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UNIVERSITY OF MIAMI

VOCAL RECORDING TECHNIQUES FOR THE MODERN DIGITAL STUDIO

By

Kelly Kristy Garner

A DOCTORAL ESSAY

Submitted to the Faculty
of the University of Miami
in partial fulfillment of the requirements for
the degree of Doctor of Musical Arts

Coral Gables, Florida

May 2014

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UNIVERSITY OF MIAMI

A doctoral essay submitted in partial fulfillment of
the requirements for the degree of
Doctor of Musical Arts

VOCAL RECORDING TECHNIQUES FOR THE MODERN DIGITAL STUDIO

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Vocal Recording Techniques for the Modern Digital Studio (May 2014)

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In today's recording studio environment, singers are standing behind some of the very same vintage microphones and using some of the same outboard gear as during the analog era. However, recording sessions are no longer being driven by analog multi-track recorders; instead, the session information is usually being saved onto a PCI computer card housed in a very fast computer. Processed through the analog vocal chain of microphone, pre-amplifier, compressor, and equalizer, the signal then passes through an A/D converter where the analog signal is transformed into digital information, a collection of samples represented mathematically by zeros and ones.

During the recording stage, careful consideration must be made regarding microphone choice, studio set-up, gear choice and performance techniques vocals, including vocal percussion. Vocal editing, vocal tuning, pre-mix preparation and reference mixing are not technically a part of the actual recording process, but will be briefly discussed as topics that need some amount of consideration since they all play a role in affecting the final performance of the recorded vocal.

Recording in the digital platform is quite different from the analog platform used in decades of the past. The simplest explanation of those differences can be boiled down to analog recording representing a continuous picture of the sound as oppose to the snapshots that are taken in the digital platform. Obviously, with such a concept, some

information within the audio being recorded has the potential of being left out or lost between "snapshots."

That said, the end goal of recording a vocal is to attempt to capture the vocal as accurately and pristinely as it exists acoustically in nature, there exists a multitude of ways that an engineer/producer can affect the actual performance of the recorded vocal. These range from the choice of equipment being selected as it is recorded or edited to the things that the engineer/producer might say to the singer in order to elicit the singers' best performance.

The purpose of this study is to identify and discuss techniques for recording the Voice within the modern digital studio environment. An attempt will be made to trace the process of recording the vocal from studio set-up up until, but not including, the formal mixing stage. In preparation for recording the solo vocal or vocal group, the audio engineer must also make some decisions about which microphone, microphone pre-amp, compressor, whether or not there is any need to use an equalizer, and proximity/positioning of the microphone. Recording vocal percussion is also a technique that can require a different set-up and skill set from the engineer depending on the technique of the performer. Several ideas and concepts for recording vocal percussion will also be considered.

Last but not least, all the techniques discussed will be demonstrated in a recording so that one can discern sonic differences between use and non-use of a particular technique. The main theme of this discussion will be identifying and discovering strategies to improve the quality of recorded vocals in the digital domain.

DEDICATION

This work is dedicated to my loving and supportive parents, James R. Garner and Martha Yancey Garner. It is also dedicated to the memory of my grandparents, Corene D. Garner, G.W. "Hoyt" Garner, K. C. "Cecil" Yancey, Sr. and especially my grandmother Elise K. Yancey, who as a valiant educator never ceased to push me beyond my limit and encourage me to believe that with God's help, I could achieve even the things that seemed to be impossible!

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CHAPTER 1

INTRODUCTION

The initial experience for a singer standing behind a microphone to record a vocal track, also known as a "vocal," can certainly be intimidating. Merely hearing oneself inhale on a microphone for the first time reveals the startling clarity with which a vocal microphone captures the slightest sonic detail. Thirty-five years ago, if a young vocal artist were to record a vocal track in a state-of-the-art studio, they would be doing so on two-inch tape winding meticulously through a twenty-four track, multi-track reel-to-reel recorder.[1]

In today's recording studio environment, singers are standing behind some of the very same vocal microphones, positioned the very same way, and using some of the same outboard gear.[2] However, recording sessions are no longer being driven by those analog multi-track recorders; instead, the session information is usually being saved onto a PCI computer card housed in a very fast computer. Processed through the analog vocal chain of microphone, pre-amplifier, compressor, and equalizer, the signal then passes through an A/D converter where the analog signal is transformed into digital information, a collection of samples represented mathematically by zeros and ones.[3]

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1. Paulo Jorge Botelho, "The Sound of the Medium" (PhD diss., Princeton University, 2008), In ProQuest Dissertations and Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/304508207?accountid=14585 (accessed October 14, 2012), 101.

2. John Eargle, *Handbook of Recording Engineering*, 4th ed. (Boston: Kluwer Academic Publishers, 2003), 85.

3. William J. Phillips, "Making Tracks: Digital Recording Technology and the Democratization of Cultural Production" (PhD diss., New York University, 2010), In ProQuest Dissertations and Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/763609104?accountid=14585 (accessed October 14, 2012), 212.

There are also additional considerations and techniques to consider in recording vocals, some of which worked in the past and that continue to work now, others which have changed because of innovations and new concepts. During the recording stage, careful consideration must be made regarding microphone choice, studio set-up, gear choice and performance techniques for group or solo recorded vocals, as well as vocal percussion. Vocal editing, vocal tuning, pre-mix preparation and reference mixing are not technically a part of the actual recording process, but will be briefly discussed as topics that should be considered since they, can potentially, all play a role in the ultimate result of the recorded vocal.

Recording in the digital platform is quite different from the analog platform used in decades of the past. The simplest explanation of those differences can be boiled down to analog recording being a continuous picture of the sound as oppose to the snapshots that are taken in the digital platform. Obviously, with such a concept, some information within the audio being recorded has the potential of being left out or lost between "snapshots." This is a discussion that will be more thoroughly considered in a later chapter.

In addition to the discrepancies between digital and analog waveforms, the nature of condenser microphones is one of adding a small amount of distortion to the recorded signal.[4] Ribbon microphones do not tend to add this quality, but bring other challenges to the table like darker color and microphone fragility.[5] So with vocals recorded in the digital domain, the goal is to try to get the sound of the recorded vocal as close to the

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4. Tom Lubin, *Getting Great Sounds: the Microphone Book* (Portland: Course Technology, 2010), 34.

5. Ibid., 35.

actual acoustic version as possible. There are techniques and ways to use a vocal microphone, a microphone pre-amplifier, compressor and equalizer that can begin to alleviate the "digital sound" obstacles in the initial stage of actually recording the vocal. The end goal of recording a vocal is to attempt to capture the vocal as accurately and pristinely as it exists acoustically in nature.

Recording a vocal is one of the audio engineer's most difficult tasks because of variations in dynamics and color, not to mention editing around a singer's breath, room noise and studio acoustical factors. There exists a multitude of techniques that engineers and producers have available at their disposal to affect the actual performance of the recorded vocal. These range from the choice of equipment being selected as it is recorded or edited to the things that the engineer/producer might say to the singer in order to elicit the singers' best performance. Capturing a "best" performance in a recording is quite a formidable task and in many ways a team effort. There are many contributory factors in recording a vocal that emerge from "both sides of the glass" so to speak. This concept will also be discussed further in a later chapter.

The purpose of this study is to identify and discuss techniques for recording the voice within the modern digital studio environment. The process of recording a vocal will be traced from studio set-up up until, but not including, the formal mixing stage. This study will begin by discussing the studio set-up and all preparations that must be made to prepare the session files for recording the solo vocal or vocal group. The way in which the singers perform behind the microphone in the midst of the session is also an important consideration. The elements of proximity to the microphone and positioning amongst group members will be discussed along with the different permutations of these

positions that can be used throughout the session to affect the recorded sound. In preparation for recording the solo vocal or group of vocals, the audio engineer must also make some decisions about which microphone, microphone pre-amp, compressor and whether or not there is any need to use an equalizer. Recording vocal percussion is also a technique that can require different microphone set-up and skill set from the engineer depending on the approach or technique of the performer. Several ideas and concepts for recording vocal percussion will be considered.

The process of recording a vocal track in the digital domain includes the act of converting the analog signal into digital information. These two worlds being so dramatically different in character will be discussed as well as options to possibly incorporate the best of both platforms into the process. Possible options and many other techniques will be considered through email interviews to industry professionals from Nashville, New York and Los Angeles.

Last but not least, all these techniques that are discussed will be demonstrated in a recording so that one should clearly be able to tell the difference between use and non-use of a particular technique. The main theme of this discussion will be finding strategies to improve the recording quality in vocals so that the best characteristics can be implemented to improve the sound of recorded vocals in the digital domain. Even though there are audio engineers using some or all of these techniques, it is this author's hope to present a resource that will function as a useful guide for vocal recording.

There are several questions that will aid our investigation going forward as we begin to have discussion with experts in the field of artist production and audio engineering. First of all, in what configuration does a recording engineer usually set up

the vocal booth to record vocals? Furthermore, in what shape should the microphones be set-up for group vocals as opposed to solo vocals and how close should each group member or microphone be positioned?

Secondly, what are some characteristics of different voice-types present in representative spectrograph analyses that might aid in an attempt to match the perfect microphone with individual voices?

Thirdly, what microphones are typically used for recording particular voice-types and why are those pairings more frequently used?

Fourthly, what is the best "vocal chain" combination of microphone, pre-amplifier, compressor and equalizer for vocals to be recorded as true to the acoustic vocal as possible? What frequency characteristics should be considered during set-up by the engineer as equipment is being chosen for the vocal chain?

Fifthly, in what way does the decision processes and atmosphere created by the producer and/or engineer affect the performance of the vocal?

Sixthly, what happens to an analog vocal when it is converted into a digital waveform? Is it possible for analog signal information that might be lost in the A/D conversion process to be recreated artificially in the digital domain? If so, what are some things engineers and/or producers can do to recreate or account for the lost information?

Lastly, in what ways can the vocal track be manipulated in the vocal editing and vocal tuning process and should this be considered part of the performance? Do these manipulations affect the level of musical integrity or is this now just considered a part of the modern-day performance of the entire production team?

Reference mixes are created just prior to the mix stage to allow the producer or artist to have a accurate sense of what kind of vocal they have recorded and what the mix might sound like in order to decide if indeed the recording process is complete. Pre-mix preparations include cleaning up tracks and preparing what is on the session to be easily mixed by the mix engineer without any confusion. Both of these late stage pre-mix concepts are addressed briefly in a later chapter.

The study only includes information about the act of recording vocals as it pertains to the recording studio. Recording "live" vocals are not a part of this discussion. There is no specific genre of performers or vocal groups chosen as a focus, but this document contains more of a general discussion about vocal recording with techniques that could be applied to any genre. Some discussion comes from professionals that have worked in different genres, but have been asked questions that pertain to general aspects of vocal recording, not genre specific.

CHAPTER 2

REVIEW OF LITERATURE

The literature reviewed for this paper is taken from literature that has been written about various aspects of the vocal recording process. There have been peer-reviewed works written on all the following sub-groups included in this study: the history and process of vocal recording, microphones, the progression from analog to digital, as well as vocal editing. This study will review the literature within each section and give a synopsis of important findings found in each body of work.

Vocal Recording

Inventing a device that had the ability to capture the human voice is an event that changed the course of history. Phillips describes Edison's phonograph invention going from being an office dictation machine to eventually paving the way for the world to have pre-recorded music.[6] Another invention that had a great impact was that of the vacuum tube microphone. It facilitated higher fidelity playback. When magnetic tape hit the stage, it allowed for the unprecedented ability to edit and manipulate recorded performances. Thomas Edison invented the phonograph and thus single-handedly changed the course of history in the recording industry.

The phonograph became a central figure in the attempt to elevate American music through the dissemination of recorded music.[7] The possibilities and the limitations of

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6. William J. Phillips, "Making Tracks: Digital Recording Technology and the Democratization of Cultural Production" (PhD diss., New York University, 2010), In ProQuest Dissertations and Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/763609104?accountid=14585 (accessed October 14, 2012), 212.

7. Steven George Jones, "Rock Formation: Popular Music and the Technology of Sound Recording" (PhD diss., University of Illinois at Urbana-Champaign, 1987), In ProQuest Dissertations and

early recording technology shaped almost every aspect of jazz performance and composition. The impact of recording is strongly shaped by the time, place, and cultural context in which the technology is used.[8] Prior to the invention of the phonograph, Karl Marx observed an unchangeable truth about music,

> The service the singer performs for me satisfies my aesthetic need, but what I consume exists only in a(n) action inseparable from the singer, and as soon as the singing is over, so too is my consumption. When sound is recorded and preserved in a physical medium, however, the listener's consumption need not end when the singing is over, for the music can be separated from the performer and be replayed without the artist's consent.[9]

The portability and repeatability of recorded sound are two of technologies crucial attributes. In 1877, the New York Times proposed that people in the future would seek out cylinder recordings of great speeches of the day in the same way they would collect fine wines.[10]

Howard Massey describes the habits of a vast array of producers that have worked with the most popular artists in the world. Some of these producers include George Martin, Glen Ballard, Brian Wilson, Arif Mardin and Phil Ramone. He tells personal stories as to instances each of these producers have had with artists while they were recording the entire album, including while they were recording vocals and some of

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Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/303472903?accountid=14585 (accessed October 14, 2012), 143.

8. Steven George Jones, "Rock Formation: Popular Music and the Technology of Sound Recording" (PhD diss., University of Illinois at Urbana-Champaign, 1987), In ProQuest Dissertations and Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/303472903?accountid=14585 (accessed October 14, 2012), 144.

9. Mark Katz, *Capturing Sound: How Technology Has Changed Music* (Berkeley: University of California Press, 2004), 21.

10. Ibid., 22.

the gear that was used in the process.[11] For instance, Glen Ballard has a wide assortment of microphones that he owns personally including an AKG C12, Neumann M49, Neumann U47, and a Neumann U67. He used the C12 on Alanis Morissette and loved it so much that he never even tried another mic. He also used medium compression and monitored the level on her fader continuously. Massey mentions that Ballard knew that on Alanis' "You Oughta Know" he scorched the vocal in a few spots, but everyone loved it so much and she was such a "one-take" singer, they all decided to just keep it the way it was. He recounts Al Schmitt's approach to recording vocals including how he thinks a singer should stay nine inches to a foot away from a vocal microphone. He says the diaphragm will not respond correctly, but always tries to make sure its recorded on-axis. He also talks about the vocal chain that he tends to use most frequently.[12] His approach is one perspective but others disagree completely and think that the presence of the voice cannot be adequately captured unless the singer is within three to four inches from the diaphragm and on-axis.

Microphones and The Vocal Chain

Microphones are the first stage in the amplification and processing of an acoustic sound. They capture every atmospheric pressure that passes by them. Microphones are transducers, which simply means that they change acoustic sound vibration into an oscillating electrical signal. They transduce the meaning and emotion in the singers' voice. So clearly, the goal of choosing the correct microphone is in trying to capture the most

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11. Howard Massey, *Behind the Glass: Top Record Producers Tell How They Craft the Hits* (San Francisco: Miller Freeman Books, 2000), 32.

12. Ibid., 33.

realistic and genuine version of the singers' performance as possible. Choosing the right microphone is like an artist choosing the correct pencil or brush. One would not want the artist to feel limited in creating their art just by the type of brush or pencil they will be using as they try to draw the picture they have already formulated in their minds' eye.[13] Condenser microphones are most commonly used for recording vocals and there is a vast array of different microphones available. A step beyond choosing the correct microphone lies another part of what is called the vocal chain. This chain consists of microphone, microphone pre-amplifier, compressor and optional equalizer. The careful pairing of different types of these components can result in a recorded vocal that is as close to the analog acoustic vocal as can be possible in the digital age. Choosing the best microphone and vocal chain components for each individual voice is the most important step in recording a vocal. If this is done correctly, a true representation of that voice will be achieved.[14] A microphone pre-amplifier that blends well within the vocal chain, might not have exactly the same characteristics in frequency response as the microphone. Sometimes a flat responding microphone will be paired with a microphone pre-amplifier that contains more of a colored sound. In the same way, compressors will sometimes add a color to the tone as well. This supports the argument of attempting to find the perfect vocal chain combination for individual voices.[15]

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13. Tom Lubin, *Getting Great Sounds; the Microphone Book* (Portland: Course Technology, 2010), 2.

14. Ibid., 3.

15. Alexander U. Case, *Sound FX – Unlocking the Creative Potential of Recording Studio Effects* (Oxford: Focal Press, 2007), 42.

The Evolution from Analog to Digital

The human voice is considered, as is any other acoustic instrument, to be an analog signal. There are a variety of differences between analog and digital as well as the use of analog equipment to color digital signals.[16] These as well as other aspects of recording will be reviewed throughout this section.

Botelho discusses strategies for tracks that have been recorded in the digital domain to sound more like they were recorded in analog. He begins the discussion of this section by suggesting use of a tube pre-amp and an analog compressor.[17] He suggests a FMR Audio RNC1773 as an analog compressor that is surprisingly good. He also says that another possibility is to run your signal through analog guitar effect pedals, but this approach can add some noise to the signal. The amount to which this changes the sound of the recorded signal depends on the gain levels and/or the noise gate. If an instrument has already been digitally tracked, the sound can be sent back down the analog signal path, basically retracking it to add some analog warmth. However, in using this procedure, an offset might have to be incorporated onto that particular recorded track because of delay that might have been added as the signal looped out and back into the system. Botelho also interviewed Mac Quayle, a mix engineer that has remixed Madonna, Sting Annie Lenox and Beyonce.[18] Quayle started mixing with an analog mix bus. He basically takes eight stereo mix pairs out of his Pro Tools studio set-up and runs those outputs into a Dangerous Two-Bus analog summing mixer designed specifically for

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16. Mark Katz, *Capturing Sound: How Technology Has Changed Music* (Berkeley: University of California Press, 2004), 213.

17. Paulo Jorge Botelho, "The Sound of the Medium" (PhD diss., Princeton University, 2008), In ProQuest Dissertations and Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/304508207?accountid=14585 (accessed October 14, 2012), 187.

18. Ibid., 189.

this purpose. The mixer does not do any processing, it just adds the outputs together using the high quality analog mixing bus. He then takes the stereo mix coming out of the analog summing mixer, goes into an analog compressor and then back into the digital world through the analog to digital converters of a TC Electronics finalizer. This technique makes even reference mixing sound bigger and warmer and also creates better stereo imaging. He says it is still not quite the same as using two-inch tape, but its certainly better than doing it all through the computer. Botelho makes some good points and suggestions for recapturing the warmth of analog in the digital world.[19]

Analog is called "analog" specifically because of the way the electricity vibrates in the circuit. The vibration is analogous to the way sound vibrates in an acoustic instrument.[20] In analog, you get a smooth waveform; whereas in digital, you get a stair-step effect in the waveform. This stair-step effect can be smoothed or reduced a little with various techniques, but nothing will completely replicate the smooth waveform of analog. The reason the digital wave appears as steps is because, as the recording is taken, slices are taken down and each slice of time is recorded and represented with a number. From one number to the next, the numbers change. Therefore you get the stair-step effect. Compare the difference between analog and digital with fluorescent and incandescent lighting. The fluorescent lighting is supposedly not a natural light, not containing the full spectrum of light. In the same way, digital does not contain all the

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19. Paulo Jorge Botelho, "The Sound of the Medium" (PhD diss., Princeton University, 2008), In ProQuest Dissertations and Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/304508207?accountid=14585 (accessed October 14, 2012), 190.

20. Steven George Jones, "Rock Formation: Popular Music and the Technology of Sound Recording" (PhD diss., University of Illinois at Urbana-Champaign, 1987), In ProQuest Dissertations & Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/303472903?accountid=14585 (accessed October 14, 2012), 74.

information either.[21] An analog synthesizer gets its sound from peculiar characteristics of the analog device and electronics that tend to distort the waveforms in ways that are warm and pleasant. One could speak of the sound coming from the vocal cords in the same way, as sound is being reshaped by the mouth, throat and other resonating vocal cavities. These very distinct ways of distortion tend to be very difficult to emulate with digital circuitry and impossible to duplicate exactly. There is always going to be a different resulting waveform being generated by a digital instrument trying to emulate an analog instrument as compared to one that is actually being produced by a real analog instrument.[22] One major advantage of the digital platform is the ability to program instruments that could never be programmed in an analog environment and the option to audition virtual instruments quickly without having to spend a lot of money purchasing an analog instrument.[23]

Vocal Editing

Before the digital audio workstation era began, intonation problems were handled during the tracking stage. There was a continuous struggle between using a track with great emotion but with at least one note out of tune, compared to a track that was in tune but with no emotion. In the digital world, it is no longer an issue as to what can be corrected in the vocal track. Almost everything can be fixed, but the issue now exists of how much musical integrity does one want to maintain in creating their art. Although this is a very personal decision, the fact still remains that almost anything can be done to

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21. Steven George Jones, "Rock Formation: Popular Music and the Technology of Sound Recording" (PhD diss., University of Illinois at Urbana-Champaign, 1987), In ProQuest Dissertations & Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/303472903?accountid=14585 (accessed October 14, 2012), 75.

22. Ibid.

23. Ibid., 76.

correct issues in the recorded vocal. Gibson describes the use of a plug-in on the vocal track as being one way to fix the intonation. The plug-in he likes to use is Auto-tune by Antares. He also discusses using a light amount of compression as a plug-in on the track in order to just make the levels more consistent while tracking.[24]

24. Bill Gibson, *The S.M.A.R.T. Guide to Mixing and Mastering Audio Recordings* (Boston, MA: Thomson Course Technology, 2006), 56.

CHAPTER 3

METHODOLOGY

The purpose of this study was to explore techniques for recording vocals in the modern digital studio. Although the process for doing so is multi-faceted and complex, the intent of this study was to offer a step-by-step analysis of the most current viable options for recording vocals in the modern studio environment. Even though many other instruments are also recorded in the modern studio, this paper only discusses the various techniques surrounding vocal recording.

The vocal recording process begins with setting up a microphone and ends after vocal editing and vocal tuning have confirmed that there have been enough vocals recorded to proceed to pre-mix preparations. A variety of microphones is discussed along with their specific nature and character of sound in regards to their use for recording vocals. The "vocal chain" as it is called is discussed sequentially as well as the human element that involves the performer stepping in front of the microphone, proximity of the performer to the microphone and to each other in a group setting. The vocal chain is also greatly affected by the particular microphone pre-amp, compression, equalization, and analog to digital converter that is used to convert the analog vocal into the digital realm being housed in the computer. Another performance aspect discussed includes the involvement of the producer and/or engineer in terms of how and to what extent their actions affect the recording.

This study does not include any information about "live" recordings or any gear that might be utilized specifically for "live" recording situations. The conversion of the

analog into the digital domain is an area that has been approached from many different angles as the conversion from analog to digital is the primary place where the vocal tends to change character and sound somewhat different from the actual live acoustic sound. The difference between analog and digital sounds are discussed and the problem of digital voice files tending to lose some of the warmth of sound sonically are also considered, as well as possible solutions.

Once the vocal has been recorded into the digital domain, there are additional techniques discussed from an assortment of plug-ins, such as equalization, compression, and harmonic enhancements. There are also tools such as Vocalign and Melodyne that are helpful in the editing process just prior to mix. Vocalign is a program that will align group vocals at the click of a button and Melodyne is a vocal tuning program that is also capable of multiple editing techniques like moving and elongating vocal notes. Both these programs are considered the industry standard and are briefly discussed. Some of these editing plug-ins and stand-alone programs are controversial as to how much they insure or destroy musical integrity, but nonetheless are considered and included as possible tools following the vocal recording process.

Although the mix process is not discussed in this context, reference mixing as it pertains to insuring the best recorded vocal and conclusion to the process is briefly considered. Once the vocals get to the mix stage there is also the spectrum element to consider and where one might place the different vocals and voice parts across the stereo spectrum with panning. This panning placement, which affects how spacious the vocals sound, is also effectively used in reference mixing to allow the producer to hear what has been recorded and if it is going to be enough. This study includes a recorded sample of

raw vocals with no editing or manipulation compared and contrasted with the same recording just prior to the mixing stage. A recording of different vocal chain combinations paired with particular voice-types are also demonstrated and included in this study.

Questions were posed to experts in the field of audio engineering and artist production. The following questions were sent to these professionals in the form of an email questionnaire:

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

4. What is your favorite microphone set-up for a solo vocal and for group vocals?

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

6. What is your approach to recording vocal percussion?

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

The following engineers or industry professionals responded to an email questionnaire regarding all of the above questions: Ed Boyer, Bob Clark, David Hall, Matt Pierson, Tom Reeves, Dave Sperandio, Paul Tavenner, Paul Wickliffe, and Nathan Zwald. These individuals are all either professional producer/engineers or producer/artists working in Los Angeles, Nashville or New York.

CHAPTER 4

STUDIO SET-UP AND SESSION PREPARATION

The manner, in which the studio is set-up and the preparation of the session within the digital audio workstation, is paramount for insuring a successful recording session. There are a few different ways to set up the room in which the singers will be singing. First of all, they need to be facing the control room window in order to see cues from the producer and/or the engineer. In a studio where there is no control room window, the cues can sometimes come on a count track that the producer prepares ahead of time. Also the producer might have to step onto the recording stage or into the vocal booth if seeing a cue is of utmost importance. It is a good idea for the engineer to set up an additional set of headphones in the room with the singers for this very reason. The engineer should also always have an extra cue station set up in the control room so that he can hear exactly what the singers are hearing.

