section describes my search for a mastering venue and the compromises I made with the intent to minimize any undesirable elements in the signal path.

The first choice for the mastering venue was M-Works Mastering in Cambridge, MA because of its state-of-the art mastering facility. The mastering room featured EgglestonWorks Andra speakers, a slew of top-quality analog processing, and utilized Sequoia as its primary DAW. This was truly a professional-grade, no-compromise mastering facility. Unfortunately, the timing was not in my favor as M-Works relocated midway through my project. (M-Works now shares a mastering room with iZotope Inc.) Furthermore, the move introduced a new room and PMC speakers, which sound great, but are less familiar to me. **Figure 14** is a photograph of M-Works' mastering room pre-

iZotope.

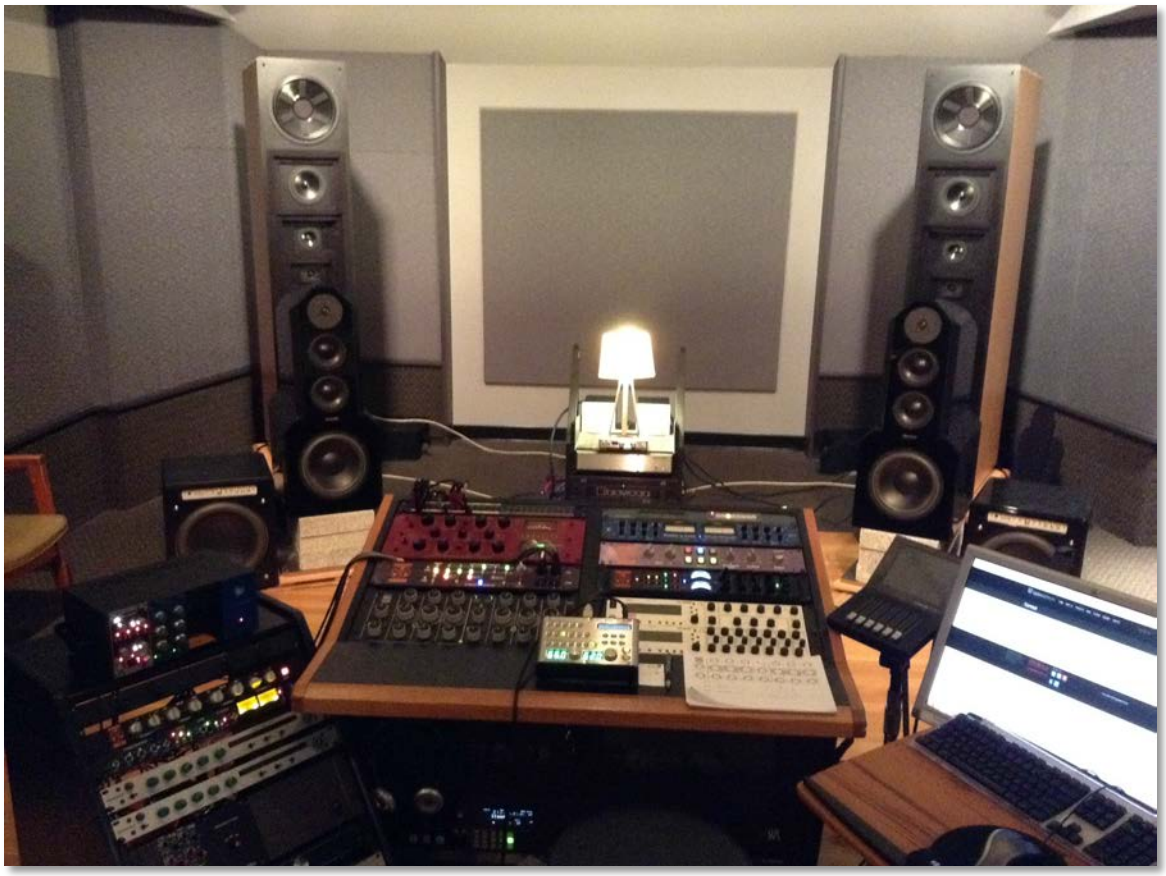

Figure 14. M-Works Mastering Main Room (December 2015)

My solution was to conduct the mastering session at UMass Lowell's main critical listening space, room 114. This space featured self-powered (active) SLS S1266 monitors, the same speakers used by Jeff Lipton at Peerless Mastering in Boston, MA. Room 114 was a mastering-grade listening room because of its acoustic design and speakers. This room also was a very familiar space because I have spent many hours critically listening to music and mastering there for over a year and a half.

With this of venue, it was realized that there are a few analog devices that I could borrow from UML's API mixing room 213 and used in a makeshift mastering rig. The plan was to utilize UML's mobile recording rig that included a Metric Halo ULN-8 interface and also use the API 2500 compressor and Manley Massive Passive (Mixing Version). This way, I would be able to use a hybrid signal chain for the mastering. This system would offer a familiar room and the use of analog processing, both key facets to quality mastering. See **Figure 15**.

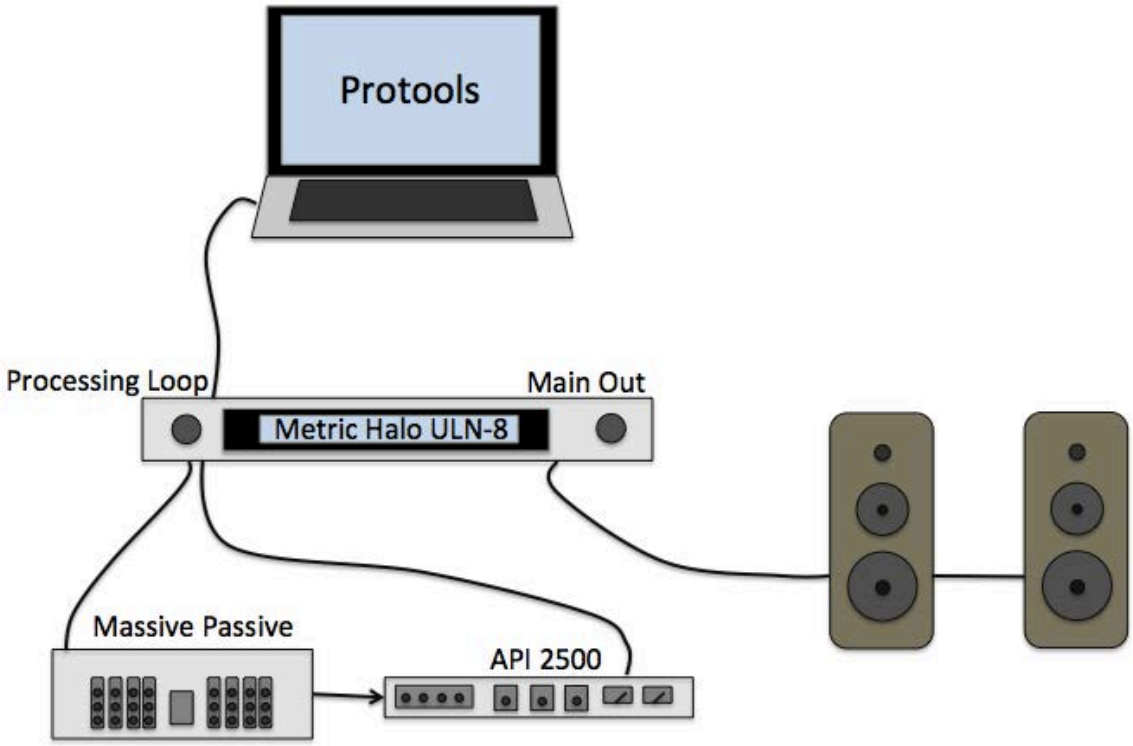

Figure 15. 114 Proposed Hybrid Signal Path

Unfortunately, after I tested the Metric Halo system I decided to abandon the plan. There were a few reasons for this decision. The first was my unfamiliarity with the Metric Halo converter. It did not help that one of the

channels of D/A was not working properly and provided an unusable amount distortion to the signal. This is not a huge issue because another pair of channels of I/O could be used, but upon further listening I decided that the signal chain was less than desirable. The second reason for altering the current plan was because in order to achieve a hybrid signal chain, I would have to use non-mastering grade cables. The combination of questionable conversion quality and dubious cabling led to the decision to do the mastering entirely in the digital domain.

The third and final stage of evolution in this mastering process abandoned the analog processing and the Metric Halo converter entirely. Instead, I decided to process the audio with iZotope's Ozone 7 mastering software. Ozone 7 features a multitude of plug-in modules designed for mastering. One can do the creative processing of a master entirely with its modules. Ozone, however, does not offer complex fade editing, track sequencing, metadata, PQ entry, or the ability to create a final DDP file. It is a great tool for EQ, dynamics, and imaging processing, but is incapable of all of the tasks required by a professional mastering engineer to create a true final master. Alas, this system utilizes the best signal-processing and mastering environment that UML had to offer at this time. Please refer to **Figure 16** for a visual of the final signal path.

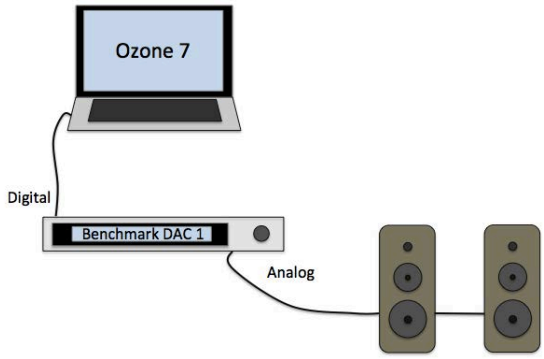

Figure 16. Final Signal Path

A Note on DAW's: Ozone 7 and Pyramix 7

After much careful thought and deliberation, I decided to master entirely within the stand-alone application of iZotope's Ozone 7. This offered a complete array of plug-ins and modules designed specifically for mastering. The modules themselves sounded great and were very powerful. Although iZotope produces professional-grade audio technology, their products are easily approachable for the "pro-sumer" market. This market includes people who love music, recording, mixing, and mastering, but do not necessarily have much technical or professional experience.