Secondly, there are several different ways for setting up the microphone configuration, but the most important component is for each singer to have their own microphone. The studio may or may not have enough condenser microphones for a "one person per mic" scenario, depending on the size of the vocal group and the size of the studio. If the studio does not have enough condensers, then two singers who are singing the same part would be acceptable on the same microphone. The main issue with putting more than one singer on a microphone is that the ability to make the correct choices for the vocal chain are greatly diminished. The way to achieve the perfect combination is to choose a microphone that compliments the particular voice-type, followed by the best

combinations of microphone pre-amplifier, compressor and equalization if needed. Putting two singers on one microphone makes this process very difficult to do. A microphone that is generally a good choice for females, or generally a good choice for males, might be the only choice in this situation. There will be more discussion in a later chapter about voice-types, microphone selection and the best choices for the vocal chain.

If the group of singers is a large vocal ensemble (nine to sixteen voices), the best microphone set-up is to put them in a large circle. In this configuration, they can all see each other and at least some of them will be able to see cues from the control room if needed. The recording studio that is being used may or may not be big enough for this many people, so that needs to be considered prior to booking the session. If the ensemble being recorded is a smaller group (three to eight voices), it is typical to position the microphones in a semi-circle facing the control room window so that they all can see cues from the producer. With two singers, a "v-shape" so that they can still see each other and also see the producer is the best option. When recording a single voice, the singer should be positioned directly in front of the control room window, but not so close that the microphone picks up a "slap-back" soundwaves from the voice's soundwaves hitting the control room window and bouncing back to be picked up by the microphone. For this reason, some find it helpful to either position the singer a little further away from the glass or angle the singer and microphone slightly so that the diaphragm of the microphone is not sitting parallel to the control room glass. As a sidenote, the control room window glass is usually angled down at the floor on both sides and each piece of glass is at a different thickness in order to keep sounds from the control room from passing through the window. The angle of the glass should help with this "slap-back"

effect, but one can never be too careful and should always try to eliminate potential acoustic problems.

Acoustics within the room where the singers are singing is a very important consideration. If the recording studio is not already equipped with acoustical traps, there are some concepts that need to be considered and some make-shift acoustical traps that can be used. Try to avoid acoustically untreated parallel walls on either side of a group of microphones. Most professional studios do not have any parallel walls within the tracking room. In the case that the only recording studio available has parallel walls in the tracking room, try to place some object or some kind of blocking in front of the parallel wall so that the sound waves will diffuse enough to no longer be a problem. This will help eliminate any standing waves[25] between parallel walls. If there are any baffles that exist in the studio, those should be placed around the singers to create non-parallel false walls. These baffles are also usually filled with an absorptive and reflective material that will also aid in the acoustics of the room.

The best stands to use with heavy condenser microphones in the studio are either the Atlas SB36W Studio Boom or Ultimate Support MC-125 Professional Studio Boom. There may be others just as useful, but these have a very heavy base with extension segments that can be used if needed. The bases to these stands should all be positioned in the center of the circle where the singers will be standing. The stands should be positioned by coming up and over, hanging the microphones upside down. The biggest reason for using the stands and positioning the microphones in this manner is so that each singer can have a music stand directly in front of them sitting directly under each

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25. Standing waves in rooms can cause certain resonant frequencies to either be unduly enhanced (nodes) or completely disappear (antinodes).

microphone. The music stands should be outfitted with some kind of soft, absorptive material so that the "slap-back" affect from the hard surface of the stand is lessened. This is especially important with the stand being directly under the microphone. Each stand should have a stand light and all connections for these stand lights should be positioned in the center of the circle in the same place where the bases of the microphone stands are sitting. If the set-up of the microphone is coming up and over, the ability for the singer to turn pages in their music and make notes is much easier. The diaphragm of the microphone should be positioned at a height that is just below the base of the nose and just above the mouth, somewhat in line with the upper lip. So much of the singers' resonant sound comes not only from their mouth but also from the mask area of the face, this position of the microphone will best capture the true acoustic sound of the singer.

A pop-screen should be placed just in front of and about an inch away from the diaphragm of the microphone. On solo singers, a double pop-screen, with the second one making a "v-shape" with the first one, should be used. This will eliminate the plosives from making their way onto the track of the recorded lead vocal. With group vocals, plosives will be a little less noticeable, so one pop-screen should be adequate. The lead or solo vocalist should be no more than two or three inches away from the pop screen, but in a group setting the singer can be a little further away. The reason for the closeness of the solo or lead vocal is to capture as much presence in the voice as possible. Breath and sibilance, two things that seem to be less desirable are actually the very things that make a voice sound present and close to the listener. It creates a much more intimate quality to the singer's voice. There are some producers that prefer to position the microphone above the singer, tilted down and much further away. The problem that exists with this

position is two-fold. First, the microphone pre-amplifier has to be at a hotter level, which also creates a higher noise-floor, capturing more extraneaous room noise, etc. Also, the further one gets away from a condenser microphone there will be more high end and less low end frequencies. When recording a voice, it is best to try to capture as much warmth in the voice as possible. For this reason, the high and away approach for condenser microphone positioning doesn't seem to work quite as well. There is also some "presence" lost with this technique.

There should be some kind of headphone cue system that allows each singer to get a mix of enough of the vocal tracks coupled with an adequate amount of the stereo mix of the instruments. If possible, it is always helpful if the singer can have a "more me" potentiometer or "knob" in order to be able to hear themselves adequately. There are so many types of cue systems available, some better than others. The more "sends" that can be routed to the cue system, the more ability the performer will have to control what they are hearing. This will give the performer the greatest opportunity to record a vocal that is the best representation of what they are capable of recording.

The headphones that the engineer puts out for the singers should be ones that have ear cups that will give a good seal and keep leakage from the headphone cups from bleeding onto the microphone. The Beyerdynamic DT770-PRO-250 Dynamic Closed Back Headphones are some of the most popular and have a cushion seal around the cups that are pretty reliable for not leaking sound. There are several ways the singer can wear their headphones, but some seem to work better than others. Some people like to position one side of the headphones off of one ear and keep the other one on the other ear, others like to keep both sides of the headphones directly on each ear. Another technique is to

position on earphone in front of one ear approximately halfway off and the other earphone behind the other ear approximately halfway off. The main idea for the performer to remember with wearing headphones is to find the technique that helps them hear most efficiently. It is encouraged for vocal performers to try different approaches with headphone technique before determining what works best for them. The primary reason for taking either or both headphones slightly off is for the performer to be able to also hear themselves in the room acoustically as they usually do in practice. This headphone technique tends to help the singer's pitch and tuning.

The manner in which the recording session is prepared or set-up within the DAW software application by the recording engineer is also an important consideration that will affect the vocal performance. At the point in which vocals are recorded, one could assume that the instrumental tracks already exist within the session unless the song being recorded is an "a capella" arrangement. With that in mind, the instrument tracks should be routed through one particular stereo "send" channel, one that will show up on a stereo pair of the singer's cue box. If there is not enough channels for the "send" to show up as a stereo track, it can also be sent to a mono channel, as long as the singers can adequately hear the tracks. The singers will also need to be sent their vocals on a different channel that they can again control the levels with potentiometers. If there is the ability for the engineer to give them separate "more me" potentiometers, then that would be optimum. This allows the singer to have the instrumental track, the other singers and themselves on separate channels so that they can raise each level to whatever makes them most comfortable while singing.

The engineer will also probably want to create groups for the vocals as a whole and for individual sections (i.e. soprano, alto, tenor and bass). This will allow the engineer to quickly move between voice parts that might need to "punch in" a particular section or part of the arrangement. When a group is engaged within the DAW, the engineer can select one of the group members to record and all the tracks for that group will be simultaneously engaged to record. This concept is the same for engaging mutes and solos. Moreover, the group attributes can be configured to include whatever attributes the engineer desires.

Another helpful technique for the engineer to use in session set-up is that of putting in section markers (i.e. intro, verse 1, chorus 1, verse 2, chorus 2, bridge, chorus 3 and outro). It is also a good idea to input measure numbers within the "markers" window throughout the entire arrangement. Singers like to have a consistent lead-in or "runway" before the record light is on and they are recording. If the engineer has measure numbers easily readable, the engineer in grid mode with single measures selected can highlight two measures prior to the "punch-in" point and know that the track will begin playing exactly two measures prior to the "punch-in" point. These techniques will greatly affect the speed at which the session proceeds and will give the producer and performers more of a freedom to direct their attention to creative concepts instead of being concerned with technical issues or delays. Every engineer needs to be prepared to catch a singers' good idea at a moments notice, otherwise that good idea could fade or be forgotten. An engineers' performance and particularly the speed at which he operates can greatly affect the quality of vocal the performers are able to deliver. A creative atmosphere between the artist and producer is also much easier to maintain when there is fewer technical

obstacles and if the engineer can almost seem to be invisible, keeping everything working on his end without any problems or issues. Many producers, of course, often ask the engineer their opinion about certain takes or other questions that arise, but the engineers primary responsibility is to keep the session running smoothly without any issues and to capture the most pristinely recorded vocal sound possible.

Another option or amenity for the engineer to provide the singers in their headphone mix is reverb. Even though many singers like to hear themselves dry without any reverb while recording, some singers want to hear a little reverb on their voices just to have an idea of the end result. This effect should always be sent on a different "send" if possible, so that each singer would have the ability to use as much or as little reverb as they prefer.

All headphone cue boxes, microphone stands, power cables, and microphone cables should be put inside the circle of microphones. There should be as few cables crossing the area where the singers stand as possible. Just behind the area where the singers will be standing at the microphones, there should be one chair available for each singer to use. This will give the singers the opportunity to be seated in between songs or breaks so that they don't have to remain standing the entire time.

In summary, the type of condenser microphones that are set up for each singer will vary according to their voice-type and the frequency response characteristics of certain microphones. The advantage to using individual microphones for each singer is to be able to perfectly match the voice-type to the characteristics of a particular microphone. The concepts of voice-type considerations and vocal chain combinations will be discussed further in the next few chapters.

CHAPTER 5

VOICE-TYPE CONSIDERATIONS

When recording a vocal track, it is necessary to first consider the type of voice(s). The most basic voice-types are soprano, alto, tenor and baritone-bass. Each of these classifications exhibit vast differences in sonic quality which must be considered when choosing the best microphone for each voice. As a baseline, the fundamental frequency ranges for each of these voice-types are as follows: bass (87-392 Hz), tenor (131-494), alto (175-698) and soprano (247-1175). Male voice harmonics range from 1-12 kHz, while female voices range from 2-12 kHz.[26] Different voice types, however, can be singing the same note and have a very different sound because of something called formants[27] that are present in the voices. A Bb (4) sung on an "a" vowel by a tenor and soprano contain the same fundamental frequency (F_o), but the formant frequencies of the soprano are higher. The soprano shows the highest amplitude for the fundamental, but the tenor shows the highest amplitude for the first formant. The singer's formant for the tenor is also quite high.[28] This begins to explain why a soprano sounds so different from a tenor when singing the same note. The singer's formant[29] for male singers and altos is the clustering of the F_3, F_4 and F_5 formants. These formants usually sit in the range of 2500 Hz to 3500 Hz. Sopranos also have another energy concentration around 9000-

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26. Bruce Bartlett and Jenny Barlett, *Practical Recording Techniques*, 5th ed. (Burlington, MA: Focal Press, 2002), 102.

27. Formants are defined as the spectral peaks of the sound spectrum of the voice or an acoustic resonance of the human vocal tract. Formants are the distinguishing or meaningful frequency components of human speech and of singing. The formant with the lowest frequency is called f_1, the second f_2, and the third f_3. Most often the two first formants, f_1 and f_2, are enough to disambiguate the vowel.

28. Julia Davids and Stephen LaTour, *Vocal Technique: a Guide for Conductors, Teachers, and Singers* (Long Grove, IL: Waveland Press, Inc., 2012), 95.

29. The Singer's Formant is a property of the singing voice to be heard over a very loud orchestra because of a "bump" in the average frequency spectrum of the voice at about 3000Hz.

10,000 Hz.[30] Sundberg argues that sopranos depend more on the amplitude of the fundamental frequency than that of the singer's formant. At higher pitches, the harmonic frequencies are more widely spaced, making it hard for the singer's formant to match the harmonics generated by the vocal folds. Sopranos are more likely to benefit from a strong fundamental or first formant, as listener's ears are highly sensitive to high soprano pitches. However, sopranos singing in their lower register and in mixed voice typically will exhibit and benefit from a singer's formant.[31] According to Lee S-H et al, for sounds produced in the head register every voice type except altos produced a significant energy concentration between 2.2-3.4 kHz and also between 7.5-8.4 kHz.[32]

With this brief review of some of the differences between voice-types, the need for choosing the correct microphone to enhance frequencies that will compliment the frequency analysis of a particular voice is quite apparent. The pairings between voice and microphone may not always be a perfect match; however, with some consideration, choosing a microphone that does not accentuate the frequencies that are already naturally increased in a particular voice-type could be imperative to the success of the vocal recording. Mendoza, Valencia, Munoz and Trujillo conducted a study to determine if there were acoustical differences between male and female voices. They used extended speech samples in 31 women and 24 men. Recorded readings of a text were reviewed by means of Long-term Average Spectrum analysis. They extracted the amplitude values (in decibels) at intervals of 160 Hz spanning a range of 8 kHz. The results showed a

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30. Sang-Hyuk Lee, Hee-Jun Kwon, Hyun-Jin Choi, Nam-Hun Lee, Sung-Jin Lee and Sung-Min Jin, "The Singer's Formant and Speaker's Ring Resonance: A Long-Term Average Spectrum Analysis," *Clinical and Experimental Otorhinolaryngology* 1, no. 2 (June 2008): 92.

31. J. Sundberg, "The perception of singing," in *The Psychology of Music*, ed. D. Deutsch (San Diego, CA: Academic Press, 1999), 173.

32. Sang-Hyuk Lee, Hee-Jun Kwon, Hyun-Jin Choi, Nam-Hun Lee, Sung-Jin Lee and Sung-Min Jin. "The Singer's Formant and Speaker's Ring Resonance: A Long-Term Average Spectrum Analysis," *Clinical and Experimental Otorhinolaryngology* 1, no. 2 (June 2008): 93.

significant difference between genders with a special interaction between gender and frequency level. The female voice showed greater levels of aspiration noise, located mostly in the spectral region of the third formant. This boost causes the female voice to have a more "breathy" quality than the male voice.[33] Also, the center frequency of the singer's formant for basses, baritones and tenors are about 2.4, 2.6 and 2.8kHz respectively.[34]

In laymen's terms, a soprano voice would not need a microphone that boosts the upper frequencies between 7.5-8.4 kHz. Those frequencies for a soprano are already enhanced naturally. On the other hand, an alto with a rich, dark sound might need an extra boost in the 2.2-3.4 kHz range. Since most tenors already have a naturally boost in the 2.5-3.5 kHz range, they might need a microphone that boosts the upper frequencies. Microphones best suiting the baritone-bass voice would tend to be ones that boost the upper frequencies since their voices are already heavily weighted with lower frequencies. However, the choices of microphone could vary as much as voices vary. More discussion on microphone choices and subsequent combinations of the vocal chain will be discussed in Chapter Six.

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33. E. Mendoza, N. Valencia, J. Munoz, and H. Trujillo, "Differences in voice quality between men and women: use of the long-term average spectrum (LTAS)" *Journal of Voice* 10, no. 1 (March 1996): 59-66.

34. Thomas D. Rossing, *Springer Handbook of Acoustics* (New York: Springer Science+Business Media, LLC, 2007), 102.

CHAPTER 6

RECORDING EQUIPMENT CONSIDERATIONS

Microphones

There are three types of microphones: dynamic, ribbon and condenser (sometimes called capacitor). A dynamic microphone capsule, or transducer, has a coil or wire attached to a diaphragm and suspended in a magnetic field. When sound waves vibrate the diaphragm, the coil vibrates in the magnetic field and generates an electrical signal similar to the incoming sound wave. Another name for the dynamic microphone is moving coil microphone. In a ribbon microphone capsule, a thin metal foil or ribbon is suspended in a magnetic field. Sound waves vibrate the ribbon in the field and generate an electrical signal. A condenser or capacitor microphone capsule has a conductive diaphragm and a metal backplate placed very close together. They are charged with static electricity to form two plates of a capacitor. When sound waves strike the diaphragm, it vibrates. This varies the spacing between the plates. In turn, it varies the capacitance and generates a signal similar to the incoming sound wave. Because of its lower diaphragm mass and higher damping, a condenser microphone responds faster than a dynamic microphone to rapidly changing sound waves.[35]

The condenser microphone is the microphone most commonly used for recording vocals. The ribbon microphone is used occasionally, but only in situations where the vocal being recorded is extremely bright or harsh, mostly because of the dark nature of

35. Bruce Bartlett and Jenny Barlett, *Practical Recording Techniques*, 5th ed. (Burlington, MA: Focal Press, 2002), 81.

the ribbon microphones. A dynamic vocal microphone such as the Shure SM 58 is designed for and usually only used for live applications.

There are two types of condenser microphones: true condenser and electret condenser. In a true condenser microphone (externally biased microphone), the diaphragm and backplate are charged with voltage from a circuit built into the microphone. In an electret condenser microphone, the diaphragm and backplate are charged by an electret material, which is in the diaphragm on the backplate. Both microphones can sound equally as good, but most engineers prefer the more expensive true condensers.[36]

Some of the condenser microphones that have been used to record legendary recording artists throughout history are as follows: Neumann U-47, Telefunken 251, Neumann U-47 FET, Neumann U-67, Sony C-800G, AKG C-12 (original), AKG C414 EB and Neumann M 49. These are considered by many professionals to be the best vintage vocal microphones, but require a sizeable financial investment. There are other condenser microphones that have garnered some attention in the last few years, such as the Rode NT1-A, Rode Classic II, Neumann U-87, Brauner VMA, Neumann M 149, Audio Technica 5040, Neumann M 147 Tube, Rode NTK Tube, Cathedral Pipes Notre Dame, Cathedral Pipes Regensburg Dom, Cathedral Pipes St. Mary, Manley Reference Gold, Manley Reference Cardioid, Sterling Audio ST6050, Sterling Audio ST77 FET, Miktek CV4, Miktek C7, Audio-Technica AT4050, Rode NT1000, and Blue Bottle. These condenser microphones range from the very inexpensive to the expensive, but all are considered by professionals to be very good for recording the voice. The CAD Audio

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36. Bruce Bartlett and Jenny Barlett, *Practical Recording Techniques*, 5th ed. (Burlington, MA: Focal Press, 2002), 81.

Trion 7000 is a ribbon microphone also worth considering. For the purposes of this paper, only the characteristics of the frequency response graphs and the sonic quality of the recorded vocals that these particular microphones capture will be discussed and considered. It is the author's intent that the usage and documented reviews of these vocal microphones will be a testament to each microphone's performance quality for recording vocals in the modern digital studio.

The Neumann U-47, a multi-pattern tube condenser microphone built in 1947, is known as the most revered vocal microphone in history.[37] It responds best on close-up male vocals and is known for smoothing out harshness and adding some warmth to the vocal sound. Frank Sinatra recorded almost everything he did on this microphone during his great recordings of the 1950-60's. Some other artists that have been known to use this microphone include Bryan Adams, John Mayer, Lisa Loeb, Macy Gray, Patti LaBelle, Michael Buble and James Blount.[38] The frequency bump on this microphone in the cardioid pattern happens at about 2kHz for an amount of +5dB.[39] The U-47 has a certain 'tubby' sound with excellent low-end response and punchy mid frequencies. Also, the VF14 tube, which is an integral part of the U-47 sound, also tends to add an authority to the midrange that you don't experience with any other tube microphone.[40] See figure 6.1.

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37. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Neumann/U-47 [accessed November 27, 2013].

38. Gary Ladinsky, "Tips for Studio Recordings," Design FX Audio, http://www.dfxaudio.com/quick-tips-page.html [accessed November 27, 2013].

39. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Neumann/U-47 [accessed November 27, 2013].

40. J.J. Blair, "The Great American Mic Hunt," *Electronic Musician* (September 2005), http://www.emusician.com/news/0766/the-great-american-mic-hunt/134977#sthash.67D870b8.dpuf [accessed November 27, 2013].

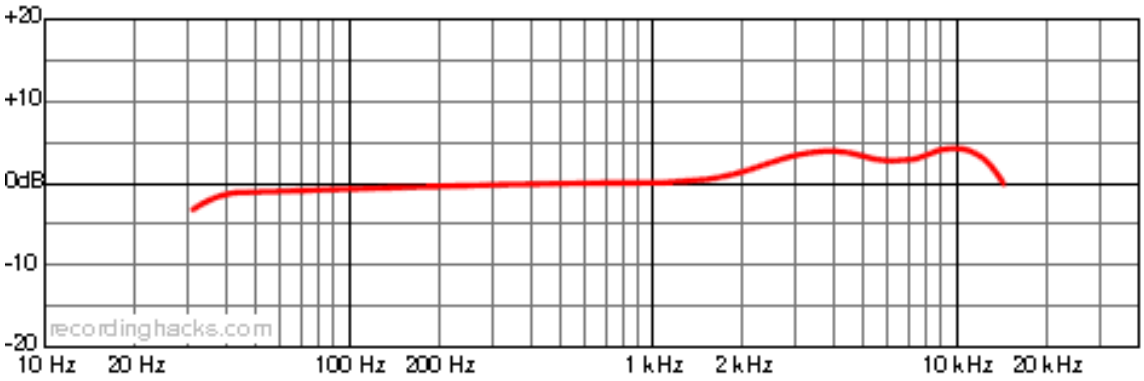

Figure 6.1-Neumann U-47 Frequency Response Graph.

Telefunken ELA M 251 E Microphone has been used by many artists including Peter Gabriel, Van Halen, John Mayer, Anita Baker, Green Day, Norah Jones, Christina Aguilera, Jessica Simpson, Fiona Apple, Michael Jackson, Kanye West, Dwight Yoakam, Martina McBride, Kelly Clarkson, Stevie Wonder, Celine Dion, Elton John and Usher.[41] The microphone has long been known in the industry as having a rather flat frequency response. In other words, a good choice for almost any voice type, but it is sometimes too bright when used on thin voices. Officially though, this microphone has a slow rise frequency bump at 3kHz to about +2.5 dB, then has a more dramatic surge at 8kHz to about +6dB.[42] See figure 6.2.

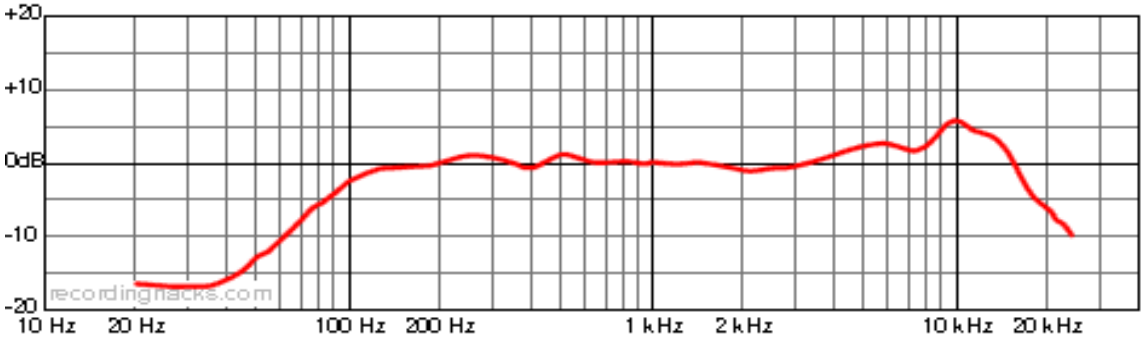

Figure 6.2-Telefunken ELA M 251 E.

41. Gary Ladinsky, "Tips for Studio Recordings," Design FX Audio, http://www.dfxaudio.com/quick-tips-page.html [accessed November 27, 2013].

42. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].

The Neumann U-47 FET Microphone was intended to be a phantom-powered, FET version of the U47 Tube Microphone, designed a few years after the VF14 tube at the heart of the U47 became unavailable. The sound of the U-47 FET is a little more closed in and maybe punchier than the U-47 Tube. The U-47 FET was manufactured from 1969 through 1986. Robin Zander from Cheap Trick uses this microphone to record vocals.[43] See figured 6.3.

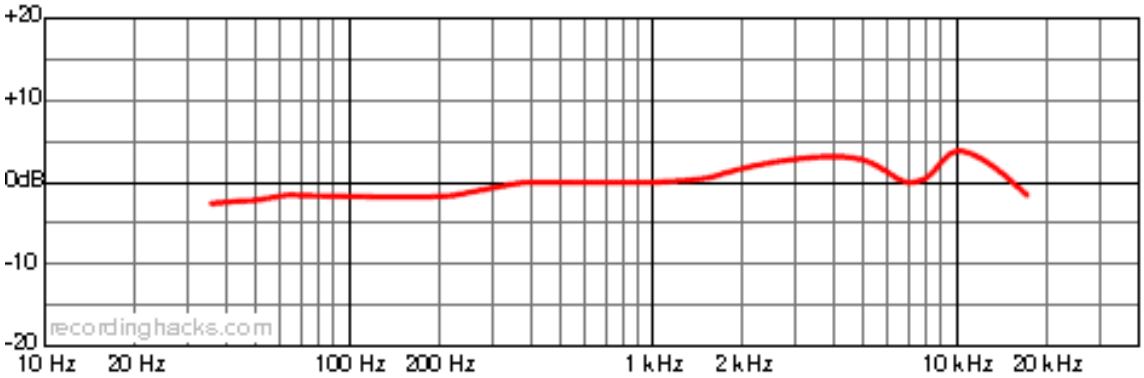

Figure 6.3-Neumann U-47 FET.

The Neumann U-67 Microphone has traditionally been a good choice for male vocals. It has a little bit of a high end bump to the frequency range, which works great for men, given them that extra added upper frequencies that they lack in comparison to women's voices. Some of the artists who have used this microphone to record vocals include John Mayer, Josh Groban, Bon Jovi and Liz Phair.[44] It is a microphone with attitude and openness that works great for pop and rock vocals. It's an especially good microphone to use on young singers.[45] See figure 6.4.

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43. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Neumann/U47fet [accessed November 27, 2013].

44. Gary Ladinsky, "Tips for Studio Recordings," Design FX Audio, http://www.dfxaudio.com/quick-tips-page.html [accessed November 27, 2013].

45. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].

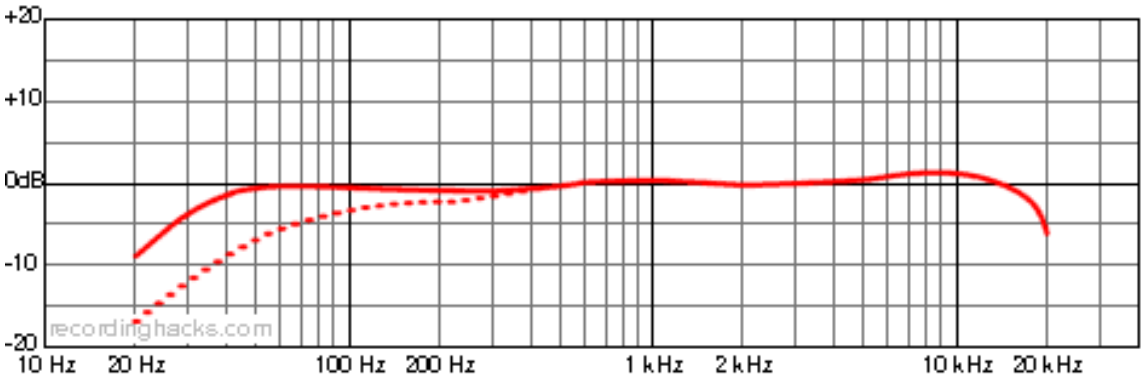

Figure 6.4-Neumann U-67.

The Sony C-800G Microphone has been a favorite of country, r&b and pop artists for many years. It has been known in the industry for having a little bump in the lower and very highest frequencies. So, for a female it adds some body or warmth in the lower frequencies, but also adds back the gloss that the female voice needs for a very present sound in the high frequencies. This might not be as desirable a sound for other genres such as jazz. This microphone has been more prominently used for pop vocals. Dr Dre, John Legend, Mariah Carey, Mary J. Blige, Eve, J. Lo, Pussycat Dolls, R. Kelly, Carrie Underwood & Kelly Clarkson are some of the artists that have been known to use this microphone for recording vocals.[46] See figure 6.5. The 0 degrees line means the source is directly in front of the microphone. 90 degrees is sound recorded from the side and 180 degrees is sound recorded from the backside of the microphone. All of these measures are taken with the microphone in the cardioid pattern.

46. Gary Ladinsky, "Tips for Studio Recordings," Design FX Audio, http://www.dfxaudio.com/quick-tips-page.html [accessed November 27, 2013].

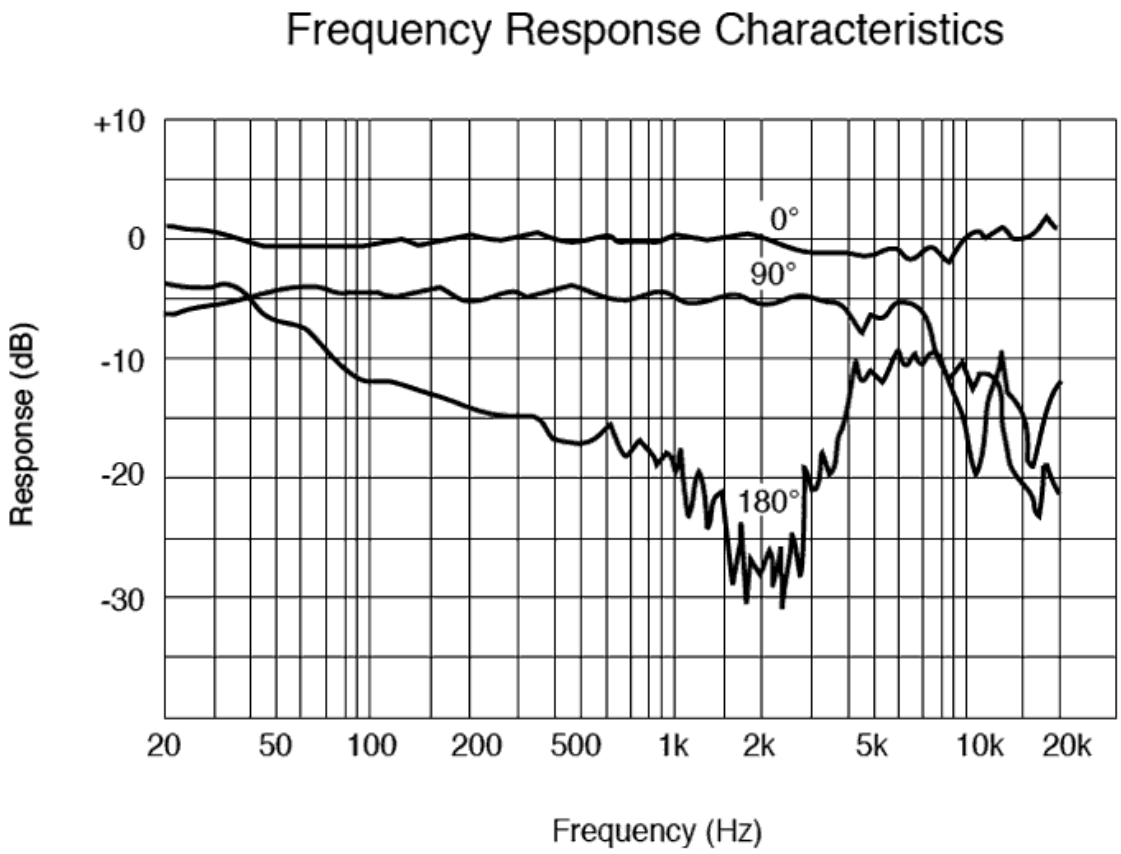

Figure 6.5-Sony C-800G.

The original AKG Acoustics C12 Multi-Pattern Tube Condenser Microphone was a silver model built in 1953. There have been several replicas made through the years after it was discontinued in 1963. The current model is manufactured in green with the name C-12 VR. The VR stands for Vintage Replica.[47] This microphone has been used by Stevie Wonder, Josh Groban, Andrea Bocelli, Whitney Houston, Celine Dion and Kelly Clarkson.[48] The frequency response pattern for the C12 is extremely flat with a dramatic high-end sloping bump that starts at about 6kHz. See figure 6.6.

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47. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/AKG-Acoustics/C-12 [accessed November 27, 2013].

48. Gary Ladinsky, "Tips for Studio Recordings," Design FX Audio, http://www.dfxaudio.com/quick-tips-page.html [accessed November 27, 2013].

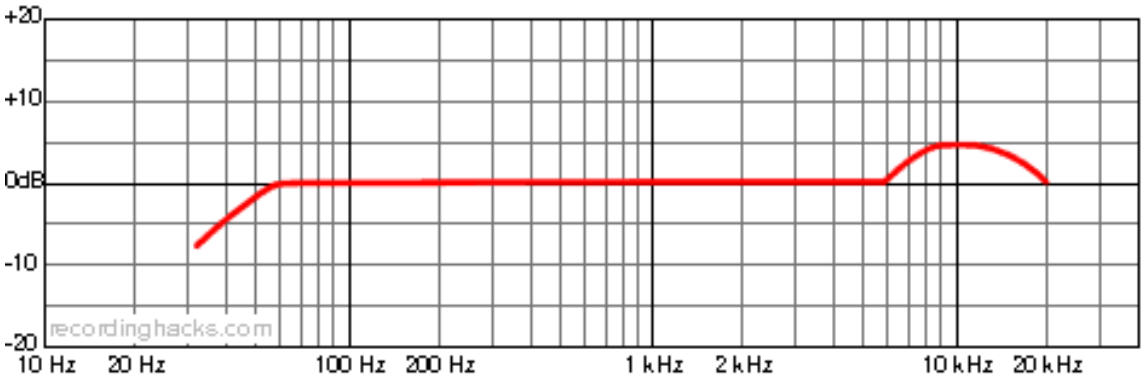

Figure 6.6-AKG C12 (original).

The AKG C414 EB Microphone was produced in 1976 and was the second generation to the original model called C414 comb. This microphone has historically had a smooth mid-range, not so much bass response, but was really known for its extended top-end that has been described as shimmering, airy or subtly bright.[49] This microphone could also be especially useful in a group vocal setting. See figure 6.7.

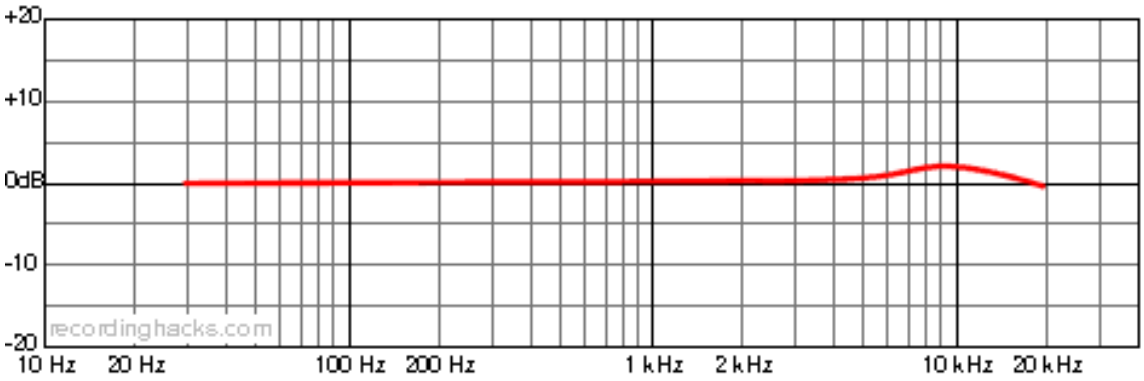

Figure 6.7-AKG 414 EB.

The Neumann M 49 Multi-Pattern Tube Condenser Microphone was released in 1951 and discontinued in 1974.[50] The M49 is one of the most classic microphones that the Neumann company ever manufactured. It is incredibly smooth with a lovely top end

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49. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/AKG-Acoustics/C-414-EB [accessed November 27, 2013].

50. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Neumann/M-49 [accessed November 27, 2013].

and lots of weight in the lows. The only down-side to this microphone is that it has some self noise and tends to break up just a little when pushed hard. This microphone was once used by B.B. King, Tony Bennett, Rod Stewart, Barry Gibb, Miles Davis, Barbra Streisand, Etta James, LeAnn Rymes, Duke Ellington, Simon and Garfunkel.[51] See figure 6.8.

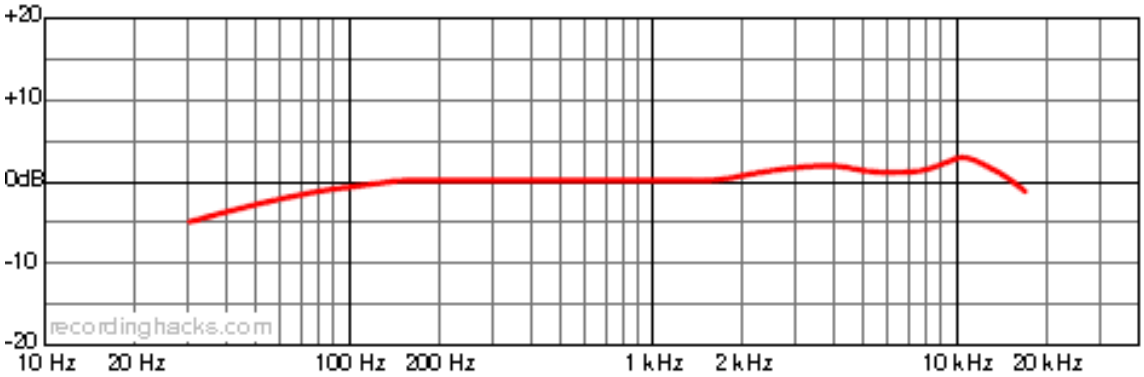

Figure 6.8-Neumann M 49.

The Rode NT1-A Cardioid Condenser Microphone is one of the more recent additions to microphone possibilities and is reasonably inexpensive. It has acquired the reputation of being best used on sopranos and one of the quietest condensers ever built. This microphone has a pretty smooth frequency response all the way from 20 Hz to 20kHz with only a small bump between 100-200Hz and another small presence bump around 12kHz. This would be a very good microphone option to also use for group vocals.[52] See figure 6.9.

51. Gary Ladinsky, "Tips for Studio Recordings," Design FX Audio, http://www.dfxaudio.com/quick-tips-page.html [accessed November 27, 2013].

52. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Rode/NT1-A [accessed November 27, 2013].

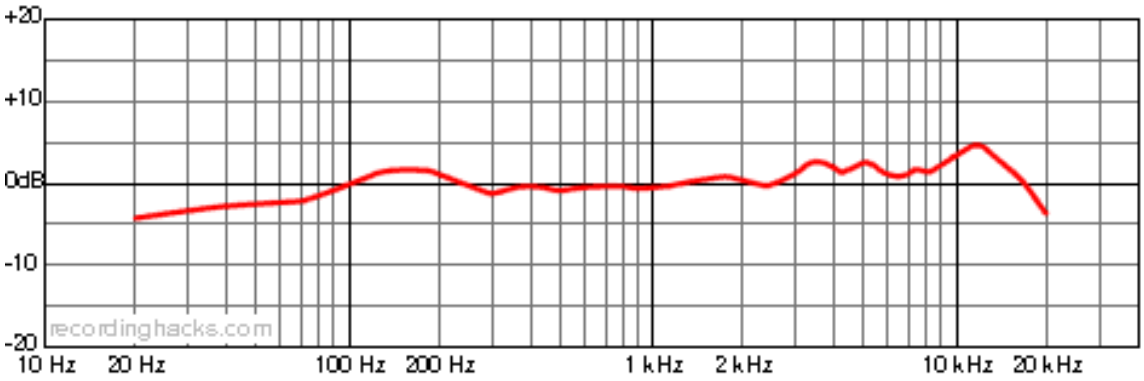

Figure 6.9-Rode NT1-A.

The Rode Classic II Microphone is the big brother to the Rode NT1-A. It has become many industry professionals' favorite new mid-priced condenser microphones. The frequency response is generally flat from 20Hz to 20kHz (within ±3dB) with a smooth rise presence lift from 2kHz to around 12kHz. The presence rise, because of the high center frequency of the boost and the gentle nature of the curve, has an open, airy quality rather than a harsh presence.[53] Instead, it seems to deliver a warmth and richness to the vocal. The following artists have used the Rode Classic II to record vocals: Unkle, Matt Sorum, Rocket Science, Duff McKagan, Blue Man Group and You Am I.[54] See figure 6.10.

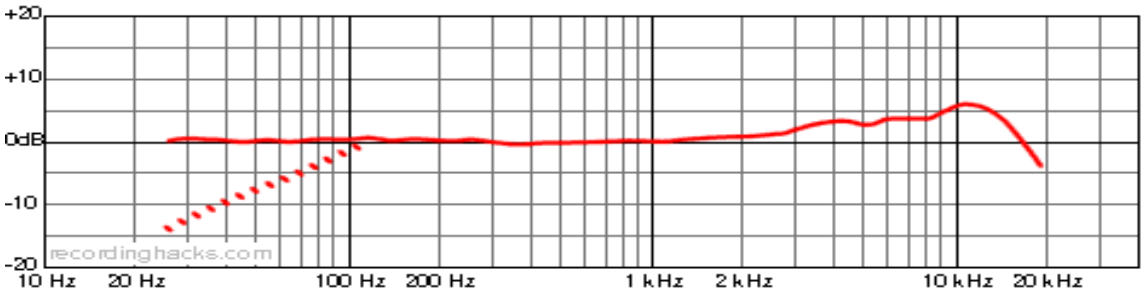

Figure 6.10-Rode Classic II.

53. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Rode/Classic-II-Limited [accessed November 27, 2013].

54. Rode Microphones, "Classic II Vintage-Voiced Valve Condenser Microphone," http://www.rodemic.com/mics/classic2 [accessed November 27, 2013].

The Neumann U-87 Multi-Pattern Condenser Microphone was designed as a solid-state version of the U 67 tube microphone. It retained the K67-style capsule of the U67, but replaced the tube amplifier circuit with a FET/transformer design. This microphone was released in 1967. With a bit of a high frequency boost, this is not one of the most favorite Neumann microphones, but might be a nice to use for group vocals.[55] See figure 6.11.

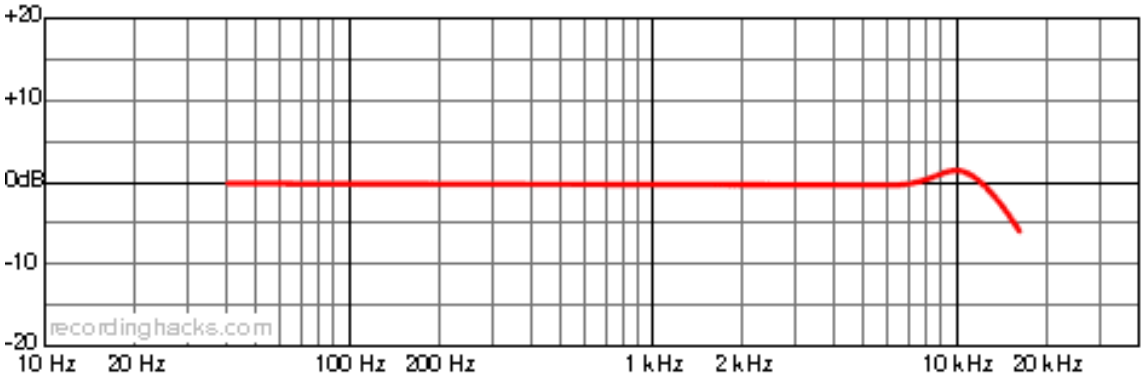

Figure 6.11-Neumann U-87.

The Brauner VMA Microphone is a hand built large-diaphragm tube condenser combining the VM1 and VMX circuitry in one microphone allowing either the 'natural' or 'charming' Brauner tonalities. A switch on the external power supply selects which circuitry is active, either VM1 or VMX. The microphone contains two complete circuits rather than filters to alter the tone. It is basically like have two microphones in one. This microphone was used by Queens of the Stone Age and Celine Dion for vocals. The frequency range is somewhat larger than other microphones spanning from 18 Hz to 24

55. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Neumann/U-87 [accessed November 27, 2013].

kHz.[56] (The Brauner company does not publish frequency graphs of their VM series microphones.)

Neumann M 149 Tube is a high-end tube microphone that is derived from the Neumann U47 and M 49. It is known for having a "finished mix" sound on singers with a bold midrange character, an extended high frequency range and great amount of output. This would probably not be a first choice for lead vocal microphones, but might be nice in a group vocal situation.[57] See figure 6.12.

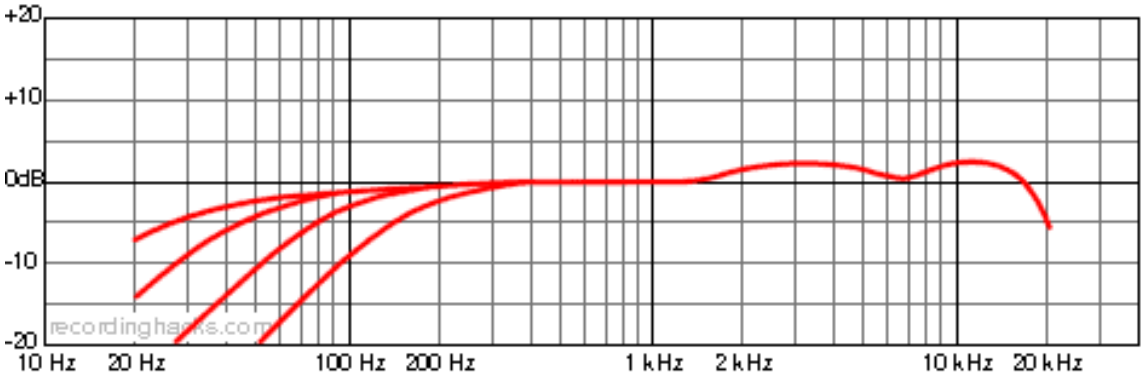

Figure 6.12-Neumann M 149.

The AT5040 is a brand new microphone from Audio-Technica, one that is supposedly tailored for top-flight vocal recording. It is the flagship microphone in a new 50 Series line and was introduced at the 2012 Audio Engineering Society show in San Francisco.[58] This microphone is said to contain new cutting-edge technology and design principles. It does away with the common circular diaphragm in favor of a rectangular one. European manufacturers, like Pearl Microphone Laboratory, have been selling microphones with rectangular diaphragms, but the AT5040 takes the idea a step further

56. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Brauner/VMA [accessed November 27, 2013].

57. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Neumann/M-149 [accessed November 28, 2013].

58. Paul Vnuk, Jr., "Audio-Technica AT5040 Studio Vocal Microphone," *Recording* (May 2013), http://www.recordingmag.com/productreviews/2013/05/76.html [accessed November 28, 2013].

by combining four rectangular diaphragms into one large "super-diaphragm". Looking at its frequency graph, three areas jump out: a low-end bump between 20 and 80 Hz, a strong mid rise from 1.5 to 4 kHz, and a smaller high-end plateau from 8 to just above 10 kHz. All of these levels, particularly the pronounced mid peak, fits perfectly in line with this microphones intended use as a vocal microphone. While it does have a bump in the 10 kHz area, it is nowhere near as bright as many modern condensers that sit high in the 8 to 15 kHz range. It has been described as an extremely smooth top end microphone with controlled sibilance. A dip of approximately 3dB occurs between 4.5kHz and 8.5kHz, a prime sibilance location. The microphone's reactivity slowly tapers after the 10kHz mark. An approximate 2 to 3dB boost in response below 80Hz could create a proximity effect on a male vocalist, but could be rolled off in the mix.[59] See figure 6.13.

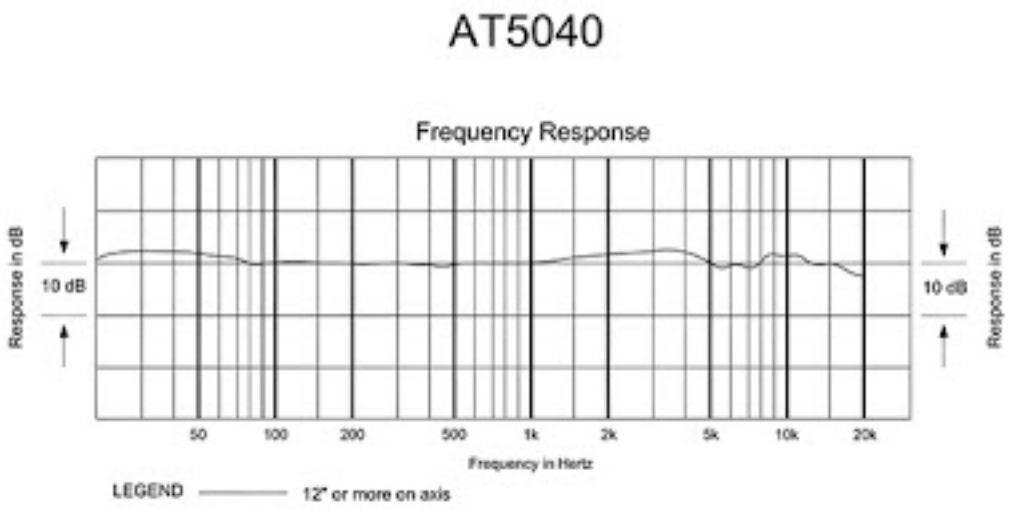

Figure 6.13-AT 5040.

Neumann M 147 Cardioid Tube Condenser Microphone is a fixed-cardioid valve microphone which uses the same large dual-diaphragm capsule design, the

59. Paul Vnuk, Jr., "Audio-Technica AT5040 Studio Vocal Microphone," *Recording* (May 2013), http://www.recordingmag.com/productreviews/2013/05/76.html [accessed November 28, 2013].

classic M7, as is found in the legendary U47 and M49 microphones. It is great for recording both male and female voices. With the microphone distanced at around 18 inches it produces a clean, articulate sound. Working close proximity, it gives a richer, warmer and fuller sound. The frequency response graph shows a gentle LF droop from +2dB at 4kHz to -2dB at 50Hz. There is also a small dip at 7kHz before a 2dB peak at 10kHz. The top end drops off quite quickly above 15kHz, which is typical for large-diaphragm capsules. The response suits voices well, with the right degree of presence for clarity of diction, but without emphasizing sibilance.[60] See figure 6.14.

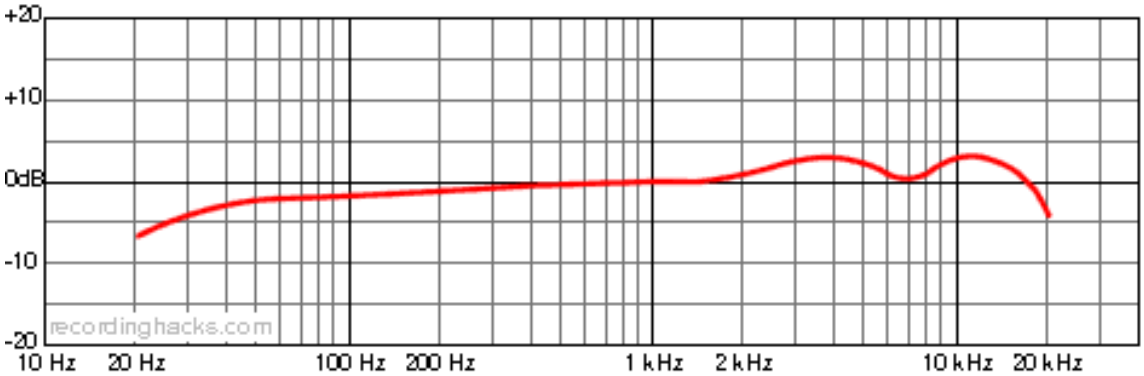

Figure 6.14-Neumann M 147.