While working with Ozone 7, I found that the modules were not very specific in how they functioned. For example, their limiter module, the maximizer, had four different limiter options. They were different algorithms for their IRC (Intelligent Release Control) limiters. The question then arose, which one does what? Is there one IRC that sounds better for a certain style of music? To further complicate the matter, some of the IRC's had different selections within the individual algorithm like, "smooth" or "crisp". To make an automotive analogy, Ozone 7 can be thought of as a high-end sports car with an automatic transmission. When I used it for this project, it was flexible, agile, and extremely powerful but it was never very clear how the results occurred. In the end, as always with mastering, it was best to trust my ears. I found it helpful to listen to the different settings and see which ones sounded best for each particular application.

Furthermore, Ozone 7 featured a convenient method for A/B monitoring, which was one of the reasons why the stand-alone application was so useful. There was a button (**Figure 17** the blue ear) that I could click that level-matched the mix to the master. In other words, with this button engaged, I was able to quickly bypass all of the mastering processing and listen to the original mix and then also the mix with the processing at the same loudness level. That way, I was able to evaluate the effectiveness of the processing and make sure that I was doing an appropriate amount of processing.

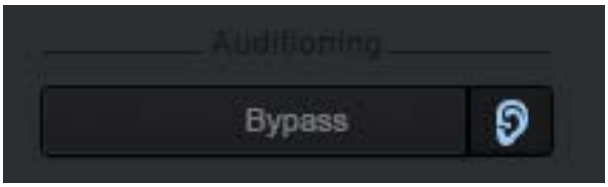

Figure 2. Ozone Level Matching Option

Another key feature of Ozone 7 was that it offered its new analog-inspired modules: tape saturation, an EQ, and a compressor. These analog modules seemed to be an obvious answer for attaining a bit of 'color' in the form of harmonic distortion. The first thing that came to mind with these options was to use the compressor in a parallel circuit because that would be a great place to add 'color' if need be.

Once I completed the effects processing in Ozone, I imported the files into Pyramix 7. I used this program to generate the sequencing, PQ, metadata, and disc

image. This was the last step in creating the final master and the only creative aspect to this process was when I applied fades and spaced the songs.

Preparing the Mixes

All of the mixes that were received were at 96kHz 24-bit except for one, Brandon Vaccaro's, which was changed from the original 24-bit to 32-bit somewhere within the mixing stage. I converted the mix back to 24-bit using iZotope's RX software. This is not generally desirable because any conversion or processing provides an opportunity for degradation of the original file. However, with proper dithering, a 32-bit file can be truncated to a 24-bit file without much of a problem. Furthermore, iZotope RX is well known for being one of the best programs for sample rate and bit depth conversion.

The Mastering Session

I conducted the mastering for this recording project at UMass Lowell's Sound Recording Technology Room 114 on Friday, March 18th. The physical signal path of this set-up was relatively minimal. The mixes were processed

entirely 'in-the-box' with Ozone 7. The signal was then sent digitally from UML's MacBook Pro to a Benchmark DAC1 D/A converter. The analog signal was then routed to the amps and speakers.

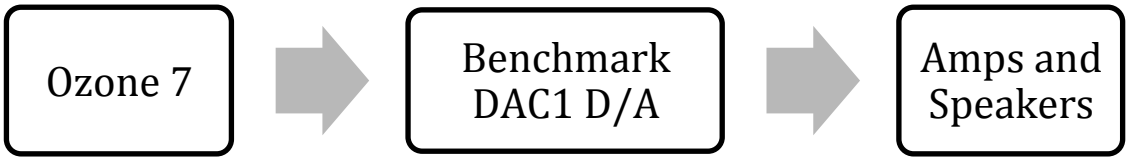

Figure 18. Final Signal Path

While setting up the session, I used Birdsong at Morning's most recent release, *A Slight Departure*, to get acquainted to the room and speakers. I also used the album as a reference recording while mastering throughout the day. I chose this recording as a reference because it was a great example of a high quality production. In addition, this exhibited a previous of work by Alan; it showed the artist's trends and production style. With this in mind, I assumed that the Alan generally wanted accurate sounding elements in the mix. For example, I noticed the drums sounded natural, without much processing in the mix. Another noticeable characteristic of the recording is the bright and crisp (high-frequency emphasis) sound of the acoustic guitars.

III. Results

First Impressions

The mastering process should be a fluid interaction between one's first impressions and the implementation of any processing effect. Within the first few listens of the mix, the mastering engineer will have an idea of the intent of the artist and what processing needs to happen in order to best highlight it. The mastering engineer will also listen for any problematic aspects of the mix. Generally the most obvious issues in a mix are in the frequency domain.

The song, "All the Sadness," was pensive and moody, perfect for a more 'vibey' and 'warm' character in the mix. I found that this character was common throughout all the mixes. The mix engineers successfully worked not for a "perfect" sounding mix, but rather incorporated some "vibe" into the mix. During the mastering, I made all the decisions around this "vibey" atmosphere. The following is a description of the first impressions of each mix.

Mix 1: Alan Williams

Alan William's mix had a peak value of 0.0dBFS so I lowered the gain at the input stage of Ozone by 3dB. Once that simple step was completed, I still noticed too many low and low-mid frequencies in the vocal and acoustic guitars.

This was caused by proximity-effect in the recording stage. The acoustic guitars also seemed to lack some high-frequency "sparkle." Lastly, the bass guitar was getting a little masked and cluttered by the obtrusive amount of low-midrange. Overall, the mix was very nice and these first impressions were simply problems found in mastering. In other words, this mix was ready for mastering since there were no large issues that needed further revision in the mix.

Mix 2: Nick Dragoni

The next mix, by Nick Dragoni, had three issues that I observed within the first few listens. Like Alan's mix, there was still some low-mid proximity effect left in the mix. Secondly, the string section seemed to have an abundance of mid-range frequencies around 550Hz. Finally, the overall sound stage was relatively narrow and was especially apparent in the strings.

Mix 3: Brandon Vaccaro

Brandon was very thorough in his mixing and freely took many artistic liberties. Brandon's mix stood out because the other mixes remained relatively neutral, presenting the music as clearly as possible, without any distractions. Brandon especially paid attention to the lyrics and interpreted the song through his mixing decisions. One example of this was at the end of the song, the final line was, "I just can't say I'm sorry anymore". Brandon then used a delay to ironically

repeat the phrase, “I’m sorry.” This is just one example of his creative mixing decisions.

Brandon also sent two versions of his mix. One was bounced ‘in-the-box’ and the other was bounced to tape and then back into the DAW. Upon review of the two mixes, I decided that the tape version had a slight but pleasant amount of tape-based artifacts. This added nicely to the overall gritty character of the song.

Aside from the interesting, the other thing I noticed was that the peak value of the mix was -7dBFS so 4dB of gain was added at the input of Ozone. While assessing the timbre of the vocal, I noticed there was little low-mid and mid frequencies and also an emphasis on the high frequencies. Another issue was that the vocal seemed a little low in level relative to the rest of the mix. Lastly, I thought the stereo width could be increased to better use the extremes of the sound stage to its fullest potential.

Mix 4: Bradford Swanson

Bradford Swason’s mix was possibly the most ‘vibey’ of the bunch. There was a general emphasis on the low and low-mid frequencies. I did not think of this as a problem, but rather an artistic decision, which worked very well for this particular song. Another compelling aspect to Brad’s mix was his use of space and reverb on the strings. He was able to achieve a uniquely pleasurable sense of distance and large-room decay on the strings. Lastly, Brad also sent a few

versions of his mix. He sent a mix with the vocal raised, one with the vocal lowered, and another with the vocal at the level that he liked best.

Like some of the other mixes, this mix was a bit low in level so I added another 3dB of gain was added at the input of Ozone. Furthermore, there was a relatively sizeable amount of proximity effect on the vocals, especially around 209Hz. After listening to the different versions it was apparent that sometimes in the chorus of the song, the vocal seemed to get lost, masked by the surrounding sounds. Chose the "vocal up" version of Brad's mix for the final mastering.

The Mastering Process: Common Steps Taken

After a few listens through the original mix and culminating first impressions, there were many ways to begin mastering. For this project, I employed a specific method in implementing the desired effects. The following is a step-by-step explanation of my preferred approach to mastering the four mixes, as seen in **Figure 19**.

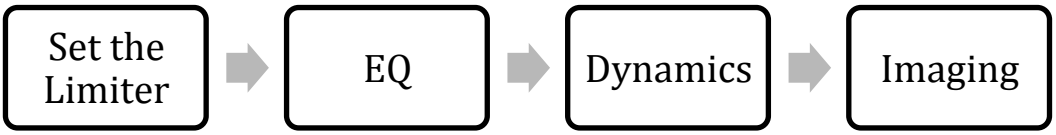

Figure 19. Common Processing Chain

1: Choose and Set the Limiter: I began the session with the monitors set at a constant level. Again, I found it best to listen to the music at around 85dB SPL.

With that in mind, the mastering began with an assessment of the various brick-wall limiters within Ozone 7. The intent with choosing and setting the limiter first was that the effects processing could be accomplished while listening to the effects of the limiter. For example, how did the limiter sound and how did it react to any dynamics processing? How did it react to any EQ adjustments? Those were important questions to ask because the processing will be done in conjunction with the sound of the final limiter.

2. Apply Appropriate EQ: As mentioned previously, problematic frequencies were possibly the easiest issues to spot. In the case of this song, there may be too much low-mid in the vocal because of proximity effect during the recording process. The best time to tame any problems is during mastering with the addition of EQ. Generally a mastering engineer will do reductive EQ more frequently than additive EQ. The idea behind that was to reduce the abundant frequency and get it to match evenly with the rest of the frequencies. However, sometimes there could be a moment when additive EQ is necessary for example, if an acoustic guitar could use more “brightness” in the high frequencies.

3. Dynamics: The next step in the process was to add any necessary compression. For this project, the compression ratios were low and the thresholds were high. The aim was to delicately compress any loud sections so that they could be brought up in level on a global scale at the final limiter stage. The compression

was light and the effect was very subtle, which was the intent with the compression.

In general, this particular song did not require much compression because the level remained relatively consistent throughout its duration. It also did not need much compression because the final master did not need to be overly loud. Again, sometimes the idea behind compression in mastering is to reduce the extreme peak levels so that the overall level of the song can be raised. This did not need to happen with this song. The aim for this project was to preserve any natural transients and overall musical ebb and flow.

4. Imaging: Once the frequency and dynamics processing occurred, there may be an issue with the sound stage, in particular, the stereo width. Ozone features many ways to manipulate the stereo width with their "Imager" module. This particular module can increase or decrease the width of a selected frequency range. I found this tool especially helpful for this project because some of the mixes did not utilize the stereo width to its fullest potential.

The Imager module in Ozone is very powerful and it was important to not get carried away with the settings because it could potentially cause more harm than good. For example, spreading out the stereo width could leave holes in the perceived imaging. The center would remain but the left and right could be pushed so far than there is a gap in the imaging between the center and the extreme left and right.

The Mastering Process for Each Mix

Master 1: Alan Williams

The mastering of Alan's mix began with an assessment of the various limiter settings within Ozone. My past experience with Ozone showed that IRC III was generally the preferred algorithm. However, after some listening the IRC IV setting seemed to be more appropriate to this mix. In fact, IRC IV was used for the entirety of the mastering on this project.

Once I chose the limiter setting, I set the threshold to -6.5 and the final output ceiling was set to -0.3. The threshold was low enough to catch just a few peaks and did very little overall limiting. The ceiling was set to -0.3 to avoid any inter-sample peaks.

Figure 20. Alan Williams Limiter Settings

The first issue that I noticed in the frequency domain was the abundance of low-mid frequencies due to proximity effect. After searching for the problematic frequencies, I made a reduction at 183Hz. That was the main frequency-based issue caused by proximity effect.

Upon further listening, I discovered several more problematic frequencies in the lows (89Hz), mids (613Hz and 1.7kHz), and highs (4.5kHz). All of these frequencies required a reduction around 1.5dB except for the high frequency, which only needed a reduction of 0.5dB. Furthermore, I decided that the acoustic guitars could benefit from some additional high frequencies so I employed a Baxandall shelf to boost 8.3kHz and above by a small but noticeable, 0.3dB. I added some additional EQ was added to the mastering chain so that the problematic frequencies would be tamed even further. This also included the use of the Dynamic EQ, which was essentially a multiband compressor with the character of an EQ. This means that the compression can be added to a specific frequency and that the character of the threshold has a 'Q' control. **Figure 21** shows a display of the EQ used where the Y-axis shows the amount of addition or

reduction in dB and the X-axis shows the audible frequency range in Hz.

Figure 21. Alan Williams EQ Settings

The dynamics module was next in the chain, which I used as a compressor in M/S. I found the dynamics module to be very powerful and diverse in scope. For this mix, it was used to lightly compressed the mid channel while not compressing the sides at all. Additionally, there is a multiband feature to the compression so I used two bands of compression on the mid channel. I used a low-band that compressed the low frequencies more dramatically than the highs. This was an effort to control the dynamics of the vocal and bass, which were in the center.

Even though the Imager module was not used on this mix, there was some sound stage manipulation. The mix already had an appropriate amount of width but it seemed like the extreme left and right could be a bit louder. This was resolved through the use of the Dynamics module in M/S. The mid channel was

lowered in overall gain by 0.4dB and the sides were raised by 0.8dB. This especially gave a boost to the strings and acoustic guitars.

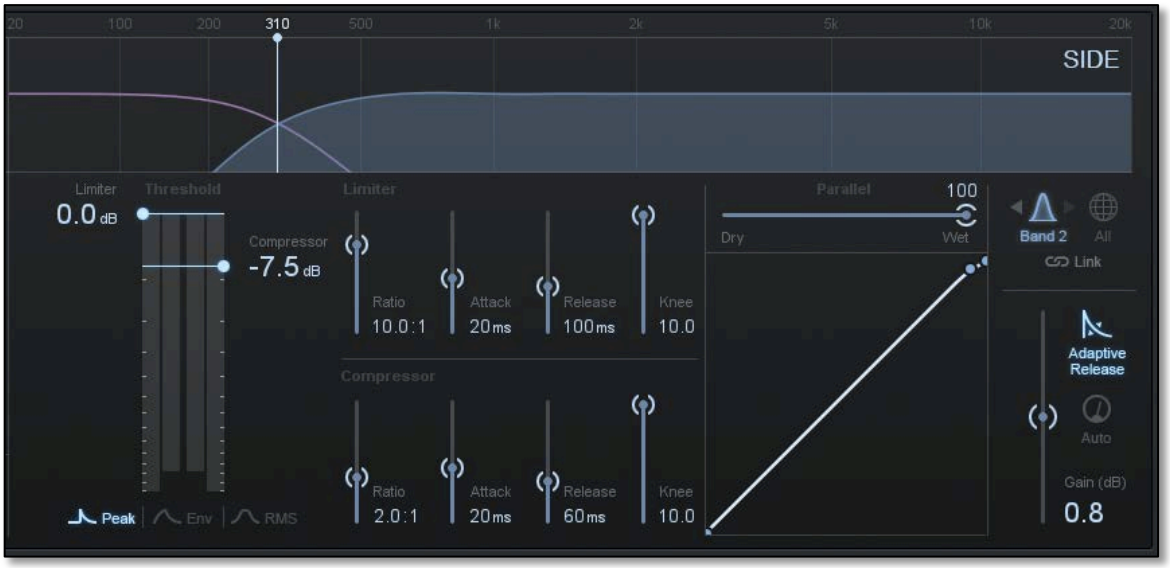

Figure 3. Alan Williams Dynamics Settings

Master 2: Nick Dragoni

I worked on Alan's mix for about an hour and then I decided to move on to the next one. Nick's mix was a bit different than Alan's, but they did share some commonalities, especially in the frequency domain. There were frequency reductions in the low-mids (180Hz), mids (555Hz and 1.3kHz), and high-mids (2.65kHz). Furthermore, there this mix also required a slight boost of 0.4dB in the

high frequencies with a Baxandall shelf at 6.4kHz.

Figure 23. Nick Dragoni Stereo EQ Settings

After this global EQ adjustment, there still seemed to be an abundance of low-mids in the side channels, specifically in the acoustic guitars, electric guitar, and strings. I used another EQ module in M/S to reduce this low-mid build-up in the sides. This reduction was centered at 199Hz and was reduced by 2.5dB.

Figure 24. Nick Dragoni M/S EQ Settings

The dynamics processing was quite minimal for this mix. I set the threshold on the compressor so that it reduced only the few loudest peaks. Additionally, this Dynamics module was also used in M/S, serving a similar purpose as Alan's mix. The sides seemed as if they could use a slight boost in gain. Therefore, I raised the gain in the side channel by 0.8dB relative to the mid channel. Again, this is a slight boost, but it raised the apparent loudness of the sides.

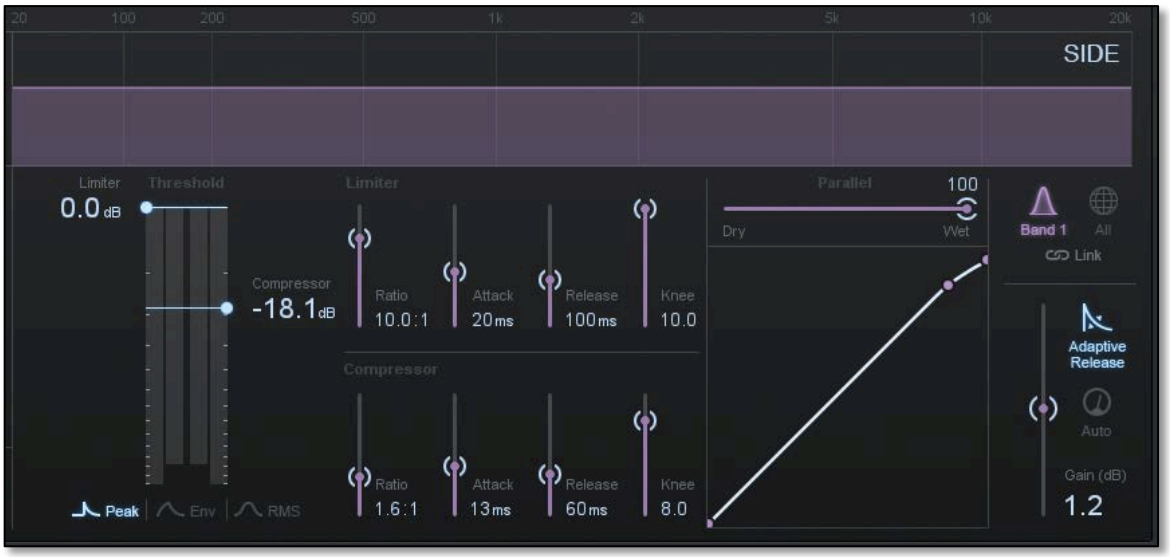

Figure 25. Nick Dragoni Dynamics Settings

One of the first impressions for this mix was that the stereo width was a bit too narrow. I used the Imager module to alleviate this problem. This module was split into two bands, one for the lows and the other for the rest of the higher frequencies. I increased the high-frequency band by "22.8" which spread the image out just enough. Meanwhile, I narrowed the low frequency band to help

with the perceived “muddiness” of the guitars in the sides.