Rode NTK Cardioid Tube Condenser Microphone is described as smooth with an expansive airiness. The NTK is Australian designed and manufactured with an ultra-wide dynamic range, very low noise, and stunning tube warmth. The NTK's rich valve sound is ideal for vocals. The NTK is a quiet microphone, but seems to blossom when hit with more level.[61] See figure 6.15.

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60. Ian Gilby, "Neumann M147 Tube Studio Condenser Microphone," *Sound On Sound* (December 2000), http://www.soundonsound.com/sos/dec00/articles/neumann.htm [accessed November 28, 2013].

61. Rode Microphones, "NTK Valve 1 Condenser Microphone," http://www.rodemic.com/mics/ntk [accessed November 28, 2013].

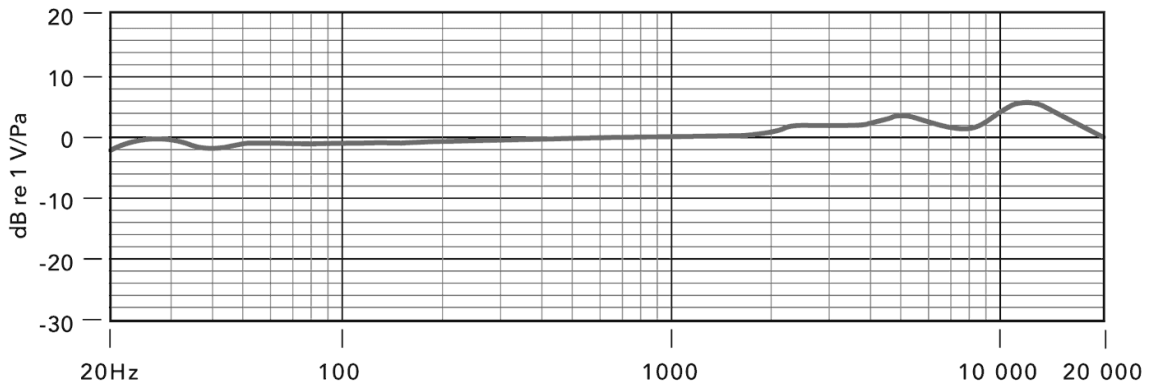

Figure 6.15-Rode NTK Tube.

The Cathedral Pipes Notre Dame Tube Condenser microphone is a new microphone on the scene only having been released in 2012. With a handmade M7 capsule and exact clone of the U47 body, it is described as sounding close to a well-maintained U47 Tube or as a modern U47 with "bite."[62] No frequency response graph exists for this microphone.

The Cathedral Pipes Regensburg Dom multi-pattern tube condenser microphone is known as a "vintage German design with bite." It is modeled after one of the recording industry's most treasured microphones, the Neumann U47. The Regensburg microphone has been said to deliver an "in your face" attitude, while retaining a warm vintage tone suited for a variety of applications. It has quickly become known as a "go-to" vocal microphone.[63] No frequency response graph exists for this microphone.

The Manley Reference-Gold Multi-Pattern Tube Condenser Microphone was Manley Labs flagship tube condenser microphone. This microphone contains airiness and a superb ultra-high frequency response. Michael McDonald, Jack Johnson and G-Love have used this microphone to record vocals. The Reference Gold gives a boost

62. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].

63. Charles Dickinson, "The Regensburg Dom Vintage Tube Microphone," Cathedral Pipes, http://www.cathedralpipes.com/regensburg.php [accessed November 29, 2013].

between 12-18kHz, giving a clean, intimate sound.[64] No frequency response graph exists for this microphone.

Released in 1991, the Manley Reference Cardioid Tube Condenser Microphone has a frequency response that is a little broader than most from 10Hz-30kHz. Etta James (2005) and Avril Lavigne are two artists that were known to record vocals on this microphone. Voice-over legend Don LaFontaine was not ashamed to call this his "microphone of choice." The Reference microphone is similar to the reference gold in that it has a high frequency edgy resonance, a similar proximity effect and pretty good immunity from pops and sibilance problems. The Reference Cardioid sounds close to many of the vintage European tube microphones such as the U47 when they were first built. Its tonal balance and character is admired, especially for vocals. In situations where an engineer might lean on a compressor to boost 5 or 10K and score a bit more punch, the Reference Cardioid could be the alternative. This microphone can display the subtleties of a jazz vocalist and also catch the rough edgy of rock and blues. The lows in this microphone are tight, well-defined, and big without being muddy. It keeps a clarity of the image throughout the frequency spectrum.[65] A frequency graph of this microphone was unavailable.

The Sterling Audio ST6050 "Allen Sides" is the first microphone in Sterling Audio's Signature Series. It is a premium FET condenser voiced by Allen Sides of Ocean Way Studios, designed primarily for studio vocal applications. The ST 6050 is a spectacular vocal microphone comparable to the U47 FET. Sterling tried to come out

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64. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 29, 2013].

65. EveAnna Manley, "Manley Reference Cardioid Microphone," Manley Laboratories, Inc, http://www.manley.com/refcard.php [accessed November 29, 2013].

with something comparable to a U47 and missed the mark by a bit, but came out with something that became a classic of its own. The frequency response is relatively flat but has a gentle boost of +6dB starting at 2.5 kHz up till 10 kHz.[66] See figure 6.16.

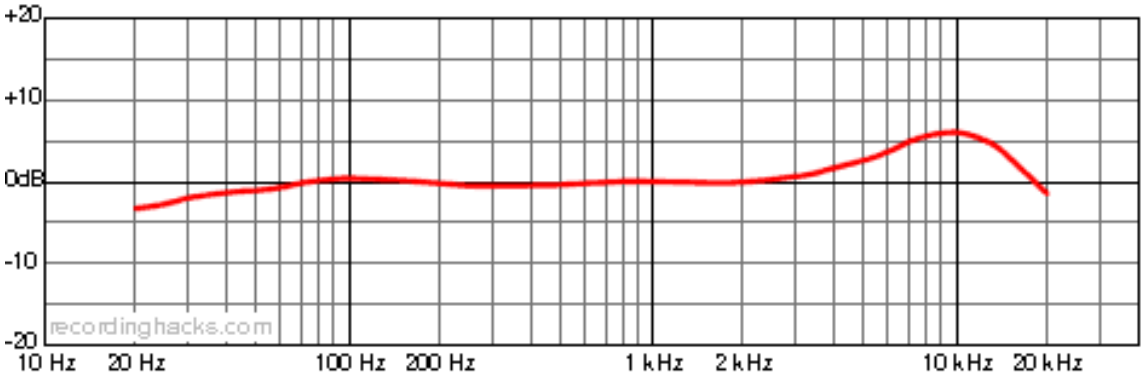

Figure 6.16-Sterling Audio ST6050.

The Miktek CV4 Multi-Pattern Tube Condenser Microphone is Miktek's flagship condenser. It is a large diaphragm microphone with a little more warmth and color than a U47 or the Cathedral Pipes. The CV4 is very well balanced, dynamic and doesn't enhance problems.[67] The top frequencies are crisp without being brittle, the mid frequencies are strong, present and the bottom is huge, but not loose. On a voice, it sounds neutral and accurate with a bit of magic. In addition, there was certainly no tradeoff in smoothness or detail.[68] See figure 6.17.

66. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Sterling [accessed November 29, 2013].

67. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].

68. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Miktek/CV4 [accessed November 29, 2013].

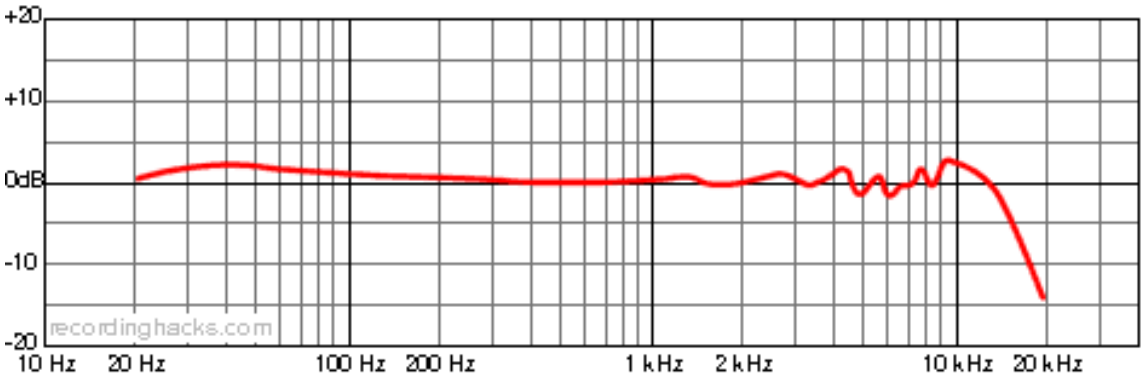

Figure 6.17-Miktek CV4.

The Miktek C7 Multi-Pattern Condenser Microphone is a three-pattern FET condenser with a large-diaphragm capsule. Sometimes it is just the ticket for vocals, slightly bright, but very useable. It has a nice overall balance and dynamics.[69] For female vocals, the C7 provides a sound that fits inside a mix with very little need for EQ. Many newly designed vocal mics have friendly response curves and specs, but no real vibe, or flattery to the voice. However, the Miktek C7 brings some real character to the table and enhances the source, without clouding or coloring the sound in an unexpected way.[70] See figure 6.18.

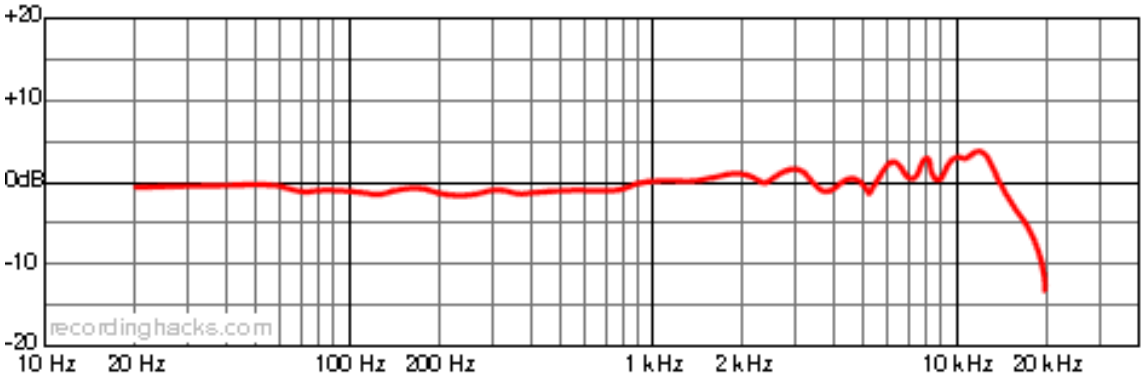

Figure 6.18-Miktek C7.

69. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].

70. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Miktek/C7 [accessed November 29, 2013].

The Audio-Technica AT4050 is a vocal underdog hero (along with the AT4033). Maybe it is not the bold character of the Neumanns, but is always useable and versatile.[71] When the AT4050 is set to cardioid mode, it produces a very warm, silky sound, but with less top-end shimmer than the AT4033. Though a touch sibilant at times, Audio-Technica's condenser is well balanced tonally, exhibiting nice presence and loads of detail.[72] See figure 6.19.

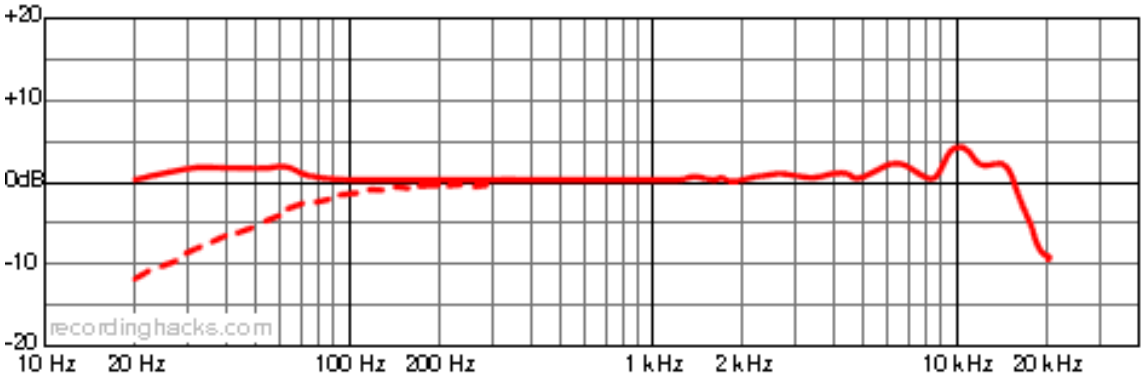

Figure 6.19-Audio-Technica AT4050.

The Rode NT1000 Cardioid Condenser Microphone is a fixed-cardioid, large diaphragm FET condenser, sometimes the perfect choice for pop vocals. It almost doesn't need EQ in a mix.[73] The NT1000 sounds like a hybrid between the NTK and the original NT1. This microphone alludes to the smoothness of the NTK and is noticeably smoother sounding than the NT1, but it doesn't have the same warm intimacy of the more expensive NTK. This is to be expected given that the NTK is a tube microphone, but exploiting the proximity effect by working closer to the microphone helps warm the sound up a little more, as does using a sympathetic compressor. The NT1000 seems to be

71. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].
72. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Audio-Technica/AT4050 [accessed November 29, 2013].
73. Matt McGlynn, "Oceanway Vocal Microphone Test," RecordingHacks, http://recordinghacks.com/2012/07/15/oceanway-vocal-microphone-test/ [accessed November 27, 2013].

based upon the U87. When compared to a U87, there is a similar basic sound spectrum, but the NT1000 is a bit brighter. The NT1000 seems to have a boost of about 1 db at 6k, and a wider boost of about 1.5db centered at 12k. This makes the use of equalization in the mix a less likely incident than with a U87, especially beneficial in the analogue world, and on distant subjects. The mid and low frequencies seem about the same balance, and the quality of the NT1000 is pretty close to the U87.[74] Given the price differential, even when compared to a good used U87, the NT1000 seems to be an excellent value.[75] See figure 6.20.

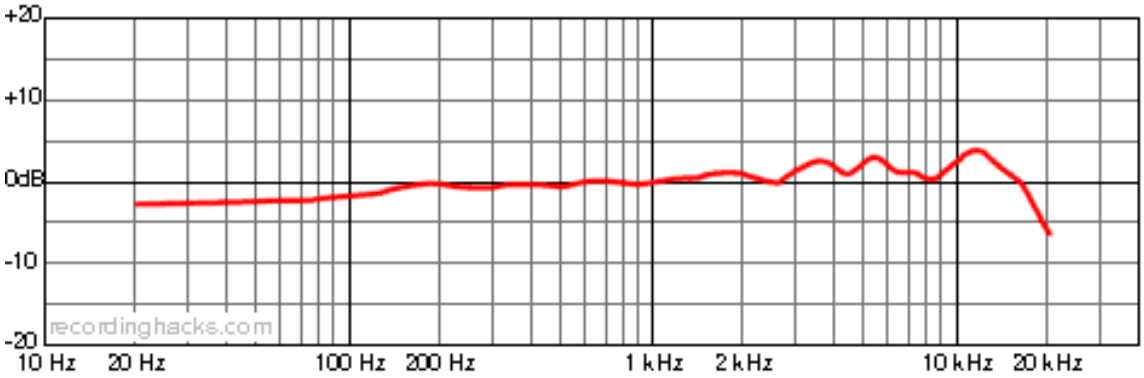

Figure 6.20-Rode NT-1000.

The BLUE Bottle Microphone was developed in 1995 and stands for Baltic Latvian Universal Electronics. It is the brainchild of Skipper Wise, a session musician and resident of southern California and his partner, Martins Saulespurens, a recording engineer from the Baltic State of Latvia. BLUE manufactures a complete line of vacuum tube and solid-state microphones in a wide range of colors. (Although according to Wise the majority are, in fact, blue; with one of five being red, green or yellow.)[76] Circuit-wise, the Bottle microphone features a choice of one of eight capsules, completely hand-built and individually voiced (in

74. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/Rode/NT1000 [accessed November 29, 2013].

75. Ibid.

76. Frederick J. Bashour, "Blue Bottle Microphone," *Pro Audio Review Magazine* (April 1999): 28.

Riga, Latvia), an EF86 pentode (used in triode mode) and a humongous, custom-built output transformer can, which is more than 2" in both diameter and height. Although optimized for 60V, one can step it up to 90V or down to a minimum of 34V. BLUE maintains that this feature is useful for matching the sensitivity requirement of different program material. This feature also changes the sound of the microphone from aggressive (at the higher voltages) to mellow and laid-back (at the lower ones). The power supply also features a soft start circuit that ramps up the tube's heater voltage and B+ separately. First it turns on the heater.[77] After approximately 80 seconds, when the cathode is fully heated, the plate voltage is gradually applied and the microphone output is muted.[78] After about three minutes, when the mute has settled into its correct operating mode, the muting is disabled and audio is output from the supply. BLUE manufactures eight lollypop-shaped capsule heads for the Bottle microphone: the B7 cardioid (single backplate, like a Neumann M7), the B6 cardioid (dual backplate, like an AKG CK 12), the B5 pressure omni, the B4 Perspex sphere pressure omni (like used on the Neumann M50), the B3 midsize cardioid and the B0 bright by itself.[79] Even when the B6 and B7 are compared to a Neumann U47 and a Manley-sized AKG C24 stereo microphone, the Bottle microphone does not sound like any other microphone. It has some of their sound quality, but has a definite character of its own. Its design philosophy was different from, say, the Lawson microphones, which were intended from the outset to sound in between a U47 and an M49. BLUE's literature states that the company studied the frequency curves from scores of vintage, tube and solid-state microphones, from which it determined the most design of the microphones, according to the literature, was created by what BLUE terms The Popular Opinion (TPO). TPO is a consensus of expert engineers and discriminating musicians on the type of sound that is needed in the recording process today; and, with that philosophy in mind BLUE has

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77. Frederick J. Bashour, "Blue Bottle Microphone," *Pro Audio Review Magazine* (April 1999): 28.

78. Ibid.

79. Ibid.

succeeded admirably.[80] The B7 capsule sounds sort of like a 47 on steroids. However, back up a little from the microphone and the sound mellows out and becomes uninteresting. The B6's extra Telefunken ELAM-type brightness would never work on a bright voice. It would be interesting to try a pair of these for classical recording. The extra brightness could give the singer the sheen one just can't get with EQ. Both the B6 and B7 capsules are very quiet. The B7 has higher output, so the resulting noise figure from the microphone seems low. The B6 has an extreme high-frequency hiss that is perhaps part of its design. One of the really nice things about the BLUE microphone capsules is that they were specifically designed for particular circumstances rather than trying to do everything.[81] The B4 is supposedly BLUE's most special of all capsules. See the B6 frequency response graph figure 6.21.

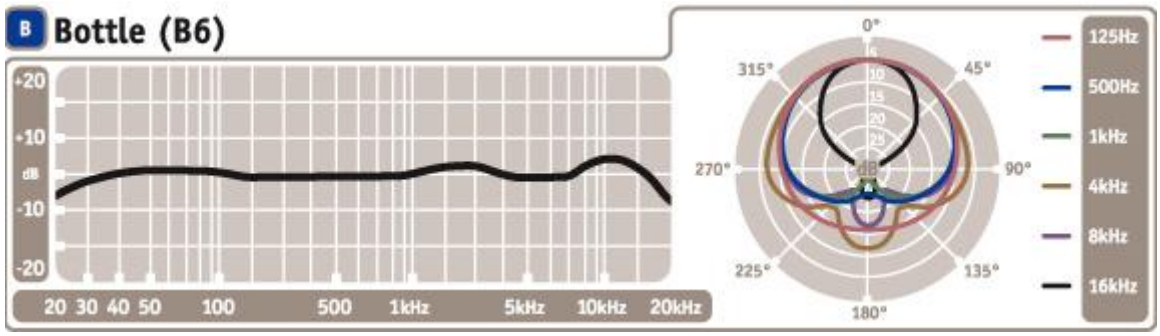

Figure 6.21-Blue Bottle B6.

The CAD Audio Trion 7000 Bi-directional Ribbon Microphone is a dual-ribbon dynamic microphone. The use of two ribbons is to increase sensitivity, however the microphone still has an extreme low output. It is about 16dB less output than condenser microphones. The frequency response is slightly elevated from 40Hz to 100Hz, flat through 4kHz, and rolls off through 15kHz; this contour likely results in a somewhat dark sound, with the subtle high end that is characteristic of most ribbon microphones. It is

80. Frederick J. Bashour, "Blue Bottle Microphone," *Pro Audio Review Magazine* (April 1999): 29.

81. Ibid.

always good to have a good ribbon microphone around for the chance of needing to record a very bright singer.[82] See figure 6.22.

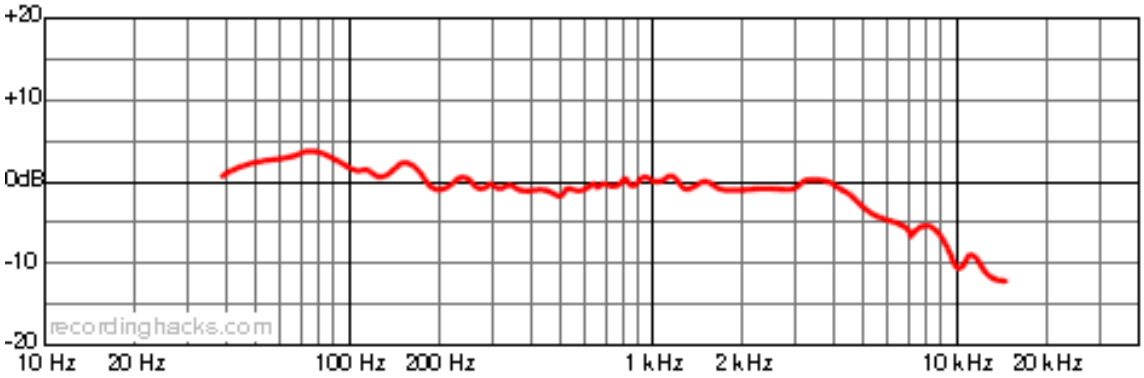

Figure 6.22-CAD Audio Trion 7000.

The Korby Kat 5 System is a microphone body and a set of five capsules that can be interchanged with the microphone still on. This is what is called "hot-swappable." The microphone capsules that are contained in the Kat 5 system are as follows: U47, 251, U67, C-800 & C-12. These capsules are replicas of their namesakes, but are also usually tuned to the vintage microphones with the same name. This writer owns this system and has had very good success with the microphones sounding very close to the vintage microphones as long as they are maintained and re-tuned to a vintage model frequently.

Microphone Pre-amplifiers

The second link in the chain of recording a vocal is the microphone pre-amplifier. Microphones need to be amplified in order for the signal to be at an adequate amplitude or level. There are different characteristics in microphone pre-amplifiers much like the differences in microphones. They all possess different color characteristics. An

82. Matt McGlynn, "The RecordingHacks Microphone Database," RecordingHacks, http://recordinghacks.com/microphones/CAD/Trion-7000 [accessed November 29, 2013].

assortment of recently popular microphone pre-amplifiers will be discussed and compared in the following section.

The API 512c 500v module is a microphone pre-amplifier designed to provide a low noise, unusually good sounding front end for all types of audio systems. The API 512c remains faithful to the circuit designs of API's founder, Saul Walker. Fully featured and hand assembled, the 512c carefully preserves the original sound character that made it so much a part of the early days of recording. This module maintains a very flat, clear response when recording a vocal. Its frequency response is +0, -.3, from 30 to 20 kHz.[83]

The Shadow Hills Mono GAMA is the single channel version of the Golden Age Microphone amplifier adapted for the 500 series rack system. The design of this amplifier provides enormous fidelity and depth across the full frequency range. There are three settings including a transformer switching matrix within the Mono GAMA that provides different colorations to the source. The positions are nickel, discrete and steel. The Nickel Transformer setting has low distortion characteristics and a flat low frequency response. The Nickel upper bandwidth has a ten kHz boost of one dB. The Discrete position is transformerless, exhibiting a very clear and uncolored signal and includes a very fast transient response. The Steel Transformer setting is the most colored and has a one-decibel boost at 40Hz with a tight cue.[84]

The Martech MSS-10 quickly established itself as the microphone preamplifier of choice for many of the audio elite because of its' natural sound. The frequency response is described as a mythical straight wire with gain. It has an uncanny ability to deliver

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83. Larry Droppa, "512c Discreet Mic/Line Pre," Automated Processes, Inc., http://www.apiaudio.com/512c.html [accessed December 23, 2013].