Figure 26. NIck Dragoni Imager Settings

Master 3: Brandon Vaccaro

The next mix I mastered was Brandon Vaccaro’s. This diverse mix changed throughout the entirety of the song. For example, the voice in the first two verses features a radio-like EQ effect. This effect was removed at the first chorus, only to return at the very end of the song. It was decided to make the EQ decisions based on the ‘natural’ sounding vocal sections.

This mix also had some of the familiar frequency-based issues that were common to the rest of the mixes. The largest reduction in EQ was centered at 180Hz. This was the only major timbral alteration. There were, however, two smaller reductions at 2.1kHz and 4.4kHz. Those reductions were a reaction to the

abundant high frequency content of the vocal.

Figure 27. Brandon Vaccaro EQ Settings

There was one specific instance where I used the Dynamic EQ module. In the first verse, there are the words, “I lost the key to your door”. The word “key” was sibilant, which caused an intense concentration of high-mid frequencies. The Dynamic EQ was used to reduce those frequencies at that moment, which were most prevalent at 2.48kHz.

Figure 28. Brandon Vaccaro Dynamic EQ Settings

Another issue was the prominence of the bass guitar in the mix. I resolved this problem by using the Dynamics module utilizing the multiband function. I assigned the bass frequencies to their own band, which crossed over at 274Hz. After that, I reduced the bass band in gain by 0.8dB and the upper frequencies were increased by 2.3dB. This essentially lowered the bass frequencies by 3.1dB relatively to the rest of the frequencies.

Figure 29. Brandon Vaccaro Dynamics Settings

The last module I used for this master was the Imager. Again, it seemed as if the stereo width of the soundstage could benefit from some mild spreading. I applied the same method that was used with Nick Dragoni's mix to this mix. I

spread the frequencies above 140Hz by “17.7” which helped increase the width.

Figure 30. Brandon Vaccaro Imager Settings

Master 4: Bradford Swanson

The final mix that I mastered was Bradford Swanson’s. This mix was probably the ‘warmest’ sounding of the bunch and, again, had an abundance of low-mid frequencies. It also had relatively less high frequency content. Even though this mix had so much low-mid information, it was not necessarily a huge issue for me. I interpreted it as an artistic decision to make the sound feel ‘warm’. Even though I understood this intent, several reductions had to occur. The most prominent reduction I made was at 205Hz and several smaller reductions were needed in that area also. These other frequencies were 140Hz, 332Hz, 409Hz, and a high-mid reduction at 2.2kHz. Additionally, I used the Dynamic EQ module to help tame the low-mid frequencies at 209Hz.

With these reductions in place, there was still a lack of high-frequency content so I used a Baxandall filter to boost the high frequencies by 0.9dB starting at 4.5kHz. This added some definition to the acoustic guitars, strings, and vocals in particular.

Figure 31. Bradford Swanson EQ Settings

I used compression very lightly on this mix, again, only slightly reducing the loudest sections. I also used the Dynamics module in M/S to raise the level of the sides by 0.3dB relative to the mid channel. This gave the sides the extra boost needed to make it stick out even more, allowing the strings and acoustic guitars to

be featured a bit more than they were in the original mix.

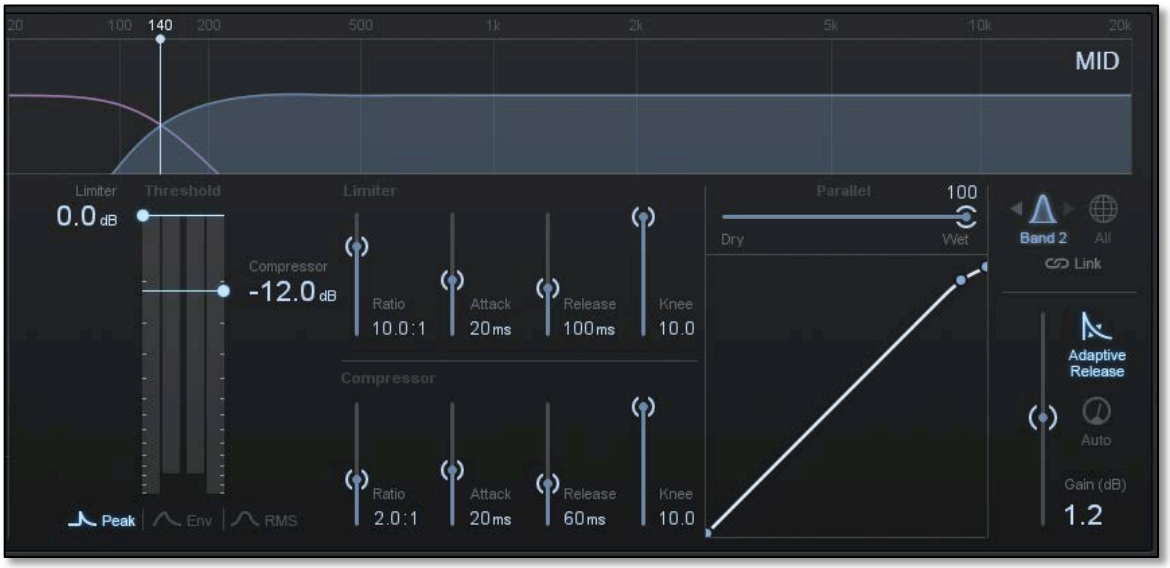

Figure 32. Bradford Swanson Dynamics Settings

Master 5: The EP Match

For this phase of the project, Alan William's mix of "All the Sadness" was incorporated into a small, three-song EP format with two of his previously released songs, "Murderous Friend" and "Down in the Hole". Again, these additional two songs were mixed by Alan Williams and mastered by Adam Ayan. These songs were chosen for their similar instrumentation and overall calm character. There was a natural progression among these three songs; "All the Sadness" made a perfect segue from "Murderous Friend" to "Down in the Hole" in terms of overall timbre and loudness.

When mastering for a single, there are no restrictions on the amount of processing used by the mastering engineer. He or she has one goal in mind: make the song sound as good as possible.

Mastering for a single gives the mastering engineer the freedom to enhance the song to its utmost potential, whereas mastering songs as a part of larger works is more restrictive. More often a song exists within a larger group of songs, either in a small group of songs called an EP (Extended Play) or a larger full-length album, called an LP (Long Play).

In this situation, it is important that the mastering engineer keeps the entire scope of the project in mind while mastering each individual song. He or she will notice common aesthetic trends throughout the work and should react to its problems within the larger context. There may be some alterations that he or she might want to do when mastering a song, but cannot because the change would make that specific song stand out, distracting the listener from the atmosphere of the record as a whole. In other words, when it comes to an EP or LP, keeping the integrity of the whole work can restrict the mastering engineer.

One of the goals for this exercise was to match "All the Sadness" with the other songs, for example, in terms of frequency content and overall character. One of the main challenging elements of this exercise was getting the vocal of "All the Sadness" to match the other songs. The processing of the mix for the EP match exercise was continued from the previous mastering of Alan's mix. With the "single" version of the master, there were a few things that needed to happen to get the song to fit with the others.

To get the vocal to match, some further EQ'ing was necessary. The timbre of the vocal of "All the Sadness" was a bit brighter than the other two songs. To get it to match with the other two songs, I made a reduction of 0.5dB with a Baxandall shelf at 5.9kHz, which dimmed the overall brightness to the desired level.

Furthermore, the "single" master of "All the Sadness" featured a touch more low and low-mid frequencies than the other two songs. Adam's masters of the two other songs were clear and focused in the lows. In order to get "All the Sadness" to match with his masters, I made a further reduction in the low-mids around 180Hz. That EQ move helped somewhat but what was most noticeable was the buildup of low-mids because of proximity effect. In order to alleviate this problem, the Dynamic EQ was used to reduce those frequencies. The use of the Dynamic EQ reduced the low-mid buildup enough to get it to match with the other two songs.

In the end, the three songs naturally worked together well as an EP. Even though the recording process was different for "All the Sadness", the same person mixed all of these songs, which was the most profound element in common and successfully matched the aesthetic of the three songs.

Sequencing in Pyramix

After the songs were matched in terms of creative aspects, I imported them into Pyramix 7 and sequenced. This involved many steps in the creation of a

final disc image. The first step I took was to put the songs in order. After some thought, the EP seemed to flow well with the order: “Murderous Friend”, “All the Sadness”, and then “Down in the Hole”. This order had a natural progression and crescendo from “Murderous Friend” to “Down in the Hole” in terms of timbral character, instrumentation, and loudness.

The timbral character in “Murderous Friend” was dark and brooding; there was not much apparent high-frequency sparkle in the vocal or guitars. “All the Sadness” expanded the high-frequency content a little bit, while “Down in the Hole” features a pleasant amount of high-frequency content.

There was also an increase of instrumentation as the EP progressed. “Murderous Friend” was relatively sparse. It included mainly vocals, acoustic guitar, strings, bass, and electric guitar. “All the Sadness” was similarly sparse but also included background vocals and piano. This was a small addition but it added to the growth of the EP. Furthermore, the musical arrangement of “All the Sadness” was a bit more complex than “Murderous Friend”, featuring a well-arranged solo string section. This lead to the next song, “Down in the Hole” which featured a much more complicated arrangement. The most prominent difference was the addition of drums.

The overall loudness of the songs also increased as the EP progressed. The RMS level of “Murderous Friend” was -14.7dB RMS, “All the Sadness” was -13.2dB RMS, and “Down in the Hole” measured -12.6dB RMS. I wanted there to be an overarching crescendo that lasts throughout the entire EP.

The next step in the sequencing of the EP was determining the appropriate spacing between the songs. This included moving the songs, in Pyramix called "clips", to the desired physical location. Next, I determined the spacing and timing of the songs, which included fine-tuning the amount of time between the clips and also the fades at the beginning and end of each clip. I determined the spacing and application of fades largely by musical intuition. I found it best to listen to the last thirty seconds of one song and then try to find a relevant pulse on which to place the downbeat of the next song. This often required several revisions to find the desired gap between the songs. **Figure 33** shows Pyramix's edit window.

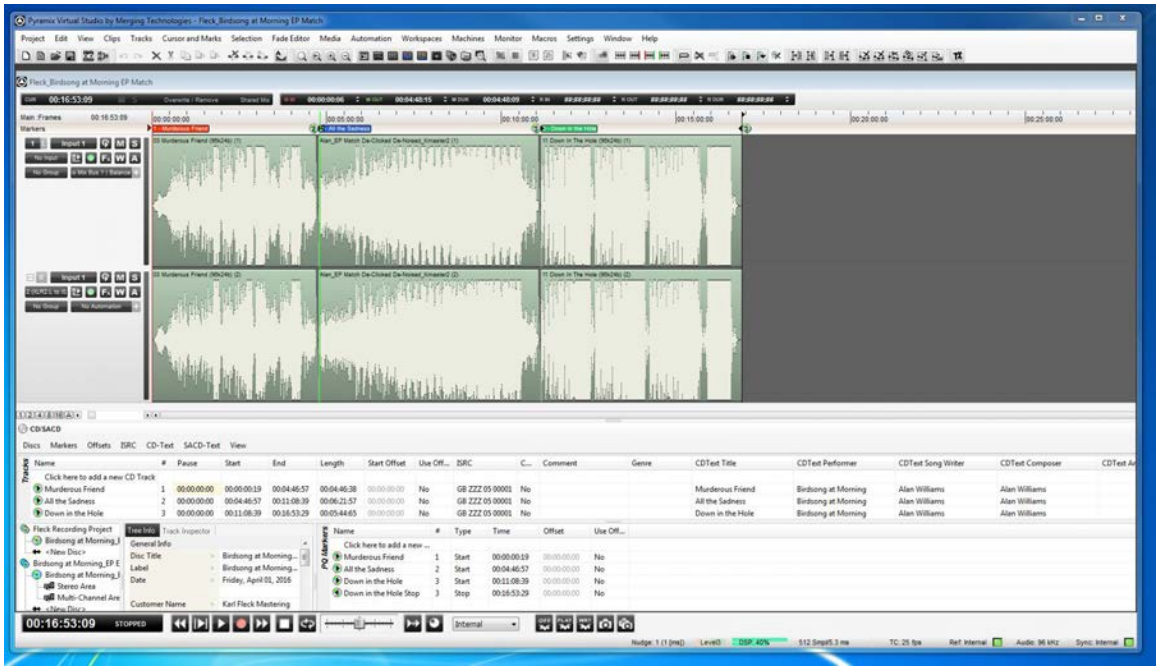

Figure 43. Pyramix Edit Window

The addition of fades also had a crucial aspect to the spacing of the EP. For this style of music, I found it best to let the song fade out as naturally as possible with the help of transparent fading. Sometimes the natural decay was a

bit too prolonged and the addition of a fade would assist the decay and allow the song to end at the desired, musically relevant time. Pyramix's fade editor allowed for the creation of specific, natural-sounding fades. **Figure 34** shows Pyramix's fade editor window.

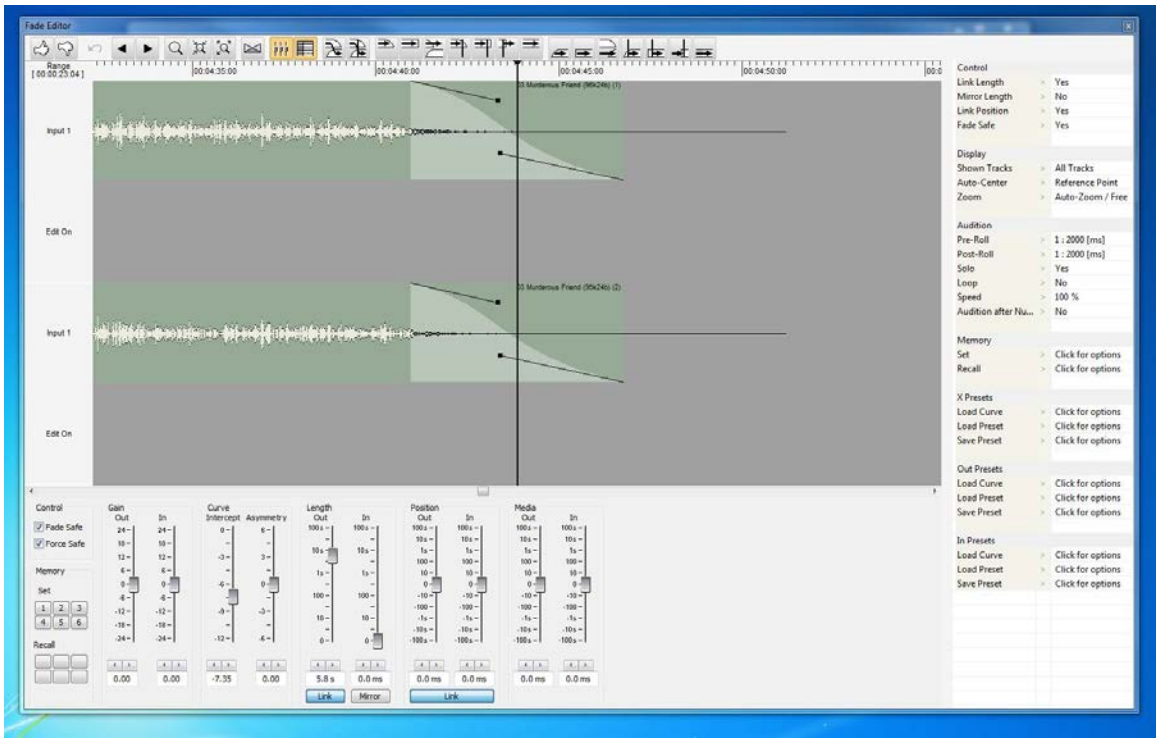

Figure 34. Pyramix Fade Editor

Once the three songs were adequately spaced and fades were added, the next step was to create the PQ points. Essentially, these points told the program where the song began and ended. These were stop and start points applied just before the song started and right after the song faded out.

Once I completed these steps, the metadata was entered. This included the text information such as the disc title, date, track title, composer, et cetera. This information is necessary for the duplication company who would use these files to create the physical CD copies.

The final step in the creation of the EP was to generate a CD image. This was a process that took all the information entered in the session and created a series of files that would be used in the creation and duplication of CD's. This was the step in the process when the full-resolution 96k 24-bit files were reduced and truncated to Redbook standard 44.1k 16-bit files. This was something that Pyramix did automatically; there was an option to generate the CD image in cooperation with Redbook standard. Once the CD image was generated, a folder was created that included all of the exported files. This was the finish line for this EP exercise. The three songs were matched in creative aspects as best as they could be, properly sequenced, and finally exported in accordance to Redbook standard. **Figure 35** shows Pyramix's CD Image Generator