84. Peter Reardon, "Shadow Hills Mono Golden Age Microphone Amp," Shadow Hills Industries, http://shadowhillsindustries.com/pdf/Shadow_Hills_Mono_Gama.pdf [accessed December 23, 2013].

astonishing realism. The real magic of the MSS-10 is that it does absolutely nothing to color the sound. With nothing cheating the audio, it is heard as it is acoustically. Its response has also been described as having a silky top with absolute clarity, knockdown punch, and fat bottom end. The MSS-10 delivers the true essence of the performance with astonishing realism. According to Andy Smith, audio engineer on Paul Simon's album You're The One, Paul's vocal recording chain throughout the album was a Neumann M149 through the Martech microphone pre-amplifier.[85] Bruce Botnick, a Producer/Mixer/Engineer on recording motion picture scores, says the MSS-10 gives him clarity, a noise floor that is well below the vintage microphones, and no unwanted coloration to the sound.[86] Jerry Finn, producer for Blink 182's recordings on albums Take Off Your Pants & Jacket and Enema of the State, described his experience of using the Martech MSS-10 with the following statement. "The first thing that I really noticed was on vocals, which sold me on it immediately. It made the vocals sound really present, which required less EQ and less compression to really get them to stand out on the track."[87]

The Great River MP-500NV is a professional quality microphone preamplifier designed to re-create the vintage sound characteristics of the early 1970s large recording consoles. Modern components give the MP-500NV a little more clarity, punch and performance under normal use. In addition, features like metering before and after the major gain stages let engineers drive it 'hard' to greatly expand the range of sonic options. This allows the engineer/producer to get different colors from the recording of the vocal.

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85. Joe Martinson, "MSS-10 Overview," Martinsound, http://www.martinsound.com/pd_m1so.htm [accessed December 23, 2013].
86. Ibid.
87. Ibid.

All microphones will benefit from the power of the MP-500NV pre-amplifier, including dynamic, condenser and especially ribbon microphones because of their low signal output. This unit is designed to fit into a '500' series rack using 2 available slots. It receives its power supply from the host rack.[88]

The A Designs P-1 has the highest fidelity of all the A-Designs Audio 500 microphone pre-amplifiers. It is known for capturing the essence of its older and larger sibling, the extremely popular Pacifica. It has a big low end, slightly forward midrange and extended highs. It shines on vocals or any instruments with sharp transients.[89]

The Neve 1073 Microphone Pre-Amplifier/Equalizer was launched in 1970. The 1073 is the first choice of leading producers and artists, delivering the unique Neve sound on some of the most famous recordings of the past 40 years. The big, punchy sound of the 1073 complements any musical genre. Handcrafted and completely hand-wired by Neve's dedicated professionals in Burnley, England, the modern-day 1073 is produced to the exact specifications of the original modules. Considerable resources have been devoted to the acquisition of the original components to ensure that the sound remains the same. The Class A design 1073 microphone pre-amplifier features 3 bands of EQ, with one fixed high frequency band, two switchable bands with cut and boost capability, and a high pass filter. All Neve channel amplifiers are designed to accept signals from a wide range of microphone and line sources. The Neve 1073 microphone pre-amplifier and EQ combination adds warmth and depth to recordings, bringing out subtle ambience,

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88. Dan. Kennedy, "MP-500NV Single Channel Preamp Module," Great River Electronics, http://www.greweb.com/mp-500nv.html [accessed December 23, 2013].

89. Lisa Montessi, "500 Series Microphone Pre-Amps," A-Designs Audio, http://www.adesignsaudio.com/500-series-mic-pre.htm [accessed December 23, 2013].

maintaining spatial positioning, and more effectively capturing a precise image. This is considered by many to be the essence of the Neve sound.[90]

Brent Averill Enterprises in North Hollywood, California began extracting Neve microphone pre-amplifiers out of Neve consoles more than twenty years ago. They developed a stereo pair unit called the Neve 1272 sold with a power supply. It quickly became a "go-to" pre-amplifier for vocals and drums mostly because it colored the sound with a little warmth or as some would say added some "punch."[91]

Compressors

Compressors are the third link in the vocal recording chain. There are only three compressors that seem to be noteworthy. Through the years the Tube Tech CL1B has proven its value providing clean musical compression for thousands of track recordings. It is probably the most famous of the compressors and once it has been heard one can understand why. For vocals and instruments like guitar bass and keys (regardless of musical genre) the CL1B delivers extremely musical and smooth compression, just what is needed to make the track fit into your mix without any muddiness or distortion, even at very extreme settings. The CL1B is very easy to use; just dial in what you need and feel the action immediately. You do not have to doubt whether the setting works, it is always an obvious difference. The compressor also seems to add an additional layer of thickness when recording a vocal. It acts like a thickness or coloration, but does not seem to change the nature of the acoustic sound. The number of hit records featuring the CL1B is countless and the majority of all top singers and musicians demand the CL1B for their

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90. David Clark, "1073 Mic Preamp & Equaliser," Ams-Neve, http://ams-neve.com/products/outboard/1073-mic-preamp-equaliser [accessed December 23, 2013].

91. Brent Averill, "1272 Module," Brent Averill Enterprises, http://www.baeaudio.com/pages/products/index.php?option=com_content&view=article&id=33&Itemid=266 [accessed December 23, 2013].

recordings regardless of the genre they perform. As an example, it is a well-known fact that all the Hip Hop/Rap stars, like Kanye West, depend heavily on the CL1B for their vocal performance. Today more than 16,000 CL1B's have been manufactured and are hard at work all over the world.[92]

Universal Audio's 2-1176 Compressor features two channels of the Class A circuitry, custom designed output transformers, and FET-type gain reduction from the original 1176LN. They are also stereo matched for improved stereo imaging. The unique sound and stereo capabilities of the 2-1176 have beckoned a wide range of artists and engineers, like Norah Jones with Jay Newland and Helmet with Charlie Clouser, to call on its services while recording their albums. With switchable options of "Link" mode for stereo compression, or "Dual" mono mode, the 2-1176 offers twice the character and twice the features of the original in one unit and includes the classic "All" buttons mode. With over 2000 units manufactured and in use since the year 2000 re-birth of Universal Audio's 1176LN, it is clear that the public is still as crazy as ever about the legendary sound of the 1176.[93]

The Universal Audio/Teletronix LA-2A Classic Leveling Amplifier is true to the original design of the famed tube-amplified T4 Electro-Optical Classic Compressor. Audio professionals around the world revere the LA-2A. The original was frequently acknowledged for its smooth, natural compression characteristics. A unique tube-driven electro-optical attenuator system allows instantaneous gain reduction with no increase in harmonic distortion. This was an accomplishment at the time that is still appreciated

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92. John G. Petersen, "CL 1B Opto Compressor," Tube-Tech, http://www.tube-tech.com/cl-1b-opto-compressor-9.html [accessed December 23, 2013].

93. Bill Putnam, "2-1176 Twin Vintage Limiting Amplifier," Universal Audio, http://www.uaudio.com/hardware/compressors/2-1176.html [accessed December 23, 2013].

today. This faithful Universal Audio reissue marks the rebirth of the Teletronix LA-2A. Much care has been taken to ensure that every new LA-2A provides the performance characteristics of the original. Each unit is point-to-point hand-wired and built in Scotts Valley, California, with every component carefully evaluated for authenticity. As a result, today's LA-2A is bringing the same legendary compression characteristics of the original to many recordings around the world.[94]

Equalization

Equalization is the fourth and usually the optional link in the chain for recording vocals. It is not often used mostly because it is not needed when recording the vocal. The main thing to remember is that whatever EQ is put on the vocal when it is being recorded cannot be removed later. For this reason, many producers and engineers avoid using EQ until later unless there is a problem or an issue that needs attention in order for the vocal to be recorded well. In this case, the only EQ that is surgical enough to address but not destroy said issues would be the Massenburg 8200. Coming from the innovator, George Massenburg, who invented the term "Parametric Equalization", the 8200 has been an industry standard for over twenty years, and can be found on virtually every major recording studio's stereo bus. Each of the five broadly-overlapping bands offers 15dB of Boost or Cut and adjustable bandwidth (or "Q") from 0.4 to 4. The lowest and highest bands also can be switched to Shelf mode. Quite simply, the 8200 is the archetype Stereo Parametric Equalizer. Its extraordinary resolution, benchmark transparency, generous headroom, and surgical precision have been the reference for

94. Bill Putnam, "Teletronix LA-2A Classic Leveling Amplifier," Universal Audio, http://www.uaudio.com/hardware/compressors/la-2a.html [accessed December 23, 2013].

many other equalizers but exceeded by no other. The 2U 8200 is a stereo unit that uses only one power supply.[95]

In summary, as one can see the equipment choices and options available for the vocal recording chain are vast and various. In the next chapter, the combinations of such equipment will be carefully considered.

95. George Massenburg, "GML Model 8200," George Massenburg Labs, http://www.massenburg.com/products/gml-8200 [accessed December 23, 2013].

CHAPTER 7

VOCAL CHAIN COMBINATIONS

The decision process of determining the equipment components of the vocal recording chain is probably the most crucial piece of the vocal recording puzzle. As one could deduce from the previous chapters, the components become more limited and specific as the vocal recording chain is built. In other words, there are more microphone choices than there are microphone pre-amplifier choices, more pre-amplifier choices than compressor choices and likewise more compressor choices than EQ choices. This is mostly because as one progress through the chain, the choices of quality components become fewer and more crucial to a pristinely recorded vocal. Out of the four components of the recording chain, the first two are the only requirements. The third step of compression is desirable, but not always available. The fourth step of equalization is optional and not often used unless there is a problem that needs a surgical solution.

The first step in finding the perfect vocal recording chain is to consider what voice-type you will be recording. Some microphones work better on certain voice-types than others. For instance, the Rode NT-1a, a very inexpensive mic, is one of the best for sopranos, but doesn't sound as good on male voices. Conversely, the U67 is a really nice microphone for tenors because it has a high frequency "present" sound that sounds good combined with the warm frequencies in male voices.

In considering the different colorations of soprano voices, let us first take a look at a soprano voice that is extremely bright. First of all, if the brightness is extreme, then one might consider at least trying a ribbon microphone. The ribbon microphones rarely

work for a singer (since they are so dark in quality) unless the voice is extremely bright, to the extent that no condenser microphone seems to work. If an engineer/producer finds themselves auditioning all the condenser microphones available on a singer and still feel the voice is extremely bright, then a ribbon might be the "one-in-a-million" solution. If a voice is just slightly bright, there is a possibility that a semi-warm condenser like the Neumann U47 might be perfect. It will just take a little of the harsh edge off, but still will have a very present sound. An AKG C-12 is also a microphone that adds even a little less warmth, but still leans more heavily to adding a darker color as opposed to adding an airiness or glossy quality.

For a soprano that is a light breathy quality soprano, a microphone that might work wonders would be a flat response microphone like the Telefunken 251 or even the AKG 414 (silver). These are both nice microphones and well-known for recording a voice pretty flat. The Telefunken 251 is better known for this than the AKG 414, but the 414 is known for adding more gloss or air to the sound. A rich heavy and warm sounding soprano would never sound very good on a U47, but needs a microphone to add a little bit of high end frequency to the rich warm character of the voice.

An alto that has a heavy rich tone almost comparable to a tenor, would probably sound great on a Neumann U67. This microphone tends to love warm rich voices. There is a certain amount of air or edge that this microphone tends to add to such a voice. It is kind of like the buzz you hear acoustically in a cello. The body of the instrument is very rich and warm, but there is a high end frequency or air with the bow hitting the string that has to be captured in the recording. It is exactly the same thing for a warm alto or rich male voice.

A lighter alto with less bump in the lower frequencies would probably sound better on a microphone like the Sony C-800 that has a bump in both the lower and higher frequencies. This microphone tends to fatten up a voice that might be a little thin.

The Sony C-800 is also a good microphone to use on an alto that perhaps has a bump in the middle frequencies of the voice somewhere around 2 KHz. This type of singer would tend to have a brighter, or more nasal sound. In this situation, the microphone would be smoothing or tapering the harsher quality existing around the 2KHz frequency in the voice.

Male voices in general all tend to sound good on a Neumann U67. The tendencies of that microphone to have a bump in the upper frequencies are a match made in heaven with the male voice. Tenor voices especially sound good on that microphone, but also would probably sound great on the Telefunken 251. Male voices in particular need not be paired with a microphone that tends to add more warmth or richness in color unless the male voice is bright and nasal. In this case a Neumann U47 or one of the microphones built to emulate it, would probably be just what is needed.

In choosing the microphone pre-amplifier, one has to consider what colors are already "on the table" so to speak, as it pertains to the voice-type and microphone that has already been chosen. Some microphone pre-amplifiers tend to have a characteristic that is considered to be more smooth or, as some engineers describe it, more "buttery." Other microphone pre-amplifiers tend to be more flat or natural sounding. The "buttery" microphone pre-amplifier would not normally be paired with a warm microphone and also a brighter microphone would not be paired with a microphone pre-amplifier that has

harsher characteristics. Sometimes a microphone pre-amplifier has a couple different settings that will affect the color. These should also be considered through trial and error.

The compressors will tend to add a bit of color to the vocal but it is usually always an improvement. The use of the term color here is more in terms of making the vocal sound fuller and fatter. This quality is what I would call approaching more analog characteristics. It is along the same lines as a doubling effect, but not really the same principle being applied. The vocal will tend to sound more present, thicker and richer than when the compressor is bypassed. Both the attack and release should always be set to slow, around 3 o'clock, when recording a vocal. The ratio should also always be set to a slight setting of a 2:1 ratio unless one is recording a very hard-hitting vocal. The threshold is always at its best around zero or slightly under. The gain or output is sometimes bumped above zero, by 5-10 dB. Essentially, the compressor should be sonically undetectable, mostly being used to capture some of the additive characteristics of the compressor and catching a few of the hard-hitting transients that might occur while recording the vocal.

An equalizer is not normally used in the chain when recording a vocal. But if there is a frequency in the vocal that is presenting itself as a problem, then the Equalizer might be quite helpful. The Massenburg GML 8200 is a stereo EQ that might be one of the most affordable for what it does. It can surgically reduce or increase a frequency in the voice and do it by the thinnest of margins. This is a good piece to have in ones' rack of equipment, but more of a luxury than a necessity.

In summary, as you can see pairing microphones with a particular voice is mostly about filling in a frequency range in the voice or carving out a range in the frequencies

that might be a little too prominent. The further down the vocal chain one progresses, the more perfect the pairing will start to sound if it is done correctly. And, if done correctly where the voices are recorded true to the acoustic instrument, very little equalization will be needed on the voice in the mix process. If the frequency response graph on the end of the chain can be close to flat without any dramatic peaks or dips, then the engineer has managed to capture a relatively true recorded waveform of the human voice.

CHAPTER 8

PERFORMANCE ON BOTH SIDES OF THE GLASS

The Singer's Perspective

Achieving a quality recording of a vocal begins with the engineer and the set-up of equipment that will be used, moves to the singer and their performance in the vocal booth, and then culminates on the other side of the glass in the control room with the producer and/or engineer. Let us first take a look at the singer's perspective and the performance aspects of delivering a quality vocal.

The singer steps into the recording studio with some amount of trepidation depending upon the background and years of experience that the singer might or might not have. Upon a singers' arrival at the studio, the producer and engineer will hopefully create a welcoming and friendly environment that will put the singer at ease. First, the singer should be brought into the control room and after chatting a bit allowed to hear the basic track of what they will be recording. If it is an a capella track and nothing yet exists, then a good healthy discussion with the producer on what the plan and approach is would be helpful. Even when doing an a capella vocal recording, there should initially be a plan of first creating some kind of a basic foundational track and then building the other vocals from there.

After the initial meeting in the control room, the singer should be given a little time to get into the vocal booth and get themselves settled and comfortable with their surroundings. There should always be a music stand, stand light, pencil, cue box, comfortable headphones with a good seal and a comfortable chair or stool. This type of

set-up will go along way in fostering a relaxed and creative environment for the singer/performer. Singing is a very personal endeavor as the singer carries their instrument around with them and is part of their body and emotions. The instrumentalist can expect their instrument to act pretty much the same when they pull it out of the case, with the exception of the temperature fluctuations they have to account for. The singer on the other hand does not have it quite as easy. The singer carries their instrument inside their bodies and is affected by many things; sickness, personal hydration, emotions, and other life situations. For this reason, the singer must remain aware of the state of their instrument and try to settle themselves down in the vocal booth, before they begin to sing.

When they put the headphones on, they need to find a position of wearing the phones that will allow them to best hear themselves. Optimally, the studio will have a cue system that allows the singer to adjust a stereo track, a piano track or another harmonic instrument and then a potentiometer to raise or lower the level of their own vocal. One common way to wear the headphones is to take one ear off and leave an ear on. The only thing to watch for with this is whether or not the pan potentiometer is panned left or right. It should be panned to the ear you are using. Another less common, but effective way to wear the headphones is to pull one headphone off half-way behind one ear and the other headphone half-way off in front of the other ear. This allows both ears to hear the track in both sides of the headphones and also allows the singer to hear themselves in the room with both ears. Singers tend to make adjustments as they sing according to what they hear. For this reason, it is important to get the headphone position right. If all aspects of the headphones are not correct and the singer is not comfortable,

the recording will never capture the singers' best performance. All the best gear and the most favored vocal recording chain cannot manufacture a good vocal performance. The singer has to be able to hear themselves in the studio in the same way they hear themselves in practice, at home or on stage.

The singer needs to learn how to communicate with the producer and engineer. When they start recording a pass and are just not comfortable and cannot hear, they need to stop and say so. If the singer records a pass that they know is bad and do not want to keep it, they need to stop and ask for another shot at it. The singer needs to always be very candid about how they are feeling about the vocal and how much more recording they feel they need to do or can do. Sometimes the singer will begin to get tired and at that point needs to take a break and hydrate with something hot or cold to drink or possibly recharge their energy level with food. Singers need to remind themselves to take their time in the studio in order to get the best vocal possible. Once the singer feels they have recorded a good take for every phrase, they should do one more complete performance for safety and for the performance aspect of it. Although some like to record phrase by phrase with a lot of punching in, it is often a better idea to record quite a few full performances of the song or at least do this in certain sections of the song. Doing the vocal this way sometimes insures more of a true performance from the singer on every track. Making a vocal comp track later after the vocals have all been recorded is the most typical way of tying together the pieces of several different emotional performances. This approach is one that the singer should either suggest or ask for during the recording session in order to insure a better performance.

This same discussion can also apply when discussing the act of recording a group of vocalists. With that in mind, the group of vocalists not only need to be able to hear themselves, but also be able to hear the other singers in the group. Taking both ears off slightly will help this dramatically as the singers will be able to acoustically hear the other singers that are standing to their right and to their left.

The Producer-Engineer Effect

The effect that the producer and/or engineer has on the recording of the vocal goes far beyond the nuts and bolts of setting up the session and moving the right gear into place. The attitude and manner at which the producer and/or engineer conducts the session can greatly affect the performance of the singer and the quality of the recorded vocal. There are, of course, certain technical responsibilities that the engineer/producer needs to fulfill. They have to set up the studio, the session files, choose the vocal chain, and other tasks; but, they also have to manage what they say and how they say it to the singer in order to achieve the best possible vocal performance. This may seem trite to some, but as stated before, the voice is very personal and usually housed in singers who are both emotional and creative. The two conditions seem to go hand in hand more often than not. The producer/engineer needs to remember that it is their job to make the singer comfortable and at ease with recording, then just allow them to sing until they feel they have done all they can do.

The atmosphere set up by the producer and engineer in the studio is also quite important. Some think that lighting makes a difference in a singers' performance. Whether it be low-level lighting, bright lights or the use of candles, whatever makes the singer feel comfortable with the ability to emote the lyric is the best approach.

The best method is to record full performances from top to bottom. This is one way that magic might begin to occur in the vocal performance because the singer feels as if they can "get into" the moment. It is always a good idea for the producer/engineer to continue to ask the singer how they feel, if they need anything and what they think about the track they just recorded. The producer/engineer can then offer suggestions that will make it better. The singer never needs to feel as though the producer/engineer is in a hurry or trying to rush. Once the producer, engineer and singer have stopped wanting to do another pass, it might be a good time to do some listening. After listening and determining that possibly all that is needed has been recorded, the engineer can begin to do a "comp" or composite of the best of all the vocal performances. After a comp is put together, it will be much easier for the producer, engineer and singer to make a final determination as to whether or not any more recording is needed. At the end of a successful vocal recording session, there is a feeling of completion amongst all the people involved in the session.

If the singer gets tired or there is a feeling of not having enough vocal recorded yet, the singer should be encouraged to either take a break or rest and come back at another time to finish up the recording. It is highly likely that the voice will sound differently on a different day, but sometimes once the singer gets warmed up, it is hard to tell much difference. Of course, it is the producer/engineer's responsibility, if this occurs, to make notes of all the equipment used and all the settings and levels in order to try and recreate the same session environment at a later date. At the end of each vocal session, it is best for everyone to take a generous amount of time to consider the

reference mixes of what has been recorded. There is no reason to be in a hurry when trying to be creative.

The producer/engineer effect is also important to consider when recording group vocals. All of these concepts can be applied to that as well. The fact that there are more singers in the room together can sometimes be a challenge for the producer/engineer but evenstill has to be managed in order for the session to not get out of hand with excessive talking, etc. It's a fine line or balance for the producer engineer as singers will always be more inclined to talk and visit with others that are in the room recording with them. The professionals tend to be very aware of the pace and timeframe the producer is operating under and because of that, know when to turn the socializing on and off. Professional singers know that they have been hired, first and foremost, to sing what they are asked to sing.

When working with group vocals, the producer/engineer often wants to double or triple the different parts that are being recorded. This technique usually occurs according to taste to produce a desired thickening effect and also varies according to genre. In other words, a jazz record will probably only double vocals in the case of a jazz vocal ensemble or group, but not a lead vocal. A pop recording might double the lead vocal in order to make it sounds thicker and most certainly will double or even triple the recorded choir tracks. In the case of background vocals, sometimes those are layered multiply times. In other words, there could be six vocal tracks of the same part or harmony. So, if there are three harmony parts, that would be a total of eighteen background vocal tracks.

CHAPTER 9

RECORDING VOCAL PERCUSSION

Recording vocal percussion is a technique altogether different from recording a vocal that is being sung or voiced. Vocal percussion is produced largely by plosives which produces harsh transients in the vocal waveform. The vocal percussion will also be recorded differently depending on what angle the plosives are directed at the diaphragm of the microphone. An attempt will be made here to discuss several of the different approaches to recording vocal percussion.

Several approaches to recording vocal percussion involve the use of the frequently-used vocal condenser microphone. The upside to using a vocal condenser microphone is the presence of the voice that will be captured. The downside is that because of the bass proximity effect, the vocal percussionist cannot get close enough to the microphone to pick up the low end frequencies that are occurring in his/her throat. If the vocal percussionist gets too close and tries to record without a pop screen, there will be all kinds of "breath pops" in the recording. A fix to this approach might be to record a scratch vocal track on the condenser microphone, just to get a general idea of what is desired. After a scratch track has been created, there are several ways the vocal percussion track can be captured. If the vocal percussionist has the ability to break out all the sounds, doing them one by one, that would be the easiest in the long-run. The vocal percussionist can begin building the track with the basic kick pattern, layering it with the snare, high hat, tom and cymbal tracks. With this technique, each sound can be treated with the kind of equalization that will only enhance each individual sound. Even still,

recording a sound that truly resembles the kick sound will be difficult without getting close to the diaphragm. The only other option for kick with a condenser is to record the vocal percussionist hitting their chest wall. This will create a sound that contains more of a low frequency thump in the 50-100Hz range. Recording the sounds separately though might be extremely difficult depending on the vocal percussionist and their level of skill.

An alternate solution in using condenser microphones is to set up several different condenser microphones around or in front of the vocal percussionists head so that the percussionist can essentially work the microphones as if the microphones were a drum kit. This is probably the most difficult approach, but would allow the engineer to later have a little more control over the equalization of all the instruments.

The third option is to use a dynamic handheld microphone like the Sennheiser e945 that would normally be used onstage for vocal recording. This is where you will get your best kick sound because of the ability to get close to the diaphragm. There will be more bass response and more punch with this technique. The snare will also have more thump and body with this technique, but the high frequencies will not be quite as crisp, that just being the nature of the dynamic microphone as opposed to the condenser.

The best approach to recording vocal percussion is probably a combination of several of these techniques. The vocal percussion track needs to be thought of as a complete “kit” of sounds. One would not want to record an entire drum kit on one microphone, so neither do you want to record the vocal percussion “kit” on the same microphone. After recording some kind of scratch track for reference, the percussionist could attempt to record the kick drum sound with the dynamic hand held microphone and also attempt recording the snare separately on the same microphone. The dynamic

microphone will also give a nice amount of body or punch to the snare sound. Doing these drum sounds separately may not be possible with the vocal percussionist, but it will at least be worth it to have some separate sounds that could be used to enhance the track later. At some point, a move to the condenser microphone might be appropriate for all the high frequency high hat and cymbal work. Then the only other thing that the vocal percussionist might want to use the hand-held dynamic microphone for is any technique that involves a swishing sound where the soundwaves are hitting the diaphragm from many different angles creating an oscillating effect. After finishing the recording, the engineer will most likely have to do some editing to clean up unwanted room noise and a bit of human error. Beat Detective, Elastic Time or Sound Replacer are tools that can be used to clean up and edit the vocal percussion track. Recording vocal percussion is an area that involves quite a bit of trial and error while working with the vocal percussionist. The techniques to record vocal percussion will continue to evolve and be perfected over time.

CHAPTER 10

THE ANALOG-DIGITAL CONVERSION

Prior to 1976, all music was recorded on analog tape recorders. The audio signal was recorded as magnetic patterns that would rise and fall in strength as the signal waves would do. Digital recorders, on the other hand, store the audio signal as a numerical code of ones and zeros. Both analog and digital recorders can accurately playback a signal that sounds like the input signal, but sonically there are subtle differences between the two.[96] To better understand these differences, let's first consider how these sound waves are represented.