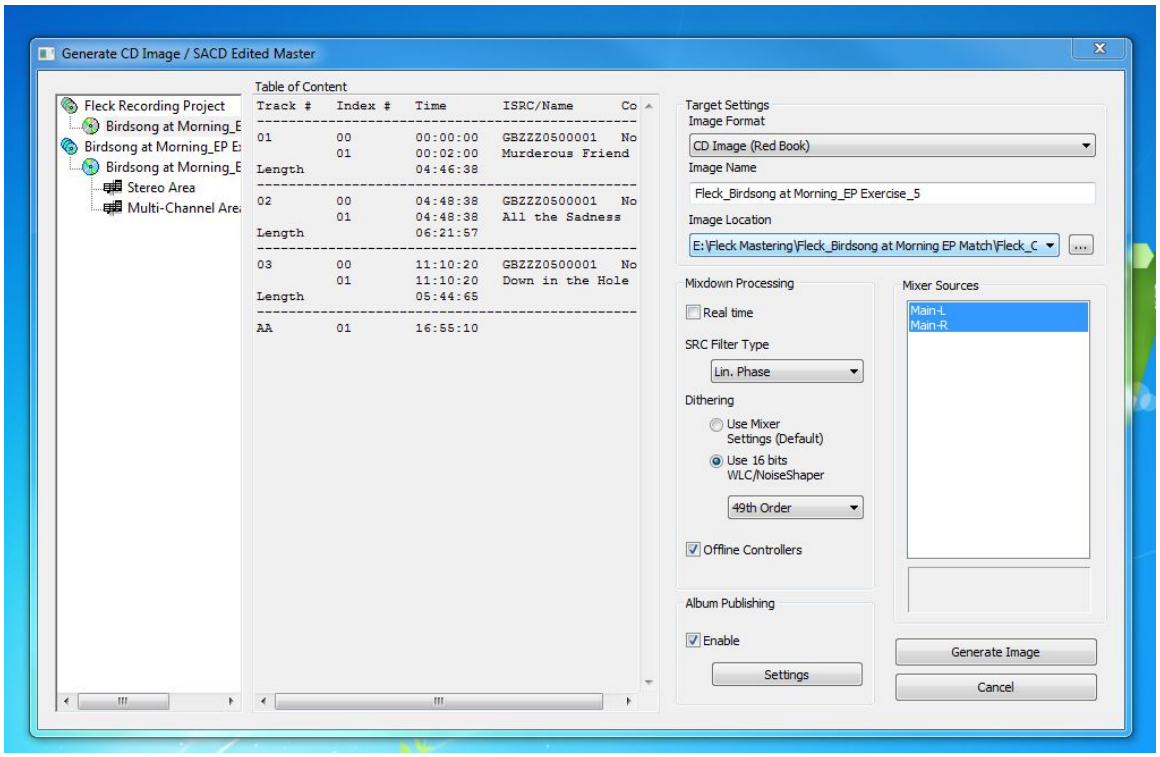

Figure 35. Pyramix CD Image Generator

IV. Discussion

The Mastering Process: Thoughts on a Technical Issue

From this experience, I learned that there were many common challenges inherent in each mix. These were either an issue innate to the recording, an issue of the mixer's monitoring situation, or a combination of the two. After these mixes were mastered in Room 114, they were additionally reviewed in UML's Rooms 205 and 223, which featured Event Audio and Genelec speakers, respectively. These rooms have always been thought to be rather "bright", emphasizing high frequencies and lacking appropriate low frequencies. With this in mind, the masters seemed a bit too bright. This brightness furthermore accentuated the amount of noise that was in the masters. This noise was probably caused by the AKG "The Tube" microphones and generated an abundance of high-frequency hiss, which extended well beyond 20kHz. While comparing the master of "All the Sadness" with the other two songs, the high frequency hiss was unacceptable needed to be removed in order to get it to match with "Murderous Friend" and "Down in the Hole".

The hiss was in all of the mixes, except for Brad's. In order to combat the hiss, the three problematic original mixes were imported into iZotope RX and de-noised. Thankfully, I was able to extract a clean noise-profile to eliminate a large amount of the hiss. With that said, this tool was used lightly, aiming not to get rid

of *all* the hiss because it was also an artistic element and adds character to the overall impact of the song.

Why then was the hiss never so apparent before? After all, each mixing engineer listened to the song many times over. Again, the hiss was always acknowledged but never considered a hindrance until it was reviewed on bright speakers. Perhaps the mix engineers also were not monitoring with bright speakers and allowed the hiss to remain.

After I completed the de-noising procedure, I then processed the mixes in Ozone with the same settings as before, just without the noise. Listening on the bright speakers was an unnerving experience because it begged the question, which speakers were most accurate? Where was home base? After much deliberation, I decided that the original mastering done in room 114 would be home base, so the original mastering effect settings were not altered to sound better on the bright speakers. In the end, this was a great example for the necessity of two things: 1. extremely accurate speakers and 2. a lot of time listening to those speakers.

After resolving this technical issue, I then resumed my attention to analyzing the slight differences between the four mixes. Under the microscope of a critical ear, the mixes were all incredibly diverse: the strings featured various reverbs, the voice had different timbres, the acoustic guitar was more or less features, etc. I observed each detail and variation and my mastering decisions reflected those differences. I should note that these decisions were made to

increments of a tenth of a decibel, demonstrating the nitty-gritty nature of mastering.

V. Conclusion

General Trends and Conclusions

On a broader scale, I found three common trends that emerged while mastering each of the mixes. These trends were the abundance of low-mid frequencies; the narrowness of the soundstage; and the correlated relationship between the mix and the master.

I attributed the low-mid build up to proximity effect stemming from the close-microphone technique used to record the vocals and acoustic guitars. While recording, there was a decision to close mic these sources, making the compromise between the mutually inclusive timbral detail and proximity effect. While mastering, I found it difficult to react to the low-mid abundance because of the question, how much was too much? It was difficult to determine the line between an artistic amount of low-mids and problematic, unintentional amount of low-mids. In the end, I left healthy amounts of low-mids in the masters because it added to the artistic timbral character of the song and gave it a "warm" feeling. I made this decision based on experience and musical sense.

The next trend that emerged while mastering the mixes was the apparent narrowness of the sound stage. Over the course of the mastering session, I increased the soundstage for two reasons. The first reason was to utilize the stereo width to its fullest potential. This widening of the image worked well for "All the

Sadness" because the arrangement was dense enough to handle the spacing. In other words, there was enough information in the side channels that holes did not appear in the soundstage between the middle and the extreme sides.

The second reason for the increase in the soundstage width was to give the musical arrangement an appropriate amount of space. This gave each instrument enough room in the soundstage to be properly highlighted without being overly masked. This was especially helpful because there were a lot of instruments that had frequencies specifically in the low-mid frequency range.

So then, why were all the mixes lacking in the width of the sound stage? This could be attributed to the mixer's monitoring situation but more likely it could simply have been overlooked. The widening of the stereo image was relatively slight and a move appropriate to mastering. Again, it was a creative step that was lead by experience and musical intuition.

The final trend that emerged was that the mix and the master were intimately related. My philosophy for mastering during this project was to enhance each individual mix in accordance with the intent of the artist and mix engineer. The philosophy was not to turn the mix into something completely different, implying that, as the mastering engineer, I should not deviate drastically from the mix. After all, the mix engineers had already approved the mixes before they were sent to mastering. If there were any technical issues with the recording, they should have been fixed in the editing or mixing stage.

VI. Recommendations

As mentioned before, audio mastering is a diverse field; there are different genres of mastering, just as there are genres of music. This project's one example of mastering "All the Sadness" is just one genre of music and requiring relatively simple mastering techniques. Some recommendations for further study are as follows:

- **Style Variations:** Different styles of music require different styles of mastering. There are some things a mastering engineer would do to a metal or EDM record that he or she would not do to a classical or jazz record. For each genre of music, there are certain guidelines for the mastering of each genre and this topic could easily be explored further.
- **M/S, side-chain, and parallel techniques:** Standard application of EQ and compression are relatively common practices for mastering engineers but the use of M/S, side-chain, and parallel techniques are areas where further exploration can occur. For example, what would happen if the typical roles are reversed and compression is used in M/S and EQ is used in parallel? Can these tools be used in a meaningful way?
- **Getting out of the box:** Because of technical restrictions, the mastering for this project exists entirely in the digital domain. This was convenient because it makes for simple recall and graphical displays of settings, etc. Furthermore, many mastering engineers prefer a completely digital signal

path. Nevertheless, the majority of mastering engineers utilize a hybrid, analog and digital signal path. For a study in contemporary mastering techniques, a hybrid signal chain should have been used because it is the most common processing path. However, there are constant improvements in digital technology both in creating new, groundbreaking technologies (ex. iZotope's IRC limiters), and in emulating classic analog equipment (ex. Universal Audio plug-ins). Perhaps purely digital processing will be the widely accepted in the future.

- **Exploring the possibilities of surround:** This study focused on two-track stereo mastering. This is by far the most common style of mastering, but there are many projects mixed in surround. A further investigation into surround sound mastering techniques would a valuable, especially for film and video game applications.

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Overall Conclusion

The mastering process is a constantly evolving system. There will always new clients with new projects, providing the mastering engineer with new challenges on a daily basis. Furthermore, mastering is a balancing act between musicianship, technical know-how, interpersonal skills, and an undying quest for perfection. The mastering engineer must be fluent in all of these skills in order to be successful in this competitive industry. My desire for this thesis is to provide insight into the world of mastering and present concrete ways of understanding and defining the mastering process through the culmination of the interviews, mastering project, and guidebook.

The Guide to Common Practices in Mastering provides the reader with a brief introduction to some of the technical aspects of mastering. This introduction offers the foundation for technical exploration in the mastering chain. It provides information about the common hybrid signal path and allows the reader to contemplate further signal-processing possibilities.

The Interview section offers some personal insights into the practicality of becoming and working as a mastering engineer. The results are wonderful assets for gaining a basic understanding of what mastering engineers do. This is important knowledge especially for a client or fellow audio engineer, but once these basic processes are understood, the mastering engineer will have his or her personal preference as to how they like to work. As the interviews prove, there truly is no right answer to any specific detail in mastering – instead the more

experience in the field, the more the mastering engineer finds the workflow that is best for them and their specific client-base.

Furthermore, from the interviews we learn how mastering consists of utilizing several general elements including DAW's, EQ's, compressors, and limiters. These are some of the tools that every mastering engineer needs in order to do their job. The difference between mastering engineers exists in which DAW, EQ, compressor, and limiter they use and the combination in which they implement each piece of gear. Furthermore, the more important difference between mastering engineers is in their personal taste, musical intuition, background, and how they use the gear they have.

From the mastering project of this study, I learned that discernment is probably one of the most valuable qualities in a mastering engineer. The mastering of "All the Sadness" is proof that the mastering engineer must be able to discern which tools he or she should use in order to best serve the provided material. In regards to this project, the master did not require any complicated signal processing like parallel or multi-band compression. Though those processes can be suitable for other styles of music, I decided they were not appropriate to the style of "All the Sadness".

In summation, experience and consistency are the mastering engineer's best friends. The mastering engineer can only make truly informed decisions after he or she knows the room, speakers, converters, cables, EQ's, compressors, limiters, and any other signal processor. This is the reason mastering engineers customize their room exactly to their preference and then change it as little as

possible. This experience along with a consistent room is a mastering engineer's "home base." Additionally, the best way the mastering engineer creates that perfect place is by listening consistently to reference recordings and mastering in the room on a daily basis.

Because the mastering engineer is the last person involved in the production of musical material, he or she must rely on that experience and discernment to make definitive decisions to best prestnt the music. The mastering engineer can be thought of as a literary editor. The written work is not original to the editor, but he or she will read and comprehend the material in order to quickly make decisions and make necessary alterations. The changes will allow the writing to be clear, concise, and ready for publication - creating a new and exciting experience for the reader.

One final note, what matters above all is that the mastering engineer produces the work exactly how the client envisioned. This was the philosophy behind the mastering of "All the Sadness." The mastering was meant to enhance the mix within the context of the intent of the artist and mixing engineer. The mastering did not change the song beyond recognition, but instead attempted to carry the mix along the same aesthetic projection, thus allowing the mix to realize its fullest artistic potential.

In conclusion, the world of mastering may seem mysterious due to its personal and subjective nature. Nonetheless, for anyone going into a career in the music industry, whether as an artist, technician, producer, or engineer, it is

important to attempt to understand all of the stages of musical production especially the final step of mastering.

Appendix

Session 1 Photos: November 15, 2015

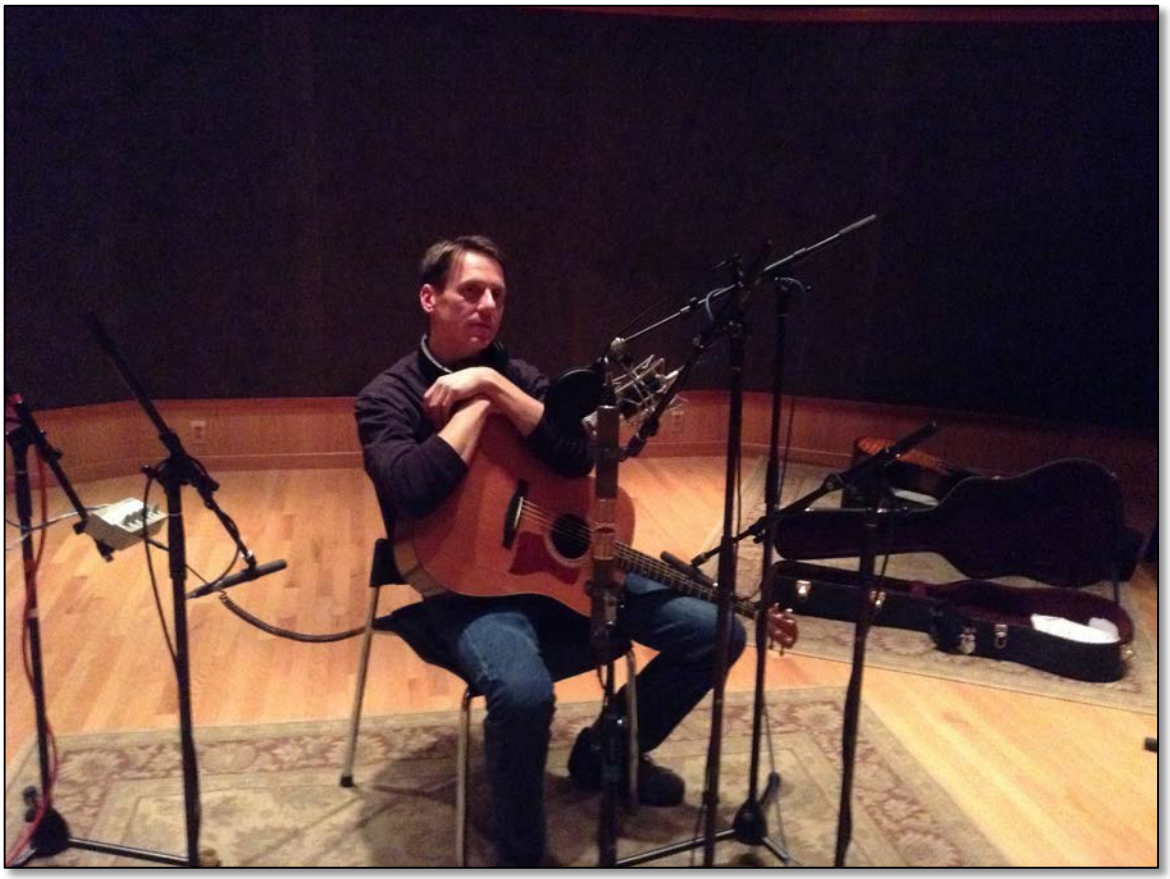

Figure A-5. Main Guitar Set-up

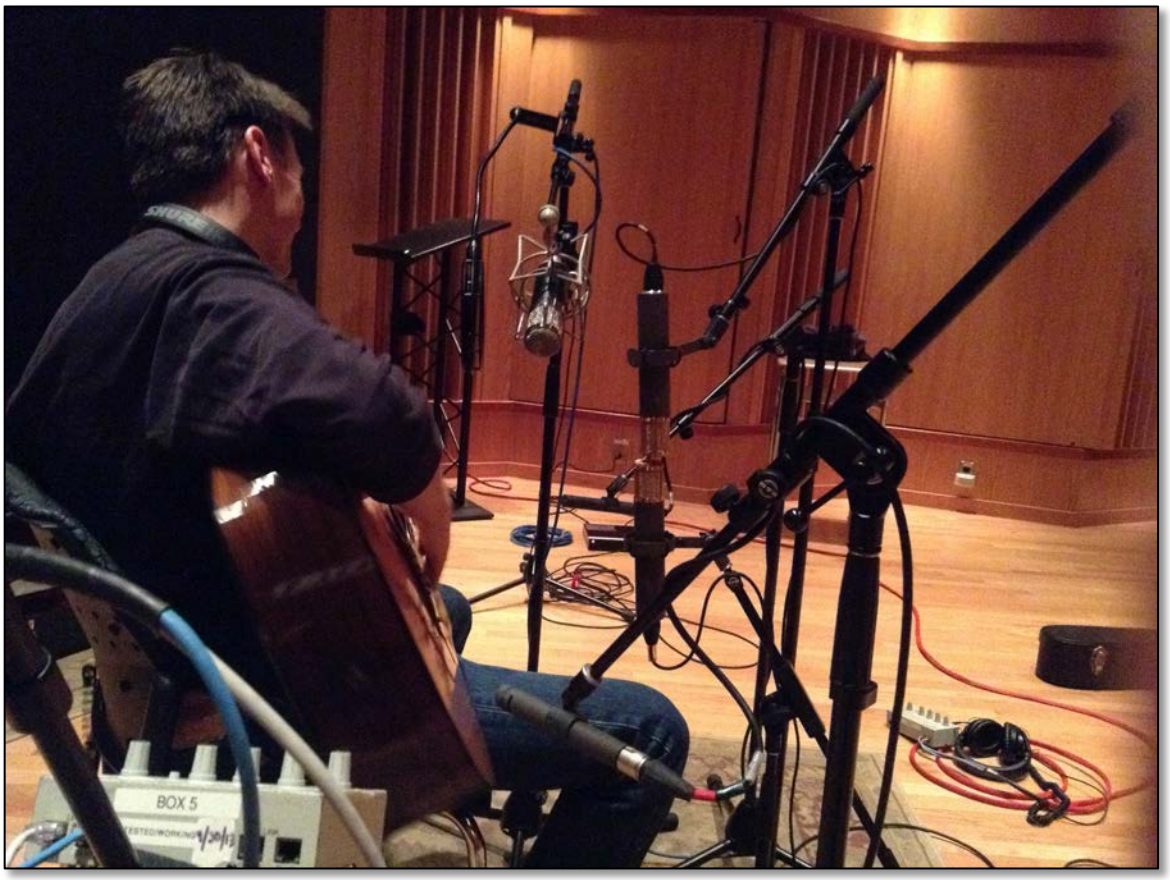

Figure A-6. Guitar Side View

Figure A-7. Guitar Alt. Side View

Figure A-8. Aux Guitar Dub

Figure A-9. Aux Guitar Dub Side View

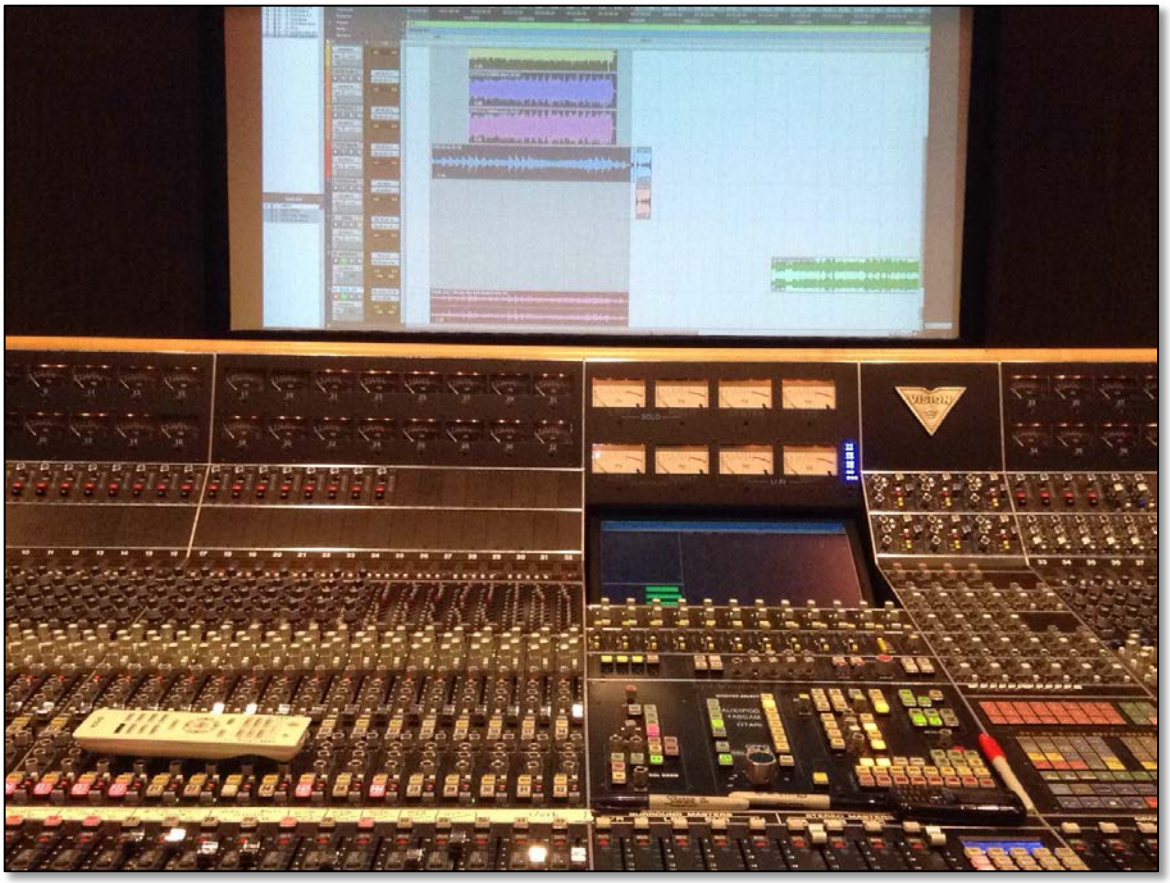

Figure A-10. API Vision Console and Protools Session

Session 2 Photos: November 20, 2015

Figure A-11. Vocal Mic Setup

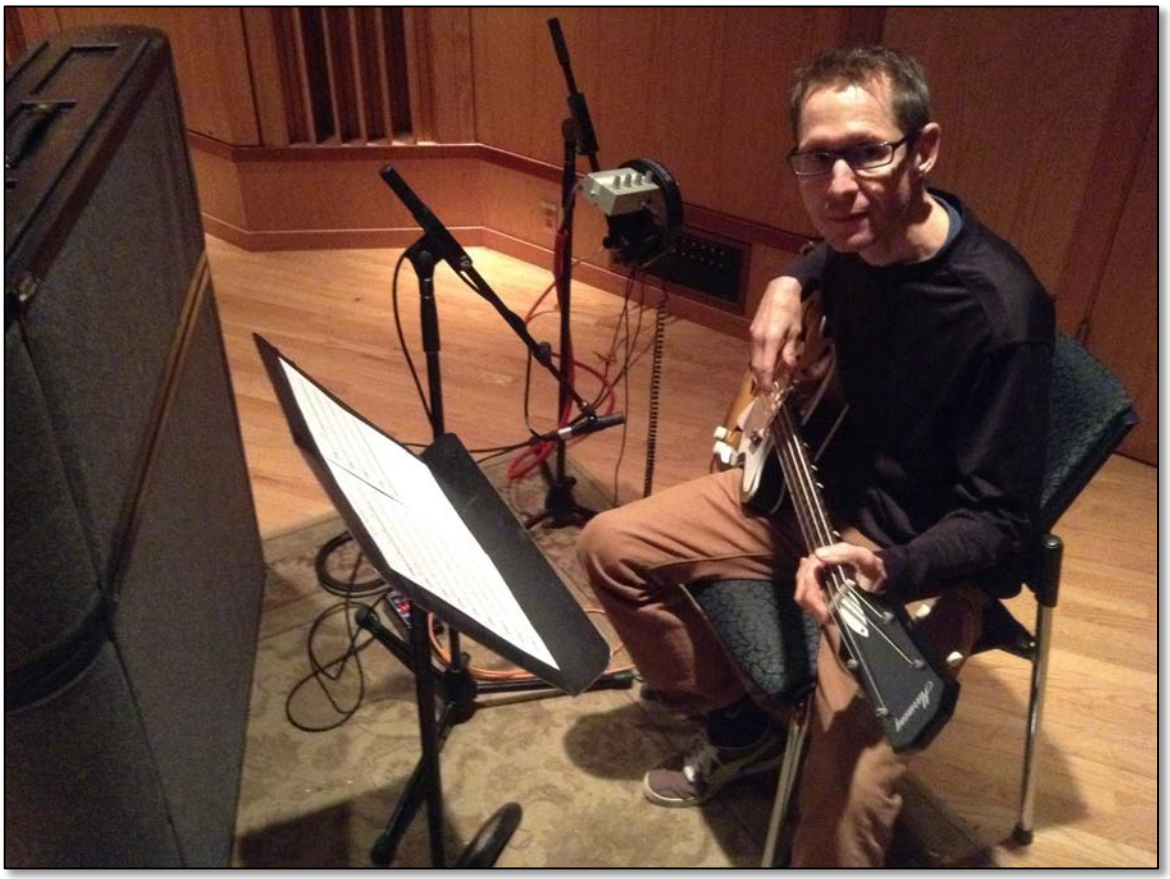

Figure A-12. Bass Guitar

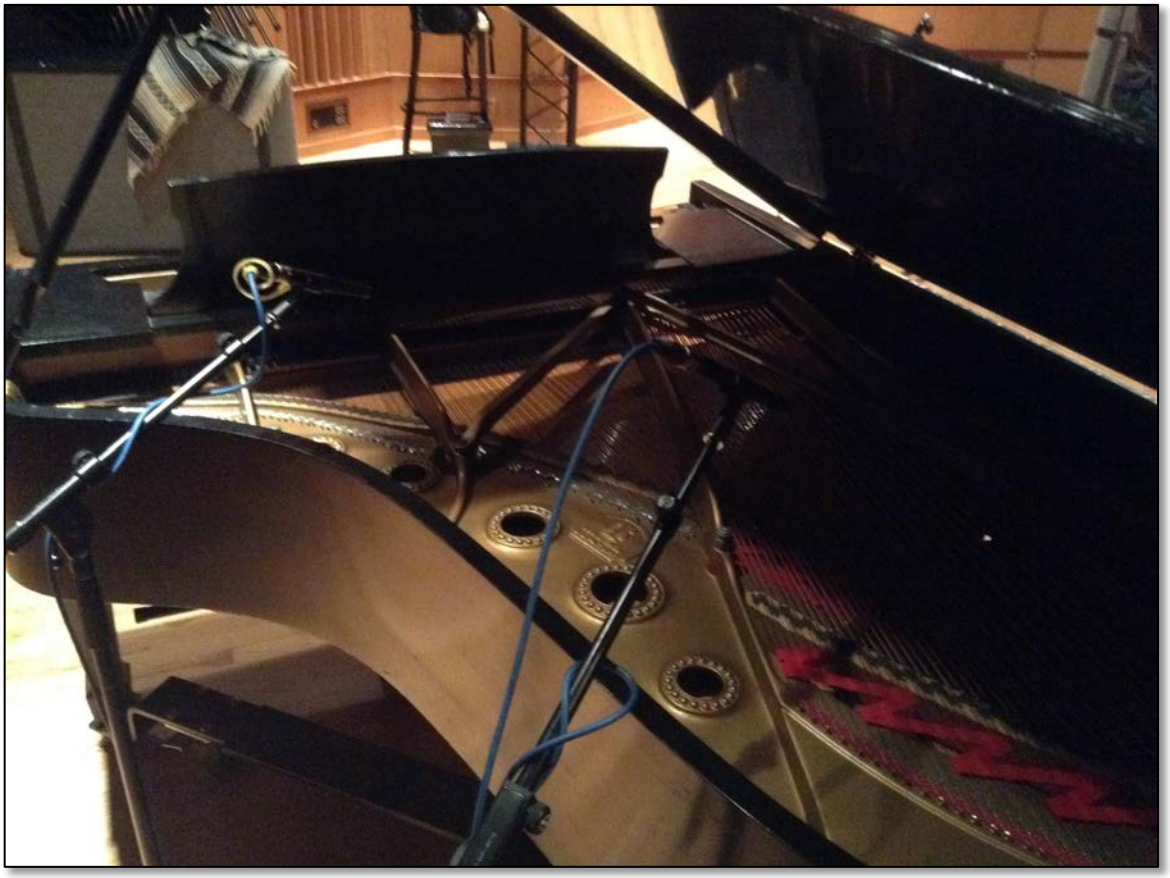

Figure A-13. Piano

Session 3 Photos: November 21, 2015

Figure A-14. Birdseye View of String Session