Signals can be expressed in two ways, either continuous or discreet. When signals are represented as continuous data, they are described as analog. When each data point of a signal can assume an infinite number of values, that is considered to be an analog representation. If between any two of these data points (no matter how close) another value exists, then that is considered to be an analog representation. However, when signals are represented as discreet points of data, those are described as digital. When signals can only assume a finite number of values, and when there exist gaps between these values, that is considered to be a digital signal. An example of an analog representation of data could be illustrated by a potentiometer. One can turn the knob continuously to any value without gaps or jumps between values. A rotary encoder, a knob that "clicks" when turned, is a good illustration of digital. It only assumes predefined fixed values with gaps in between values. Digital also imposes one more

96. Bruce Bartlett, and Jenny Barlett, *Practical Recording Techniques*, 5th ed. (Burlington, MA: Focal Press, 2002), 165.

constraint. Not only is there just a predefined set of possible values, but they are only allowed to occur at fixed points in time. This is the reason digital signals are also referred to as discreet-time signals.[97] Digital is only capable of representing a finite band of frequencies when the analog frequency band is unbounded. The frequency range of digital audio is theoretically unbounded, but would take so much storage space as bandwidth gets higher. However, this should not, in reality, be such a disadvantage in digital, as the said upper frequency bands represented would be beyond the range of human hearing.[98]

The digital signal is usually called "clean" because it adds almost no noise or distortion to the input signal. Some analog tape recorders add a little "warmth" to the sound, but it is likely due to slight third harmonic distortion, recorder head bumps (bass boost), and tape compression at high recording levels. Compared to analog, digital recordings have less hiss, frequency-response errors, modulation noise, distortion, unsteady pitch and print through. However, those characteristics are inaudible on the best analog tape recorders when the machines are taken care of and kept in alignment.[99]

The very highest frequencies contained in an analog audio signal are thought to contribute to the "warmth" of analog sound that seems to be missing from digital recordings. For example, with the scraping of the horse hair of a violin bow against the strings of the instrument a transient signals is created upwards of 50 kHz. Some argue that, although humans cannot hear frequencies that high, they create audible frequencies through a process called heterodyning. A human usually hears this phenomenon as a

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97. Colby Leider, *Digital Audio Workstation* (New York: McGraw-Hill, 2004), 23.
98. Ibid.
99. Bruce Bartlett and Jenny Barlett, *Practical Recording Techniques*, 5th ed. (Burlington, MA: Focal Press, 2002), 165.

difference between adjacent frequencies. So, the difference between frequencies 50 kHz and 51 kHz might feel to a human like hearing a 1 kHz burst of sound. However, the perceived "warmth" of analog is more likely to be the way vacuum tubes and transistors operate. Analog recording equipment often uses vacuum tubes throughout the entire signal path, whereas digital recording devices generally use only solid-state devices like transistors and diodes.[100]

A vacuum tube being used in the analog recording chain has its own set of problems. They produce a lot of heat and are essentially like a light bulb, which means there is a short life span for each tube. Many think that the vacuum tube is the biggest contributor to the analog "warmth" of sound, but only if turned up or operated higher than their linear regions. Otherwise, many feel it is difficult to distinguish the so-called "warmth" in the sound quality. When they are driven into nonlinear regions (i.e. driven to distort the input), measurable differences in the quality of sound begin to appear. Tubes react with what is called soft-clipping. They gradually begin to distort as they are driven "hotter" with an increase in input amplitude. Transistors usually exhibit hard-clipping. This is where the input gets clipped very quickly.[101] Distortion introduces harmonics to the output signal. For this reason, amplifiers are often driven to distort or "thicken" the sound and making the timbre of sound seem richer. Tubes and transistors introduce harmonics in a very different way. Transistor distortion has a strong third harmonic and because of that produces a sound that has a more harsh or biting quality.

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100. Colby Leider, *Digital Audio Workstation* (New York: McGraw-Hill, 2004), 27.
101. Ibid.

Tubes tend to produce a rich set of harmonics simultaneously, particularly in the second, third, fourth and fifth harmonic.[102]

One way that modern digital studios are infusing the analog sound into the digital domain is to record all signals with vacuum tube based microphones and microphone pre-amplifiers before converting them into digital form. Some also go a step further and record to analog tape prior to mixing in the digital domain. In essence, this is using the analog recording chain as an effects processor just for the feel that it injects into the sound.[103]

To break down the digital recording process into somewhat simpler terms, an audio signal is essentially recorded with this four step process:

1. The signal from your mixer, preamp, or audio interface is run through a lowpass filter (anti-aliasing filter) which removes all frequencies above 20 kHz.[104]
2. The filtered signal passes through an analog-to-digital (A/D) converter. This converter measures (samples) the voltage of the audio waveform several thousand times per second.
3. Each time the waveform is measured, a binary number (made of ones and zeros) is generated. This number represents the voltage of the waveform at the instant it is measured. This process is called quantization. Each 1 and 0 is called a bit, which stands for binary digit. The more bits that are used to represent each measurement (the higher the bit depth), the more accurate the measurement.

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102. Colby Leider, *Digital Audio Workstation* (New York: McGraw-Hill, 2004), 30.
103. Ibid.
104. Ibid., 166.

4. These binary numbers are stored on the recording medium as a modulated square wave recorded at maximum level. For example, the numbers are usually stored magnetically on a hard disk.[105]

For playback, the process reverses itself.

1. The binary numbers are read from the recording medium, such as a hard disk.
2. A digital-to-analog (D/A) converter translates the numbers back into analog signal made of voltage steps.
3. An anti-imaging filter (low-pass filter, smoothing filter, reconstruction filter) then smooths out the steps in the analog signal, resulting in what is assumed to be the original analog signal.[106]

How original this analog signal sounds during playback is dependent on several of the following factors. There is a coding system called *Reed-Solomon* which happens during recording and then decodes during playback. This coding/decoding system corrects for missing bits by using redundant data.

Also, if the digital recording is on a defective medium such as a scratched compact disc, then errors or missing samples can occur. In this case, errors can be corrected by interpolation. This algorithm looks at data before and after the blank sample and "guesses" what its value should be. If the errors are too much to correct, the audio either has a silent spot or a burst of noise.[107] Sometimes this is heard as a digital spike.

One of the primary factors that will affect the sound quality of the digital-to-analog conversion process and vice versa is the A/D and D/A converters. Different

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105. Colby Leider, *Digital Audio Workstation* (New York: McGraw-Hill, 2004), 166.
106. Ibid.
107. Ibid., 168.

converters will sound better or worse than other converters depending on the manufacturer or brand.[108]

Bit depth is also another way to increase the quality of the recording. 16 bit is currently the standard bit depth for Compact Disc, but will sound better if created from 24 bit recordings. The more bits that are used the smoother and more transparent the recording will sound, but will need more disk storage space.

Dither, or low-level noise, should be added to a 24-bit recording during mastering. The file should then be exported and saved as a 16-bit recording, copying the 16-bit recording to a CD. The dither will help the 16-bit recording sound more like the 24-bit recording. Dither lets you retain most of the quality and resolution that you recorded at 24 bits, even though the recording ends up on a 16-bit CD.[109]

Sampling rate or *sampling frequency* is the rate at which the A/D converter samples or measures the analog signal while recording. For example, a rate of 48 kHz is 48,000 samples per second; that is 48,000 measurements are generated for each second of sound. The higher the sampling rate, the wider the frequency response is of the recording. According to the Nyquist-Shannon theorem, the upper frequency limit of a digital recording is one-half the sampling rate. Compact discs are 44.1 kHz, so their frequency response range extends to 22.05 kHz. Sampling rates for high quality audio are 44.1, 48, 88.2, 96 or 192 kHz. CD quality is 44.1 kHz/16 bits. A 96 kHz sampling rate can be used on DVD. State-of-the-art is Super Audio CD or linear PCM at 192 kHz/24 bits, but the more likely setting is 96 kHz/24 bit.[110]

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108. Colby Leider, *Digital Audio Workstation* (New York: McGraw-Hill, 2004), 169.
109. Ibid., 170.
110. Ibid.

According to Leider, digital recording is based upon the Nyquist-Shannon theorem which defines the "Nyquist frequency" as half the sampling rate. The theorem proves that the original analog signal can be reconstructed exactly from its samples if the highest frequency in the signal is less than the Nyquist frequency. This happens when the original signal is sent through a lowpass (high-cut) filter to remove frequencies above the Nyquist frequency. If the analog signal is not filtered, then aliasing will occur. This is when high frequencies above the Nyquist frequency appear as lower frequencies. In other words, inaudible ultrasonic frequencies are converted to audible tones and is essentially why anti-aliasing is needed. If you low-pass filter the analog signal before it is sampled, then the only waveform that can pass through the sample points is the original analog waveform.[111]

Most A/D and D/A converters today use a process called oversampling to improve the sound. Oversampling is sampling an audio signal at a higher rate than needed to reproduce the highest frequency in the signal. For example, sampling a 20 kHz audio signal at 8 times 44.1 kHz is called "8x oversampling." This process is then followed by a digital low-pass filter and a gentle-slope analog anti-alias filter. The result is a less phase shift and less harshness compared to a steep, "brick-wall" analog filter used alone.[112]

A digital audio system samples the analog signal several thousand times a second and quantizes (assigns a value to) each sample. Sampling rate affects the frequency response, while bit depth affects the dynamic range, noise, and distortion.[113]

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111. Colby Leider, *Digital Audio Workstation* (New York: McGraw-Hill, 2004), 171.
112. Ibid.
113. Ibid.

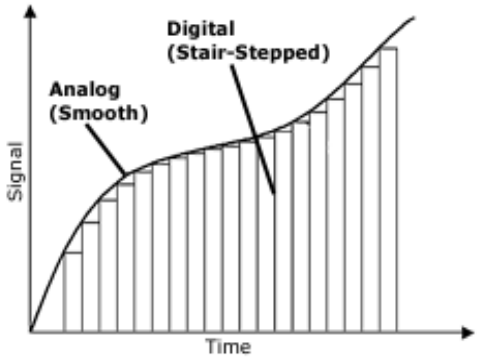

Figure 10.1 – Analog Versus Digital Sample Comparison Graph

All these components of digital sound affect the quality of the recorded vocal. The audio engineer's ability to understand these concepts and account for them is essential in recording a vocal that is a true representation of the original. The engineer and/or producer's quest to "warm up" vocals and recapture some of the character of the analog sound is one that is getting easier to produce with every step forward in recording technology. To some, there is still a vast difference between the vocal quality in the digital domain and that which was produced in the analog era.[114] However, as discussed, there are tools and techniques that have emerged and can be implemented in today's digital studio environment. These techniques get us closer than ever before to finding the perfect solution to incorporating the best attributes of the "analog" sound.

Some people still debate analog versus digital, but realists have moved past that discussion. Their only debate now is, which analog to combine with which digital. As mentioned before, the answer can be as simple as capturing to tape to take advantage of

114. Matthew Zimmern, "Exploration of Past and Present Music Production Techniques and their Influence on the Resulting Music Recordings" (master's thesis, University of Massachusetts Lowell, 2010), In ProQuest Dissertations & Theses, https://iiiprxy.library.miami.edu/login?url=http://search.proquest.com/docview/366411302?accountid=14585 (accessed October 14, 2012).

its particular "sound," then immediately transferring the tracks to a digital system before tape wear, stretching, or other things begin to occur.[115]

Or the answer might be more complex, where a studio becomes a case study in "mix and match." Most *Digital Audio Workstations* allow external hardware as inserts, in the same way as a plug-in; one can even insert a tape recorder, and process the audio through that. But without a tape recorder to get that tape sound, maybe a tape simulation plug-in would be just what is needed to achieve the desired sound.[116]

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115. Craig Anderton, "Roundup: Analog/Digital DAW Synergy," *Electronic Musician*, May 2011, 44.

116. Craig Anderton, "Roundup: Analog/Digital DAW Synergy," *Electronic Musician*, May 2011, 44.

CHAPTER 11

POST-RECORDING AND PRE-MIX CONSIDERATIONS

Once the recording engineer and producer feel as though the vocal is pretty close to being complete, the recording moves into a phase that might be referred to as post-recording/pre-mix considerations. This might not mean that recording the vocal is absolutely done, but it means that some things need to be edited and considered before anyone knows for sure if the work is actually complete. The post recording phases that are discussed are vocal editing, vocal tuning and reference mixing. Then once a final decision is made to go to mix, there will be some pre-mix preparation that will occur.

Vocal Editing

If the vocal is recorded from start to finish in one "take" or as one whole recorded vocal track, the vocal editing technique of "fading in" and "fading out" still has to be applied to the beginning and the end of the recorded vocal track in order to avoid a digital pop or click at the point where the recorded region begins and ends. If a whole track is recorded and there are multiple "punch ins," then a crossfade has to occur at the juncture of every "punch in" and "punch out" point to avoid digital pops or clicks.

As has been discussed in previous chapters, it is more common for the vocals to be recorded in multiple takes using multiple playlists. Once the producer/engineer feels there is enough of every phrase in the recording, a decision is made for the engineer to create a composite of all the best portions of all the different performances. It is typical to first of all find the recorded vocal track with the very best performance, duplicate it and rename it as the "comp." Then it is usual for the engineer to begin to comb through

the vocal from top to bottom, allowing the producer to hear all the different options from different performances. The best of every phrase of the lyric is copied into the composite track. Once the comp track is put together and everyone agrees on it, it is time to do some cross-fading. Each cross-fade always needs to happen before a breath, after a breath, during an "s" or during another silent spot. This is the only time that a crossfade can be made were it can seamlessly be done undetected for a solo vocal. On group vocals, the endings of group vocals can be lined up by cutting the endings of a held note and either dragging them shorter or elongating. If it is a group of vocals, the only trick to this technique is that the crossfades on the vocals must all happen at a different moment in time for it to be inaudible. The onsets of words can also be lengthened by either the same clip and drag/crossfade technique or by time stretching the word. Sometimes time stretching is the best on beginnings of phrases because it is only slight and just helps the entrance of all the members of the group be more on time.

There is a program called *Vocalign* that some vocal editors like to use to align group vocals. If some of the singers happen to miss their entrance rhymically or are not in time with each other, the engineer can highlight the section of vocals and select the *Vocalign* plug-in. The vocals will line up with each other when *Vocalign* is selected. However, in the view of this writer, such automatic editing can sometimes be a little mechanical, sacrificing the more natural sound that can be achieved by moving entrances "by hand." In essence, if there is any editing done, the goal is for it to sound like nothing was done.

Once the vocal waveforms are edited with crossfades and other small manipulations, a good suggestion would be to listen to the vocal waveform or group of

vocal waveforms soloed and at a fairly high volume level. If the vocal seems to have exactly the correct feel and emotion that was intended, then it might be time to move forward. If that is the case, the comp track needs to be duplicated and then renamed as a consolidated file. Then the entire vocal file needs to be highlighted all the way to the beginning of the session (all the way to the left) and consolidated under one of the file menus. To click at the end (right) of the vocal track and highlight it all the way back to the beginning of the session (left) is a safe way to do it. After the track is processed and has been consolidated, the file should be double clicked and renamed with the title of the "song," "track name," and "cons" (i.e consolidated) in the file name. Hopefully, the singer(s) pitch was also "spot on," but often in today's marketplace, that is not the case. The next section of this chapter will address vocal tuning. This is a controversial subject with strong feelings on each side. Some feel vocal tuning destroys the integrity of the recording, others see it as a typical modern-day portion of the recording process and an extension of perfecting the performance. A decision that every singer will eventually wrestle with, but a realistic option and process that will be considered here in our discussion nonetheless.

Vocal Tuning

Vocal tuning is a process that is a part of most commercial recordings in our world today. Whereas one would hope that a singer would sing every pitch perfectly in tune, that is not usually the case. There are two vocal tuning tools that are used in the industry, but only one that is undetectable.

Auto-tune was the first tuning plug-in available in the digital recording platform. It is a little difficult to use seamlessly if using in the graphic mode. The tracking,

clipping and manual moving the pitch is a little clumsy to use. The automatic chromatic and voice-type tuning selection on the plug-in is a little better but should be set on a very light or slight tuning setting. The way to tighten the tuning a little more with this plug-in is to link several tuners in succession with each other. If this tuner is set to the tight or quick setting, the tuning will definitely be audibly noticeable.

Melodyne is a tuning software that can either be used as a plug-in or a in stand-alone mode (i.e. when the software can operate on its own apart from the software of the Digital Audio Workstation). The best way to use this tuner for vocals in this writer's opinion is in stand-alone mode. The one thing to be careful of in stand-alone mode is to make sure the tuner speed is the same as the session speed (i.e. 44.1 kHz, 48 kHz, etc). The *Melodyne* session should be started only after the Digital Audio Workstation (i.e. DAW) software has been closed out. Once the *Melodyne* session has been created and the preferences checked, the engineer can import audio from the DAW session audio folder. The consolidated file that was created earlier and renamed should be in the audio folder. When the consolidated file has been imported into the *Melodyne* session, clicking on the track will open that track in a separate window. The *Melodyne* program contains tools that enable the engineer to clip the notes into different segments and tune only portions of the note. *Melodyne* also allows the opportunity to elongate and move notes temporally if needed.

Melodyne allows you to raise or lower the formants in the vocal depending on how much a note needs to be tuned---a tool that makes the voice sound more natural. When the track window is opened up, one will see "blobs" or notes with a wavy line going through each "blob." This wavy line is what is known as the Pitch Drift. This is

the actually position of the pitch throughout the duration of the note that is sung. The Clipping Tool is a cutting tool that is used to clip or cut the pitch drift line in places where it appears that the note goes out of tune. The stylistic scoop into a note and any vocal "fall offs" at the end of a word of phrase are usually clipped with the Clipping Tool just before and after the waveform representation and left completely untouched. This aids in the goal of making the vocal sound natural, as, in essence, only the middle of the tone is raised. The pitches can be raised or lowered an indiscriminant amount with the use of the control key (control+click+drag). It is best if the tuning engineer has well-developed ears and can do the tuning by ear. Otherwise, the engineer will have to import a reference audio file from the track of the recording in order to get a pitch to tune to. Most experienced tuners tend to tune by ear just to keep their focus completely on the vocal without any distractions. Once the engineer has worked his way through the entire vocal, he/she should listen to the track once more to check his work. Often there will be sections of the tuning that need to be redone or even reset to the original waveform, depending upon how the vocal sounds in context. Once the engineer feels that the tuning is complete and feels natural, it is important to save the audio.

While saving the audio in Melodyne, the tracks need to be saved to "individual tracks" according to the longest reference track. The button labeled "spot it to Pro Tools" also always needs to be selected whether its going to Pro tools or not. The "saving audio" procedure will ask you where to put it. These tuned files should be stored in a folder in the session file called "Tuning Files." Once this is complete, the engineer should go to those files and rename them with names that include "tuned" instead of "cons." This will minimize any confusion later. After this is accomplished, *Melodyne*

can be closed and Pro tools or other DAW software opened again. At this point, the engineer can import audio under the File menu and navigate to the Tuning Files folder and import the files that have been tuned. Tuned files should be imported as new tracks, while any plug-ins that were being used on the untuned files should be moved to the newly tuned files. The untuned "Cons" tracks should be muted as well as muting the sends to the headphones. These original untuned tracks should then be made inactive and deselected or hidden from the Edit window. At this point, the session should now play with the newly tuned files and should be listened to by the engineer to make sure everything was imported correctly.

Reference Mixing

Reference mixing is merely a quick mix of some kind towards the end of the vocal editing and tuning process to allow the producer to make a determination as to the completion of the work. The reference mix should have the vocal mixed a little hotter than usual just to be able to hear if the vocal recording process is actually complete. If something else is needed, more vocals could always be recorded on a "Fix" track and laced into the main vocal track. The reference mix needs to employ a little light panning of instruments and vocals in order for the producer to be able to hear what needs to be heard. An analog simulation plug-in, like *Phoenix Crane Song Dark Essence* might also be used in order to give the vocal a little more of that warmth associated with the analog sound. These analog simulations are a technique that will be used even more in the mix stage. The vocals will probably need a little reverb. This should be done by creating an auxiliary track with a reverb insert and bussing (i.e. routing) a little of each vocal over to the auxiliary track. Reverb will keep the vocal from sounding too harsh and dry. More

reverb is normally used on ballads that fast songs, due to the time delay in the tail of the reverberation. Each vocal might or might not need a little equalization. If the correct vocal chain is used, equalization will most likely not be needed. The level of the vocal will need to be automated somewhat all the way through the song. In addition, the engineer will need to keep a lookout for any digital pops, clicks, other bad edits or room noise. Prior to mix would be the time to rectify these types of problems.

Pre-mix Preparation

Pre-mix preparation is mostly a matter of cleaning up the tracks and consolidating each track into one file. Any extraneous sound in between vocals or breaths should be highlighted and deleted. The recorded region should then be faded in just prior to the breath and just after the release of the last word of each phrase. Once the tracks are clean, they should be duplicated and then consolidated into whole files from the exact same starting point just prior to the downbeat. Most mix engineers prefer to get whole files on a DVD and then simply import those individual instruments into their mixing platform, which is occasionally not the same platform as the Digital Audio Workstation the recording engineer might have been using previously.

CHAPTER 12

FUTURE ADVANCEMENTS

Where do vocal recording techniques go from here? As the recording industry evolves, new concepts of microphone diaphragms are now being manufactured. Instead of using a circular diaphragm exclusively, a few microphone designers are beginning to experiment with the use of a slightly rectangular diaphragm with four different diaphragm quadrants for different sections of the frequency range. The frequency range is then divided equally into four sections with the diaphragms tuned to gather data in that range only.

There are also some new ideas being explored in the area of tuning microphones to individual voices. Custom microphones with customized tunings are already being built and sold privately, but, to the author's knowledge, have not been offered commercially as of this writing. It is a concept that parallels our discussion in this paper, that is, a quest to capture the most naturally recorded sound for each individual voice.

As for other emergent concepts, vocal groups are beginning to understand the importance of putting each voice on an individual microphone, in order to get a better recording of each voice. The acoustic environments for recording group and solo vocals, are being reconsidered, adjusted and improved. The most desirable combinations of the vocal chain are being talked about and debated amongst engineers and producers alike. And hopefully, even in an age of vocal editing and tuning, singers are aspiring to be better musicians and more precise with their recorded vocals than ever before. These are

exciting times in the recording industry to observe if technology advances the craft or the craft advances technology, each demanding more from the other for the sake of the art.

CHAPTER 13

DISCUSSION OF EMAIL QUESTIONNAIRES

The following nine Producer/Engineers were sent an email questionnaire which they responded to and returned for inclusion in this study: From Los Angeles, Ed Boyer, Dave Sperandio and Paul Tavenner, from Nashville, Bob Clark, David Hall, Tom Reeves, and Nathan Zwald, and from New York, Matt Pierson and Paul Wickliffe. These producer biographies including details of their careers and many of their professional recording credits are found in Appendix D.

The Importance of the Vocal Chain Combination

The first part of Question #1 that was posed to the pool of interviewees was as follows: How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? The first portion of this question had varying degrees of response with all but one person initially agreeing that the Vocal Chain Combination is important to the quality of the final recorded vocal sound.

According to Ed Boyer, if the vocal quality is going to be "beat up in post" he is more lenient, but if a "pure and vivid sound" is needed, then the Vocal Chain is "super important." Bob Clark responded that it is super important to choose the best gear possible that you have at your disposal at any particular time.

David Hall appeared initially to misunderstand the question because he wrote that the vocal is the most important part of the production. The author subsequently contacted him, explaining the question and asking if he wanted to amend his answer, but he failed

to respond. Perhaps he was emphasizing that capturing the natural vocal as comprehensively as possible before adding enhancements is of upmost importance.

Matt Pierson called the process of choosing the vocal chain essential. He said that he tries to pick whatever combination works best for that particular artist. He goes on to explain that "if you don't document the human voice in an honest and detailed way, you won't be working with something that contains all of the most important emotional elements of the artist's interpretation going forward."

Tom Reeves said that he considers the vocal chain combinations to be of vital importance to the recording. He said he believes that the chain of "microphone, preamp, EQ, and compressor combinations is where vocal magic occurs" even if he already knows what plug-ins he will be using in the mix.

Dave Sperandio believes that the most important factor is the talent of the performer, but the last 30% or so of the recording "excellence" is very dependent on the vocal chain. He also adds the comment that a good or maybe even very good recording can also sometimes be captured with inexpensive equipment. This author agrees since in her experience sometimes the perfect microphone for a performer is a mic that only costs $200.

Paul Tavenner believes that the vocal chain components are important but feels that microphone placement and the talent of the performer are more important. He goes on to make a comment that microphone pre amplifiers do not affect the color or sound of the recorded vocal. However, this runs counter to this writer's experience, after many years of experiencing a significant difference in color palate between microphone

preamps. Paul's dissenting opinion could stem from his being a somewhat specialized engineer necessitating the use of a limited number of microphone pre amplifiers. This author, on the other hand, intentionally built her own recording studio (bigdogstudios.net) with a broad array of different pre-amplifiers that contained various colors. For instance, a Neve pre-amplifier is known to have a darker sound or more color, while an API pre-amplifier is famous for having a very clean and flat sound. This has been the appeal historically for artists recording at a particular studio because they like the control surface (mic-pres) whether it be a Neve, API, etc and would usually book the studio according to the color of the desired sound.

In todays' digital studio, the trend is to have a broad palate of different microphone preamplifiers in order to be prepared for any particular need or use that might arise. It is also a little less expensive to buy an array of the 500v series preamps, for example, with quite an impressive rack of color as opposed to spending $250,000 on a vintage control surface like Neve or API that only contains one particular color. Those control surfaces are great boards with a beautiful sound, but the trend is to broaden your palate of color when it comes to the microphone pre amplifiers.