Figure A-15. Microphone Setup and Director Williams

Figure A-16. String Mic Array

Figure A-17. String Session Director's View

Figure A-18. String Quartet Setup

Figure A-19. Violin Parts Overdub Setup

Figure A-20. Session Engineers Ethan King (left) and Karl Fleck (right)

Session 4 Photos: December 8, 2015

Figure A-21. Electric Gutiar Setup with Darleen in the Hall

Figure A-22. Guitar Amp Mic Setup with Room

Figure A-23. Guitar Amp Close Mics

Lyrics

All the Sadness
Music and Lyrics by Alan Williams

Verse 1:

I never meant to leave you
I lost the key to your door
I never tried to hurt you
I just can't say I'm sorry anymore

Half-Chorus:

Summer rains fall heavy
Winter winds sure to come

Verse 2:

And in the morning after
First the sting then the numb
And every song a reminder
Of all the sadness to come

Chorus 1:

Summer rains fall heavy
Autumn leaves tumble slow
Winter winds will cut you
And leave you bleeding in the snow

Instrumental Section

Chorus 2:

Summer rains fall heavy
Autumn leaves tumble slow
Winter winds blow through me
Carve your name in my bones

Outro Verse:

I never meant to leave you
I just can't say I'm sorry anymore

Biographical Sketch

Karl Fleck graduated from Hope College in 2013 with Bachelor of Arts degree in Recording Arts. Post matriculation, Karl worked as a recordist and archivist at Hope College. He also operated as a freelance audio engineer in west Michigan. Alongside audio engineering, Karl is an avid musician primarily enjoying the mandolin.