Paul Wickliffe did not specifically say whether or not he thought the vocal chain combination is important, but went on to describe the chain that he normally uses. An interesting comment that Paul Wickliffe makes is that he doesn't use EQ until the mixing stage, unless there is an obvious problem of some kind while recording. This author uses the same approach. Nathan Zwald from Nashville thinks that the vocal chain combination is "critical, with a focus on fidelity and natural capture of the performance."

The second part of Question #1 was as follows: What are your favorite Vocal Chain Combinations? The answers to this question were quite varied with a few popular microphones and other equipment being mentioned more than once. Bob Clark's favorite combination was the "ELAM-251 into an API pre with an LA-2." He also made the comment that he liked to use "just gentle compression on the recording." The LA-2 that Bob refers to is believed to be called an LA-2A by some engineers, although he may have just been abbreviating the letters.

Matt Pierson works a lot in New York at Sear Sound which has a Neve console. Bearing that in mind, a vocal chain combination that he uses frequently is "Neumann M-49 through the Neve 1081 console mic pre." Pierson goes on to say that he sometimes uses an Avalon M3 microphone pre amplifier. For male voices, he usually chooses a "Neumann U-47 'chrome top' through the Neve." He adds that there is also a rare Pultec microphone pre-amplifier that he uses on occasion. Pierson also comments that there are two other microphones that he likes for vocals, the Telefunken C12 and 251. However, he said that when these microphones are compared to the Neumanns, they usually fall short of being able to deliver the same kind of sound.

Tom Reeves has several favorite microphones and combinations that he likes to use. He is quite fond of the "AKG C-12 through an API 512 or 312, then compressed with a Tube-Tech CL1a or LA2a." The CL1a that Reeves refers to is a brother to the CL1B that was mentioned earlier in this paper. Even though the CL1B is thought by this author to be the ultimate vocal compressor, the CL1a is also good with the only difference being that it is hand-wired and tends to be a somewhat "tougher" or more aggressive compressor. Reeves adds that if the budget does not allow for a $10,000 microphone like

the C12, other good choices would include the Neumann 87, AKG 414, or Telefunken U47. Although Reeves believes that plug-in treatment can play a major role in his final vocal sound, he admits that he likes to "ride the fence" keeping "one foot in the analog vocal chain and the other foot inside the world of workstations and plug-ins." He feels that both sides of the chain are extremely important and relevant. Additionally, he says he tries to use the best of both sides of the chain.

Dave Sperandio says that his "go-to" vocal chain is a Telefunken AR51 mic through a Vintech 273 Preamp/EQ. This author is not familiar with this type Telefunken, but has used Vintech preamp/EQs before which seem to be adequate.

Paul Tavenner usually uses a Neumann U87 microphone and a Universal Audio 610 microphone preamp. He also adds that he has used a modified AKG 414BULS, Neumann U67 and U47 microphones. Tavenner adds that occassionally to get a "gritty sound," he has had excellent results with a Shure SM58. A Shure SM58 is normally a popular "live" microphone that has a very flat characteristic in color. According to this authors' experience, most uses of this microphone in the studio are with rock singers that sing with a hard, very loud tone. The diaphragm of this microphone is tougher than condensers and can handle the high amplitude a little easier. This author also owns and has had experience using the Universal Audio 610 on vocals, but prefers to only use this on a scratch vocal because it has a built-in compressor that is convenient for a scratch vocal while tracking. However, this pre amplifier is not one that this author would ever use for a high quality vocal.

Paul Wickliffe's default set-up is "a Neumann u47 (Sinatra / Beatle mic), a Hardy M-1 or API 512 mic pre-amp, a Massenburg GML 8200 EQ and a DBX 160X or UREI

LA-4 compressor." He has also been known to use a Neumann u87, M49 or vintage ribbon. He says he normally records the vocal flat except for a rumble filter and uses as little compression as is necessary.

Nathan Zwald's favorite vocal chain is "Brauner VM1-KHE (microphone) > Gordon Model 5 (preamp) > GML 8200 (EQ) > Tube-Tech CL1B or Retro Instruments 176 (compressor) > UBK Fatso or Anamod ATS-1 (analog tape saturation) > PCM4222 (A/D converter)." Nathan goes on to say that sometimes he will substitute different microphones like the "Telefunken 251 (vintage), Neumann 269 (vintage), or AKG C24 (vintage)." However, in his experience, he adds that "90% of the time the Brauner VM1-KHE wins."

Nathan goes on to talk a little bit about professional gear that actually subtracts from the sound. He discusses a scenario where he will be using the Gordon Audio preamps and then switch to a lesser quality but still professional pre amp and will hear information being subtracted from the sound. This has also been this authors' experience. Nathan quotes George Massenburg as saying that he likes to "keep a signal as clean and transparent as possible for as long as possible." Nathan also makes a point about matching the microphone to preamplifier and knowing how the sound of each will interact and how they interact electrically, specifically as it relates to resistance.

This author has not gotten into the build and electronics (resistance) of the microphones and pre-amplifiers in this paper, but agrees that this is part of the reason for the differences in color and "bump," i.e. (increase) in certain frequencies with varying combinations of vocal chain. This would be a good reason for more extensive

investigation regarding this topic. Nathan explains that this relationship effects the "EQ" of the microphone's response and ultimately the sound of the vocal being recorded.

In summary, all of these professionals bring up interesting points, but I think its clear as most of them attested to, the combinations of microphone, microphone preamp, compression and equalization are a definite part of recording a vocal and hopefully doing it in a way that is accurate and natural. Some of these producers bring up various equipment that has already been a part of our discussion and some brought up a few new options. These are all concepts worth considering and investigating as recording professionals make attempts to make the right choices in the equipment components used in the vocal chain.

Recording in a Digital Platform while Achieving an "Analog" Vocal Sound

The second question presented to the interviewees was as follows: If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound? There were varying degrees of yes, no and somewhere in between on this question from the pool of producers, including details for their reasoning.

Ed Boyer admitted in his email that, in the digital platform, sometimes things can be "tot clean." This author is assuming by "tot" he meant "totally." He says that because of this he finds ways to "dirty up" the sound a little, but "not necessarily to make it sound like tape or pushed tubes." He said he likes to "give it some grit" even if it is with amps or multiple compressors just to "drive the sound a little bit more."

Bob Clark replied with "No, I do not." He thinks that the platforms we use now to record in the modern age are "wonderful". David Hall said, "by nature of what we are doing, it is 'analog'."

Matt Pierson responded to this question by saying that he didn't think there was any difference. He said, "Sound is sound, and you're documenting something that is alive." Pierson recounts that when he used to record to tape, he would have to account for changes in the sound. He would make adjustments in the EQ or even the microphone choice depending on what the tape would do to the sound. In todays' modern studio, he says, "when recording digitally, you get almost exactly the actual sound, so it's more honest than analog." However, he admits that there are elements of the voice that are usually identified with an "analog" sound. He said things like "warmth, a breath that doesn't have too much edge, sibilance that isn't brittle, etc" are things that he will always focus his attention on. He adds that it's not as much about being "analog" as it is about it being "live."

Tom Reeves expresses that he is usually trying to "make the vocal 'pop' out of the mix." Because of this, he said he is rarely trying to warm things up any further, especially if he has already most likely used a tube microphone and mostly tube vocal chain while recording. Reeves shares that recording a vocal on two inch tape will always be a thing of beauty to him, but in a dense Pro Tools session it might not always be the sound he needs to capture. He discusses plug-ins that he goes to later in the mix to make the vocal "stand up" a little more, but admits that none of those mixing tools should "lessen the importance of choosing the right mic and preamp for a particular singer and

song." He also adds that jazz trio, orchestral or rock rhythm sections all require different lead vocal sounds.

Dave Sperandio replied to this question by stating, "Yes. I might add harmonics, tube saturation, tape saturation, "color" compression, or various forms of "desirable" distortion to warm/smooth the sound." In direct contrast, Paul Tavenner said, "No, not at all." He elaborates by saying that he has never been a fan of analog tape recording mostly because he learned his "recording skills using tape and was constantly frustrated with the problems analog brings." He feels that the current digital platforms have completely overcome their own shortcomings and even surpassed the sound quality of analog recording.

Paul Wickliffe reveals that he generally waits to add analog processing in mixdown on the stereo buss and goes on to explain the components he uses for that process in his interview transcript (see Appendix C). Paul recorded in analog for twenty-five years before switching over to digital. Because of this, he says he knows how the analog tape would normally behave and can use his digital plug-ins to simulate the same analog effect.

Nathan Zwald adds another thought and idea to the pool of responses for this question. He says, "It partially depends on the quality of the converters and clocking of the digital system." He also adds that "Sometimes the analog harmonics help to compensate for deficiencies in the digital conversion process and would be used more frequently if that were the case." In his usual vocal chain, he uses a Texas Instruments PCM4222 converter chip and feels that the sound of it is exemplary. He says that in a converter of this kind "harmonics and overtones that are often blurred by lesser

converters are pure and audible." He goes on to say that with a good converter, he will then use analog simulators for tape saturation or a slight circuit overdrive to add harmonics to a vocal in a subtle way.

Zwald likes to keep the raw audio recorded pure so that he can get back to it if needed. He makes an interesting point that you can always add coloration later, but you can't remove it from the vocal once it has been recorded. This author tends to agree with this idea even amongst much discussion concerning color with microphones and microphone pre-amplifiers. In this authors opinion, the goal should be to use the different combinations of color to capture the most natural sounding vocal as possible.

In summary, this writer found that most producer/engineers are quite happy with the digital platform that is being used, but still are mindful of attempting to incorporate elements of the vintage analog sound. It seems they all do this with a variety of methods, from tube mics, preamps and compressors to plug-ins later in the process. It seems that the further we get away from the days of analog and the more advances technology makes, we will hear less and less wishful thinking about the days of the "analog sound."

Approaches to Working with a Singer

The third question that the interviewees were asked was the following: "As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?"

Ed Boyer replied that everyone is different. He thinks some people need coddling, while others need you to stay out of the way. He says he tries to stay as honest as possible without getting harsh or disingenuous. Bob Clark feels that with singers much of your job is to be a psychologist. He thinks that most artists are insecure, no

matter how famous. He adds that they need reassurance and assistance to make the recording experience calm and positive. Clark believes that you learn each artists' needs and likes, then you use what you learn to get a great performance recorded. He says that each artist is different and what they need is different, but using a kind, positive and supportive approach always produces "magic."

David Hall feels as though the most important thing in recording a vocal is to make sure that the singer is able to be intimate and confident with the delivery of the story line. He says he does this by keeping the control room as quiet as possible. He requires interns or assistants in the control room to stay off their phones, no talking and paying attention to the process. He doesn't want the singer to be distracted in any way and adds that the singer should be able to look into the control room and see positive vibes from the people who are participating.

Matt Pierson feels as though everyone is different and every single performance is different. He thinks the producer has to be an effective combination of "therapist and director." He says he develops a level of trust, puts the artist in a musical and physical environment where they feel comfortable to open up emotionally, then directs them as much as needed. Pierson says there are many instances where he has seen what it takes to get an artist to reach a level of "vulnerability and frank honesty." Another interesting point that Pierson offers, is that it helps the singer to love what they sound like in their headphones. He thinks "there is an addictive quality to hearing yourself sound great."

Tom Reeves explains that he has "gone full circle with singers from punching every word or phrase 200 times to letting them sing through a song 3 times and then wood-shedding a 4th take." He encourages plenty of "out of the studio" rehearsal time

for the artist. With this approach, when its time to record, the artists seem to be ready to do their best. As this author does, Reeves then comps or assembles the best final vocal from those performances. He makes the point that "most singers voices change rapidly over the course of an hour." He thinks that most singers have about 4 good takes in them and past that it gets counter productive. After that happens, he feels that not only are they tired, they have most likely lost the "moment" from over-thinking. Reeves says that he tries to be ready to record at the very beginning because the best vocal moments usually happen quickly and then they are gone. He does admit though that occasionally it can take a while to get a certain phrase musically correct. He thinks the problem with producers sometimes come when the producer is not a singer themselves. Reeves says that in this situation the producer has a hard time relating to what the singer is going through. He admits that at one time, he was one of those guys, but has learned the hard way. He says that usually the producer that jumps "right in with a lot of suggestions regarding pitch and needs for things to be his/her way" is usually not the producer that will get the best results from a singer. Reeves says that he'll take passion over perfection in a performance every time, especially when he knows he has the technology to clean things up a little in the editing world.

Dave Sperandio agrees with several others that every singer is different, but says that some call for a firm hand and others a delicate touch. He goes on to elaborate that some "respond best to encouragement, others to humor, others still need anger or shame to produce their best results." These are all techniques that might work for him, but this author in particular does not agree with the use of anger or shame to produce the desired

performance. There seem to be an assortment of more positive or constructively critical approaches at one's disposal.

Paul Tavenner believes like several others that it is important for the producer to be empathetic during the recording process and offers a few examples. First, he believes that if the singer is having tuning problems then as this author has said earlier in this paper, maybe they are not hearing themselves correctly in the headphones or cannot hear enough of a harmonic instrument. Another example is that of using compression. Tavenner feels that if a producer is using too much compression, then a singer cannot be as expressive as they want to be and will unknowingly fight the compressor effect. According to this authors experience, most seasoned engineers will only use a slight amount of compression in recording, if any at all. Tavenner makes a good point that both parties on either side of the glass need to be on the same page; additionally, it is good for the producer to understand the artistic vision and goals of the performer. According to him, this is probably the most important aspect of getting a good recorded performance. He also suggest to offer useful feedback and asked questions as to how the artist is doing. Making sure the artist is comfortable and is a big part of the evaluation process is also definitely important. Tavenner also points out that often this can be done by playing back what has just been recorded and then allowing the vocalist to make an assessment of what they are hearing. One last thought that he offers is that "helping the vocalist feel empowered when he/she is behind the microphone will work wonders in getting the best performance."

Paul Wickliffe agrees with the others that every situation is different depending on the singer's level of experience and what exactly they are singing. He says what they have in common is that "they must be at ease and able to render the lyric story in an authentic and emotionally compelling way." Additionally, he feels that Pitch, diction and placement can all be worked out, but a technically flawless performance is not worth much if the singer is not making some kind of personal connection with the audience. Wickliffe is of the opinion that singers should not "self-edit" while performing in front of the microphone. Instead, they should listen to the previous recorded playback and make a judgment as to what works and what does not work. He feels that story telling is the essence of all lasting entertainment and if you don't believe what you are singing then "pick another song or sing about floor wax with a big smile for big money."

Wickliffe has learned from having lived with a great singer for thirty years that if the singer is having pitch problems, one should ask for the mix to be adjusted or either take a headphone off in order to hear in the room acoustically. He has learned that if the headphones are too loud or the vocal is not loud enough, the singer will tend to push vocally and sing too loud. If the vocal is too loud, the vocal will often sing tentatively and have pitch issues, according to Wyckliffe. He also makes the point that what works live doesn't always work in the studio. In the studio, there is no need to be heard "in the second balcony or over the Margarita blender," you are singing right into someone's ear about six inches away from a microphone. He also makes the analogy in the following statement: "It's like the difference between acting on stage and acting in film. The big gestures that work on stage look over the top in a close-up on a screen 20 feet high."

Nathan Zwald feels that getting the singer very comfortable, mix and the vibe of

the room will go a long way. He suggests that singing in the studio is 90% mental. With this in mind, he proposes that laughter, a relaxed atmosphere and a calm demeanor are good for fostering creativity. He believes that singers can feel very exposed standing behind the glass, and because of this, need to feel they are in a non-judgmental place while trying to maintain there standards of excellence and deliver the best vocal performance possible. He goes on to say that if they can trust that you will get the best out of them, then they will feel freer "to stretch and go for those special performance moments that make a great vocal recording."

In conclusion, this writer is not convinced that all singers are as insecure as some might think. If they are insecure, it could be from either a lack of experience or perhaps a bad experience. It would, however, be optimum for singers to approach a recording session with the desire to be as prepared as possible. From the other side of the glass, it would also be a novel idea for engineers and producers to remember that the voice is different from a trumpet, saxophone or drums. The voice is housed in the human body and by nature is going to be more personal and sensitive to environmental factors in the recording studio. Producers and engineers alike need to make an attempt to create a warm and inviting atmosphere than lends to the highest level of comfort for the singer with the end goal being to foster creativity.

Microphone Set-up for Group Vocals

The fourth question that was posed in the email interview was the following: What is your favorite microphone set-up for group vocals? Ed Boyer gave the short answer that he likes everyone circled up around an omni microphone. Although this is a valid approach, this writer would like to point out that very little editing can happen in

the vocals later on when recording a group with this technique. One might suggest using this technique in addition to individual microphones for each singer.

Bob Clark begins with the point that you work with whatever equipment is available; however, his favorite microphone choices for group vocals are "Tele 251, U-47, U-67, C-12." He makes a note that if the higher end microphones are not available then a U87, or AT 4060, 4050, 4033 or other microphones in that range will be great for a group. For group vocals, he prefers using the API microphone pre-amplifiers. David Hall weighs in that for soprano he would use a Tele 251 or AKG C-12, for alto and tenor he would use U-67's and for bass either the U47 or M-49. Matt Pierson says that he normally uses a couple of Neumann U-67s; and in some cases, perhaps a "C-24 stereo tube mic or a AEA R-88 stereo ribbon mic to capture a group/room sound."

Tom Reeves says that he defaults to a semi-circle setting with four microphones for SATB recording. He says that he has the best mix results when he has blend control having each part on a separate microphone. He says a good indicator of it being right is when the singers relax and settle into what they are hearing.

Dave Sperandio says that he doesn't typically record vocals in a group. One could make the assumption with this statement and knowing his work that when he does record a group he might tend to do it on individual microphones, treated like individual vocals.

Paul Tavenner says that if it is a modern a cappella group, he likes to put them all on separate microphones isolated for more control later in the mix. For multiple background vocals, Tavenner says that he often puts several singers on one microphone, double and triple tracked. He brings up a concept of making vocals sound more spacious

and realistic by having different singers stand varying distances from the mic on different recorded passes. For background vocals that are harmonies to a lead vocal, Tavenner adds that it is very important to have each singer on a separate microphone for better control in the mix. Putting different harmonies on the same microphone would make it almost impossible get the desired balance later in the mixing stage.

Paul Wickliffe suggests that it depends on the application. If he is recording Take 6, he believes it is crucial for everyone to get their own microphone as "the balance of every chord is critical to the arrangement and the engineer needs individual control to pull that off." For backing vocals on a pop song, Wickliffe agrees that "two or three singers around one mic then triple tracking each part is fairly standard." If he were recording a classical choir, he says that he would have them stand in an arc around an X-Y stereo pair of microphones. The balance of sound is then done by physically moving the singers closer to or farther away from the microphones.

For SATB, Nathan Zwald would use a separate microphone for each section and maybe an additional room microphone as some of the others have stated. For group or what he calls "gang" vocals, he states that a stereo microphone like the "Neumann SM 69, AKG C24 or a stereo set of DPA 3532s" would be a good choice.

Zwald thinks it is best for the vocalists to have eye contact with each other and the conductor or producer. He feels, as this writer does, that a slight arc is usually a good way to go about it. He adds that he tries to keep an ear on the blend and the position of certain voices in the stereo spectrum as vocal takes (tracks) begin to stack up. For recording choir, Zwald believes that a good blend can be achieved with a "stereo XY mic setup coupled with spaced pair AB omni mics." His particular placement of these

microphones would depend on the sound of the room and a DPA would be his "go-to" microphone for this application.

In summary, recording group vocals presents several challenges for the audio engineer. Having a sight line, being able to hear adequately in a group setting, finding the best vocal chain combinations for the different voice-types, set-up configuration and whether or not to use a single or additional stereo microphone are all considerations that have to be made. Again, whatever allows the group of singers to be the most comfortable and the most freedom to be creative throughout the recording, is the best configuration.

Vocal Editing and Vocal Tuning: Compromising Musical Integrity or Extension of Singers' Performance

The fifth question presented to the interviewee's was the following: How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance? Ed Boyer feels that sometimes it can be done discretely or selectively and have virtually no real degrading effect on the vocal. But, when it comes to mainstream, he believes it is an "aesthetic unto itself." He goes on to say that whether it degrades the signal or not, it is "the sound" at the moment.

Bob Clark does not feel that it compromises the integrity, but are just tools to make the recording better. He posed a question back to the author that was stated somewhat like this: Are editing and tuning any different than using a compressor or an equalizer? His argument is that both of those help improve how a singer sounds and sings. Clark gives an example, "We used to record 10 tracks of a solo, comp it down to one track and then sing another 10 and comp those." He goes on to say that then they would punch in hundreds of times to get the best pitch and timing. He feels those

instances are no different from the editing going on today. He thinks it is much easier on a singer now to get a great performance. Bob says he's worked with singers that don't need to be tuned. In the same way, he says all drummers don't need to be quantized. But in today's studio, if the need is there, it is possible.

David Hall says that he will do 6-7 recorded passes with a singer, and if he feels like he has got it, he will dismiss the artist from the room and "comp" or compile the best of each pass. Hall will then put a light chromatic tuner on the comp pass, put an effect on it and bring the artist back in to listen and make any corrections. As far as tuning goes, he feels we have "let the cat out of the bag." Hall suggests that depending on the genre, the listener is now use to hearing the vocal perfectly tuned. He admits to not liking it, personally, but that the market dictates it.

Matt Pierson does all of his own vocal comping/editing and Melodyne work and finds it to be an essential element of vocal production. Pierson makes the statement that "a live performance and a recording are two very different art forms." He feels that problems arise when the tools that are available fall into the wrong hands and are abused. He says that even with some of the incredible singers he has worked with, you would see a lot of edits and crossfades on the initial vocal tracks. Additionally, he says that he has used Auto Tune or Melodyne (exclusively over the past two years) on every vocal he has produced in the last five years. Of course, he says he uses it on some singers more than others. Pierson says that you have to know where to stop and you can never let the process remove the emotional impact of the performance. The skill is in trying to be sure that every single thing you do allows the deepest emotional impact. So, with this in mind, he thinks that removing distractions from the track is key. His basic rule is that if

his ear is drawn to it in a negative way, then he needs to fix it in order to allow the positive to shine.

Tom Reeves admits to being a "huge fan of vocal editing/tuning as long as it's a spice and not the main ingredient." He says that unless the song calls for tuning or editing as an effect, he would never globally tune a vocal. He uses "melodyne to make an already solid performance into a keeper by todays' standards." Specifically, he describes that he might be carefully affecting pitch, drift, or vibrato on any particular note. He believes that any one of those things can improve or even save a vocal phrase. As this author believes, the technology is vast but the real issue comes down to making the correct musical decision while doing it.

On the flipside, Reeves reveals that it can all be a bit subjective as well. He says that you often have to ask yourself: Is the editing making the performance better or has it taken the "soul" out of the performance by trying to achieve perfection? Many times Reeves feels that the answer is somewhere in between the two extremes. Reeves found the musical integrity part of the question to be "almost laughable." He explained that he gets "paid to make things come out the way a client 'wants' it to sound, not how it 'actually' sounds." Furthermore, he says that most artists want it to sound "perfect" so that is what he tries to give them, perfection that sounds like it was sung that way. He believes that the musical integrity question is something that we need to get over, that editing and tuning a vocal in today's marketplace is just part of the "gig."

Dave Sperandio describes vocal editing and tuning as no different from make-up and camera angles. He believes that if the tools are available, they will be used; and

furthermore, they will alter the sound of music. Paul Tavenner explains that he is more than willing to utilize vocal editing and tuning, but the bottom line is to get a vocal track with the right energy and emotion, a performance that moves the listener. He feels that editing and tuning are merely tools to improve and clean up an existing track, not manufacture energy.

Paul Tavenner typically only tunes the worst offending notes, if any. He points out that our ears are use to hearing certain flaws that all human performance contains. His thoughts are that if you correct too much it will sterilize the track and kill the energy which this author agrees with wholeheartedly. He feels that "tuning a 'money note' without changing its' overall inflection is totally acceptable." Tavenner feels that vocal tuning and editing is not so much an extension of the performance as it is an extension of the artists' concept. He also feels that "reaching a high skill level as a performer trumps all the editing and tuning techniques available." He believes that this is most true with jazz and other more traditional musical forms. Additionally he adds, "for dance or rave tracks who cares - if the groove moves the listener, or dancer, then it's acceptable."

Paul Wickliffe is against vocal tuning and vocal editing if it is what he refers to as "photoshop for pop music." His opinion is that when it is overused, it makes anyone sound artificial and just like everyone else. He believes that it "takes the humanity right out of any performance." Wickliffe seems to think that singers who recorded prior to the use of Autotune are more easily identifiable by people that are musically aware than all the artist that are recording with it in todays' market. He sees the use of AutoTune as a trend or a fad and "hopes that it fades as quickly as plaid and polyester bell-bottoms." He

rarely uses AutoTune because he doesn't like the way it sounds, so he usually uses Waves Tune instead. He considers Waves Tune as similar to Melodyne, but thinks the editing is quicker in ProTools and more transparent. As this author agrees, he feels that if one can tell that something has been fixed, then he didn't do his job.

Wickliffe explains that he is in a service business, so it is his job to make the client happy and sound good. He admits that pitch correction and vocal editing are good for business, but bad for music if used as a crutch or a cheat. Wickliffe makes a good point that fifty years ago, you were recorded or played on the radio because you were an exceptional talent; but these days, studio deception has made it possible for artist to be signed based on looks instead of talent and virtuosity. He goes on to say that "Autotune and MTV have allowed people with little ability to make a fortune while genuine talents are ignored." With this in mind, Wickliffe believes that part of that problem is that the opportunities for artists to play live venues and fine tune their craft are few and far between.

Nathan Zwald has a middle of the road opinion about vocal tuning and editing. He feels that poor vocal tuning, poor application and overuse of editing techniques can certainly compromise musical integrity. However, he states that "like any recording technique, properly applied it is simply a tool in the tool box to create great music." He believes like Hall that modern-day audiences almost expect now to hear an "in tune" vocal whether it sounds "tune-y" or naturally correct. He goes on to explain that "they often hear an 'out of tune' performance as sounding less than polished even though they may not understand why they feel that way."

In conclusion, this question, as no surprise, produced polarized responses. There are some people who are adamantly against vocal editing and especially vocal tuning. There are others who are adamantly against those who oppose it. This author feels strongly that the "perfect world" maybe lies somewhere in the middleground. As musicians, we should, of course, try to perfect our craft to the level of not needing to be edited and/or tuned. However, in the world and music industry we exist in today, it is a reality and like it or not, as some of the producers pointed out, almost expected by the public. I agree that the cat has been let out of the bag somewhat, but maybe our musicianship can find a way to co-exist with the technologies that are available, and often helpful, without compromising the musical integrity of our art.

Recording Vocal Percussion

The sixth question that was put before the panel of producers was "What is your approach to recording vocal percussion?" Ed Boyer said that as long as the microphone is not distorting with wind, then he can work with it in a studio setting. Matt Pierson said he has not recorded it much, but was not really a big fan. He says that if an artist has that particular skill in his or her bag of tricks, he makes sure to only use it sparingly. As far as recording, he said he would use the same set-up that was used for the lead vocal.

Tom Reeves believes that microphone placement and selection is more important than normal because of the enormous "plosives" and other artifacts that one deals with when recording vocal effects. He explains that tube compression with a medium fast attack/medium slow release and a mic pre with extremely fast transient response like the API 512 might be helpful. In some cases, like this author, he has found that a dynamic mic is a better choice for vocal percussion.

Dave Sperandio has an interesting idea for recording vocal percussion. He said he usually "records 2-4 live tracks," quantize those and then supplement the live kit with vocal percussion samples. This author is assuming by "live kit" he is referring to a live kit of vocal percussion. He then mixes the two to taste, whatever the artist and particular song calls for, either more organic or more produced.

For Paul Tavenner, he says it depends on the style, but normally uses a Shure SM 58 with a lot of processing (EQ, compression, etc.) to get the desired results. Like Reeves and Tavenner, Paul Wickliffe also likes to use a dynamic microphone like a Shure SM58 to record vocal percussion.

Nathan Zwald suggests to minimize the artifacts such as the plosives and sibilance in the recording process. He also adds the idea that often the audio quantization in a DAW can help in creating layered vocal percussion performances.

In summary, very little has been written about techniques for recording vocal percussion up to this point in time. As we attempt to open the conversation and draw conclusions between the usage of condenser as oppose to dynamic microphones, may we remember that the answer might again lie somewhere in the middle ground. The idea of incorporating a live recorded track of vocal percussion or "beat boxing" with samples layered on top seems to be a very interesting concept that might be the best use of both techniques. In doing so, may the end product guide our ears to exploring innovative approaches to recording percussive sounds that can be generated by the human voice.

Future Advancements in Vocal Recording

The seventh question that was put forward in the email interview was the following: Do you have any thoughts about the future of vocal recording and any advancements that might take place?

Ed Boyer believes that people will always sing and always enjoy recording their music, so it will be around in some form or another. He said he was sure that some form of technology would come along to change things, but he is not sure what that might be as of yet. Bob Clark thinks that the tools we use continue to get better as the electronics get better. He believes that "the quest for the perfect recording of the perfect performance will always continue."

David Hall makes an interesting point that he learned his craft by watching and observing. He feels that the further we get away from that structure, we start to introduce bad practices such as distortion, sibilance that could have been minimized by proper mic selection or finding the right mic pre that matches the mic and voice. He adds that better A to D converters and clocks make recording in the digital domain sound dramatically better.

Matt Pierson does not see any changes taking place that could be seen as significant improvements. He thinks the biggest concern is producers/engineers compromising based on budget limitations and/or ignorance. So he feels that it's more an issue of maintaining the current level of excellence. He thinks Pro Tools and Melodyne are incredibly refined and user-friendly at this point. There are new versions of tube and ribbon microphones being made every day, but Pierson says he has not heard anything yet that can beat out the favored vintage mics. He thinks advancement could take place

with the music itself, and with "an artist's ability to continue to document their art honestly while refining it through technology."

Tom Reeves seems to think that we will continue to see advancements in tools that will make it easier to improve or edit vocals. He says that more cost efficient microphones have been popular lately, but he sees a shift back to the more high-end choices. He thinks that young producers are trying to deal with their greatly reduced budgets. He says that they may not even own or use a real studio but have acquired racks filled with at least two channels of the best recording chains possible. He would consider this behavior in young producers an advancement.

Dave Sperandio hopes that we return to a more organic sound and have less tolerance for poor singers, but doubts that it will happen <smile>. Paul Tavenner believes that smaller audiophile quality headset microphones will eventually be a part of vocal recording with the proliferation of micro and nano technologies. He says that vocal recording now consists of a bulky large diaphragm microphone with a shock mount that is mounted on a stationary microphone stand. He thinks this limits the expressiveness of many vocalists.

The author, having sung on that type microphone for many years, has to disagree with that opinion. It is only a matter of the singer getting use to standing in front of a microphone and learning how to deliver a good performance. That is the kind of delivery that takes years to perfect. Tavenner says he is "not aware of any small, lightweight headset microphones that have a sound quality worthy of recording for an album." This author is not sure this is possible with the currently used condenser microphones by

virtue of how a larger diaphragm is more responsive to the lower frequencies in the voice as we have discussed in this study.

The smaller diaphragms automatically are used more for high frequency response situations. Therefore, this author is not convinced that the concept will constitute an advancement. The Crown headset microphones are possibly representative of the highest quality to date, though in this engineer's estimation, not truly comparable in quality to that of a large diaphragm condenser microphone.

Paul Wickliffe is concerned about "technology being used as a crutch or a substitute for study and virtuosity." He says that because of the software deception, mediocre marketing and the streaming paradigm, he fears for the "survival of the music business writ large." As others have stated, and this author concurs, technology and software are just tools that can be used to different extremes. To even have access to those tools present-day are advancements in our field, but that access can be misused in a way that deconstructs the artform. For additional comments on recording by Wickliffe unrelated to the topic of this paper, see Appendix C.

Nathan Zwald puts forward some interesting ideas about advancements that might be around the corner. He says that the next generation of digital converters will finally bring digital recording to a "level of maturity." He says that when these are coupled with the analog harmonic simulators, in both hardware and plug-in form, we will finally see the end to true analog recording mediums. He thinks at this juncture, "Digital will finally sound good." Zwald believes that one day we will see the death of mp3s. He says this hopefully will be a return back to high fidelity song distribution formats and perhaps will reprogram the publics' ears to demand a higher quality. He suggests that "the increases

in internet bandwidth coupled with high density flash storage will keep the convenience of MP3s but allow for the larger file sizes necessary to hear what's really there." He goes on to make the following statement which this author totally agrees with: "Just like our eyes now can't live without HD television, I hope our ears are next in line for an upgrade."

Additional Comments

The final question that was posed to the pool of interviewees was the broad inquiry of "is there anything else you would like to add?" This question generated a broad array of responses but nevertheless some stories and answers that are worth hearing. Bob Clark started his response by saying "a great singer is a great singer is a great singer." He said that as an engineer he does his best to capture all the artist gives, but knows that he must not let technology get in the way. Clark says that once upon a time he was recording a fantastic artist in RCA Studio A in Nashville. He says that she was in the vocal booth, with 55 musicians on the floor and they cut wonderful arrangements of three songs in three hours. The vocals on the album are the vocals that she sang on that particular session. He says there was "no punching, no sliding into the 'pocket,' no tuning necessary." The tools were available, but were not needed. Clark says to that the trick is to not use the tools just because they are available. If something needs to be fixed, it can be fixed, but he says to not use them just because you can. On the other hand, he adds that "if all those tools create and define your artist, then use them."

David Hall tells a story about a time when he had an opportunity to record Ronnie Milsap. He says that he and the producer tried to be mindful of keeping the talkback mic

open at all times so that Ronnie could hear and be a part of their dialogue. He says that they forgot to open the talkback a few times and Ronnie eventually said "I feel a million miles away from you guys." He recounts another session when he was recording vocals with an artist and there was a party going on in the back of the control room. He says that he had his attention on the artist, but somehow got pulled into the party atmosphere. The client told him later that it was very distracting to sing with the visual clutter that was going on. He made the correction and told his interns from then on to stay visually quiet, hide behind speakers, or find some way to stay out of the view of the artist. Hall says he wants to create a safe environment where the artist can be vulnerable and "naked."

Matt Pierson feels that because "the human voice is clearly the greatest communicator" it is essential when recording the voice to document something deeply emotional, showing a strong connection between performer and material. He believes that "it is important to see to it that through the technical recording process you are not only getting the raw material, but you have to have something to work with emotionally." He also thinks that the artistic vision, the connection the artist has with the material, how prepared/in sync the players are with each other, and the recording environment (setup, room temperature, food/beverages, personnel, studio locale) are all important factors to consider.

Tom Reeves hopes that the young generation of producer/engineers will continue to have a dedication to the balance between using proven, high-end recording tools and the newest digital technology available. He believes that a piece of gear being "old" or "new" does not determine its value and suggests that we should embrace both "while striving for excellence in our craft."

Paul Wickliffe offered some ideas about mixing the vocal that will not be discussed here per the scope of our discussion being focused on recording only (see Appendix C). Wickliffe makes the suggestion to never go into the studio without a rehearsal. He believes one should "work out your keys and tempos before you arrive." He also suggests that an artist should "book some low profile gigs to work out the material and take a pocket recorder with you to listen back." Wickliffe says to never cut tracks without the vocal present and suggests that if the vocal is not there then the players will unconsciously fill in the space. He also has advice for artists not to psych themselves out with the so-called "red light syndrome" but to relax, meditate, "be yourself" and "don't question your instincts while recording." He suggests to save "questioning your instincts" for the moment when you are listening to the playback. Wickliffe warns the artist to not "self-produce" but have another set of ears to be both advocate and critic in the control room. He encourages singers to be dynamically consistent with the band "by either directing them at the session or following them in the overdub" so that the engineer does not have to "ride" the fader. He suggests that "if you have to hit a note hard for effect or to hit the pitch, stand so you can comfortably rock back from the mic." He thinks this technique is preferable to compression, although he doesn't think it should sound like "you temporarily walked into an open manhole." This author disagrees somewhat with this technique since, in her experience, a very good compressor (like a Tube Tech CL1B) set on a very light setting of 2:1, with a slow attack, slow release and a slight (or no) decrease in threshold will deliver a better result than stepping away from the microphone. Obviously, as Wickliffe alluded to, the danger of

stepping away is the vocal suddenly taking on a "roomy" sound and loosing the presence that was there at a certain predetermined distance.

Wickliffe suggests that "to taper a decrescendo without the note becoming unstable, slowly turn your head from the mic." This author has reservations with this as well since, in her experience with this technique, the presence of the vocal sound will be diminished. When the voice is not singing directly into the diaphragm, this technique is known as being "off-axis" and is known to dramatically affect the vocal sound. Usually trained singers don't have an issue with this since they can diminish the sound at will with the use of good breath support.

Wickliffe suggests to singers that "if you need a big breath, turn your head." This author takes issue with this suggestion per this paper's discussion as we have been considering ways to make the performer more comfortable in order to capture the most natural sounding vocal. With the editing techniques and automation available in DAW's, a breath that sounds too big is merely a keystroke of diminishing the volume automation.

Wickliffe makes the suggestion for singers to record themselves outside of the studio. He says there is "no excuse not to record rehearsals, even vocal warm-ups and karaoke." He thinks that then you should listen to yourself and see what is or isn't working. He thinks "critiquing yourself as you sing will only inhibit you" but to be in the moment, listening objectively after the take and learning from it, is the best approach.

This author tends to think this approach could differ depending on the singer. Many singers, (the author being a case in point), who have done background vocals for many years, are use to critiquing themselves on microphone and being able to change

some things “on the fly” so to speak. In this authors’ experience, this is only possible though if the singer can hear themselves perfectly on the headset, just as if they are just hearing themselves sing in the room. This author achieves this level of comfort by halfway offsetting both sides of the headset. Ultimately, both of these approaches are probably simply a matter of how comfortable a singer can become, regardless of the recording situation.

CHAPTER 14

DEMONSTRATION RECORDING

This chapter is an explanation of the demonstration recording and description of the four major voice-types (Soprano, Alto, Tenor and Bass) recorded on different microphones prior to the subsequent steps of the vocal chain, including microphone pre-amplifier and compression. It also includes a list of recorded samples of several different vocal chain combinations and examples of editing/tuning techniques.

Sample Recording of a Solo Phrase in a Jazz Style

The following organization was applied to the order of the demonstration recording for a solo phrase in a jazz style:

SOUNDFILE PREFIX	VOCAL CHAIN COMPONENTS
1, 5, 9, 13	Soprano-Mic-Preamp-Compression
2, 6, 10, 14	Alto-Mic-Preamp-Compression
3, 7, 11, 15	Tenor-Mic-Preamp-Compression
4, 8, 12, 16	Bass-Mic-Preamp-Compression
1-4	Utilizing Pre-amplifier, API 512c
5-8	Utilizing Pre-amplifier, Shadow Hills Mono GAMA
9-12	Utilizing Pre-amplifier, Martech MSS-10.
13-16	Utilizing Pre-amplifier, Neve 1272.

Table 14.1

A melodic line/couplet was written in a jazz style and a lead sheet was given to each singer before each session began. They were all given a few minutes to learn the phrase prior to the session. The alto and tenor were given precisely the same notes, in the

Key of C, by design in order to perhaps uncover the dichotomy between vocal chain combination choices for the male and female voice. The bass was given the same notes, in the Key of C, as the tenor but down the octave. The soprano notes were transposed up a fourth to the Key of F. These keys were chosen to place the phrases right in the heart of the range where the voice-types tend to sing more frequently. The singers were told to sing the phrase freely in a jazz style and were given the starting pitch before each sample was recorded. The singers that were used are as follows: Soprano-Arianna Neikrug, Alto-Kelly Garner, Tenor-Drew Dahan, Bass-Aaron Hicks, and Vocal Percussion-Carter Soso. Each singer recorded 32 samples with four condenser microphones and four microphone pre-amplifiers with compression both on and off. The recording with each singer took about an hour to complete. The vocal percussionist recorded 20 samples.

More specifically, each sampling of the individual microphone and microphone pre combinations (i.e. 1.1a) is also followed by a soundfile that includes the Tube-Tech CL-1B on the end of the chain (i.e. 1.1b).

Sample Recording of Vocal Percussion

The following organization was applied to the order of the demonstration recording for vocal percussion:

SOUNDFILE PREFIX	VOCAL PERCUSSION SOUNDS
17	Kick (i.e. bass drum)
18	Snare
19	Hat/Cymbal
20	Oscillating or Pitched
21	Full Kit

Table 14.2

Soundfiles with the prefix "17-21" were recorded consistently with the Tube Tech CL-1B compressor and two of the microphone pre-amplifiers used previously, the API 512c and the Neve 1272. These were notated as "a" and "b" respectively. The compressor was used consistently, as that is a typical use for drums. The two microphone pre-amplifiers were used because both have historically been known to be a good choice for both drums and vocals for different reasons. The API 512c is clean and flat, while the Neve is colored and delivers warmth.

Sample Recording of a Six-Voice Group

Soundfile 22.1 is a sample of a six-voice group, *Extensions*, from the University of Miami singing Jennifer Barnes' chart of "I've Got No Strings," which was used to demonstrate various editing and tuning techniques for group vocals. The first soundfile (22.1) is a raw, untouched and unedited recording of the vocals isolated without the presence of the instrumental track. Several inconsistencies in tuning are most apparent. This is more easily perceived with the sample consisting of vocal waveforms only. The second soundfile (22.2) is of the same recorded vocals (22.1) still isolated without the presence of the instrumental track, but after they had all been individually tuned with *Melodyne* by this author as instructed to do so by the group's director. The third sample (22.3) is the tuned vocals finally mixed into the recorded instrument track.

Sample Recording of a Male Improvisational Solo

Soundfile 23.1 is a sample of a male improvisational solo, unedited and untuned with many inconsistencies in pitch. Soundfile 23.2 is a sample of the same male improvisational solo as in example 23.1, but with vocal tuning using *Melodyne* and some

editing "by hand." Soundfile 23.3 is a sample of the same tuned vocal mixed into the recorded instrument track.

Demonstration Recording Observations

The demonstration recording proves pretty dramatically that there are definite differences in color between different microphones, microphone pre-amplifiers, and whether or not compression is applied. The four microphones that were used are the Korby 47, Korby 251, the Korby 67 and the Rode NT-1A. The Korby microphones are direct replicas to the Neumann 47, 67 and Telefunken 251 and are also tuned to these specific vintage microphones. The Rode NT-1A is an inexpensive $200 microphone that has become very popular over the last few years, primarily because of its sound.

The permutations between the mic, mic pre and compressor seem to be more dramatic when a bright microphone and bright preamp are paired together or the other extreme, dark microphone with a dark preamp. These differences also differ, however, depending on the voice-type being recorded. The main goal is to find a combination that does not sound "affected" but instead sounds "normal" or "natural" and how the voice sounds acoustically. This demonstration recording likely represents only the "tip of the iceberg" in terms of research and samples of the many different variations of vocal chain combinations that could be recorded.

The vocal percussion samples are particularly polarized from the dark warm tone colors of the Neve preamp to the bright "crisp" color of the API 512. The API 512 is most notably known for recording snare and pieces of the drum kit that need a little "snap" or "crackle." The differences between just these two components can be heard across the demonstration. The different microphones used, the Korby 67 and the

Sennheiser e935 are two very contrasting microphones. The Sennheiser e935 is a dynamic mic that has a little "bump" (i.e. increase) in the highest upper and lowest frequencies. The Korby 67 is a tube condenser. The way in which the vocal percussionist performs the oscillating sounds are very different on each of these mics partly because of the way the soundwaves are hitting the microphone diaphragm. The vocal percussionist moves the mic back and forth dramatically which is easier to do with the Sennheiser, as it is a handheld microphone. It should be noted that the Korby 67 sounds better on the high hat or cymbal sounds because the nature of a large diaphragm microphone is to pick up a much broader range of frequencies. This engineer has found that the perfect scenario seems to be a combination of these two microphones depending on the sound being sought. Recording vocal percussion has become very popular over the past few years with the rise of the a capella vocal groups. Strategies to track Vocal Percussion will continue to be tried and tested in recording this particular type of vocal performance.

These recorded samples of sung and percussive vocal with four different voice-types, four different microphones, four different microphone pre-amplifiers and compression on/off strongly suggests that there exists a broad palette of color for recording engineers, producers and vocalists that is waiting to be tapped. In recording the human voice and comparing recorded samples such as these, there is more to explore and more to be discovered.

CHAPTER 15

CONCLUSION

In conclusion, vocal recording has come along way from the days of analog tape to the digital age of where we are today. There are advances in recording technology that have occurred as recently as the last few years that have changed the way the artform will be documented for all time. It is not possible to put the "rabbit back into the hat" with all the technology advancements that have been made at this point. The technology should be seen as a tool and not a crutch, but a worthy tool if a problem or issue arises that can be fixed easily.

The more important job of finding the best way to pristinely capture the sound of the human voice is what should really be paramount in our mind and discussion. There are some conclusions that can be made post our discussion throughout this paper. Before recording a vocal, a clear understanding of the voice-type that will be recorded is essential. There are protocols to setting up the studio environment that can only add to the vocalists' comfort level as they attempt to record a good performance. Producer/engineers need to consider the components of the vocal chain combination with great care, taking into account the type of voice they will be recording, and correspondingly, remaining keenly aware of the types of microphones, microphone pre amplifiers, compressors and equalization that can be utilized. The producer/engineer needs to remember to set up a safe, non-judgmental, and creatively friendly atmosphere within the studio environment for the performing artist. The producer/engineer needs to be aware of how they communicate to the singer, and be mindful of the fact that all other

recording choices that are made, ultimately influence the singers' performance. Studio engineers find themselves in the early stages of the conversation on the topic of recording vocal percussion, and it is hoped that many more advancements and contributions of techniques to the field will be the result. And finally, vocal editing/tuning will probably be a polarizing topic for some time to come. Then again, so was the telephone and the television at one point in time.

It is this author's hope that, as studio recording engineers, we can put our biases aside, and, as professionals, attempt to learn from each other in an exchange of ideas as we attempt to advance the act of recording the only sole instrument housed within the human body…the singers' voice.

APPENDIX A

INITIAL EMAIL QUESTIONNAIRE QUESTIONS

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

4. What is your favorite microphone set-up for group vocals?

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

6. What is your approach to recording vocal percussion?

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

8. Is there anything else you would like to add?

APPENDIX B

INFORMED CONSENT FORM

Vocal Recording Techniques for the Modern Digital Studio
By Kelly K. Garner

INFORMED CONSENT FORM

PURPOSE:

The goal of this research is to discuss *Vocal Recording Techniques for the Modern Digital Studio*.

Responses to the questionnaire by producer/engineers or producer/artists are intended to provide insight into the various techniques utilized in the modern digital studio while recording the voice, solo or group.

PROCEDURE:

The informed consent form and questionnaire will be sent in an email (recruitment letter) to the participants. All participants are asked to voluntarily answer the questionnaire regarding *Vocal Recording Techniques for the Modern Digital Studio*.

The participants will be asked to include in their email response whether they consent to their names being published or not. Each participant acknowledges through his/her email response that he/she has read and understood the informed consent form and further agrees to its terms. The responses will be used for research and will be included in the co-investigator's doctoral essay. Through responding to the questionnaire and editing it as the participant wishes it to appear in the document, each participant also agrees that his/her responses will be published in the essay.

RISKS:

No foreseeable risks or discomfort are anticipated for you by participating. Because this research is being conducted through email, security of your correspondence cannot be guaranteed.

BENEFITS:

Although no benefits can be promised to you by participating in this study, the information gathered and distributed later is intended to help improve the awareness of *Vocal Recording Techniques for the Modern Digital Studio*.

ALTERNATIVES:

You have the alternative to not participate in this study. You may stop participating any time or you can skip any question you do not want to answer. There is no penalty incurred should you choose to halt participation.

COSTS:

No costs are anticipated for you to participate in this study.

PAYMENT TO PARTICIPATE:

No monetary payment will be awarded sue to participation in this study.

CONFIDENTIALITY:

The participants' names and responses will be made public in my dissertation, which will be submitted to the faculty of the University of Miami this Spring 2014 and will be available for educational purposes unless he/she indicates to the principle investigator that they would like their information to be kept confidential. Please state your preference in your email response on whether you want your name to be published or not.

RIGHT TO WITHDRAW:

Your participation is voluntary you have the right to withdraw from the study.

OTHER PERTINENT INFORMATION:

The researcher will answer any questions you may have regarding the study and will give you a copy of the consent form after you have signed it. If you have any questions about the study please contact Kelly K. Garner co-investigator, at 615-300-0305 or garnerk@comcast.net, or Professor Rachel Lebon, at 305-284-5813 or RLLebon@aol.com. If you have any questions about you rights as a research participant, please contact the Human Subjects Research Office (HSRO) at 305-243-3195.

Please print a copy of this consent documentation for your records.

PRINT NAME: _____

SIGNATURE: _____

APPENDIX C

Participant Questionnaire Transcripts

ED BOYER – LOS ANGELES, CALIFORNIA – INTERVIEW
VIA EMAIL – February 17, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

Depends somewhat on the context. If it's going to be beat up in post, I'm more lenient. If it needs to sound pure and vivid, it's super important. But it's always best to start off with the best quality possible, as you can take away fidelity but you can't really add it.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

Sometimes things can be tot clean, so I find subtle ways to dirty it up. Not necessarily to make it sound like tape or pushed tubes. But just to give it some grit. Amps? Multiple compressors? Anything that can drive a little bit.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

Everyone's different. Come people need coddling. Some people need you to stay out of the way. I try to stay as honest as possible without getting too harsh or too fake-nice.

4. What is your favorite microphone set-up for group vocals?

Everyone circled up around an omni.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

Sometimes it can be done discretely and selectively and have (virtually) no really degrading effects. But, when it comes to mainstream, it's really an aesthetic unto itself. Whether it degrades signal or not, it's "the sound" at the moment.

6. What is your approach to recording vocal percussion?

I'm not too picky. As long as the mic isn't distorting with wind, I can work with it in a studio setting.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

Hmmm. You probably want something profound here? Ha. Um...I'm sure *some* technology will come along to change things, but who knows what it'll be. But people will always sing and people will always enjoy recording their music, so it'll be around din some form or another.

8. Is there anything else you would like to add?

Not that I can think of.

BOB CLARK – NASHVILLE, TENNESSEE – INTERVIEW
VIA EMAIL – February 9, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

Of course it is important. It is always best to match your choices of gear to the artist if at all possible. There can be many things that affect that – time, location, budget. You must always do the very best you can with what you have available at the time.

One of my favorites is the ELAM-251 into an API pre with an LA-2. Just gentle compression on the recording.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

No, I do not. I think how we record now is wonderful. So a good mic and pre and most important a good voice (artist) is usually a good experience and sound. The source is always the deciding factor. The most expensive mic and pre cannot heal a bad singer.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

With singers (artists) much of your job description is that of a psychologist. Most artists, no matter how famous, are somewhat insecure. They need re-assurance and help to make the recording experience calm and positive. You learn each artist, their needs and likes, and then use what you learn to get a great performance recorded. Each one is

different and what is needed is different, but kind and positive and supportive always produce "magic".

4. What is your favorite microphone set-up for group vocals?

Once again I must say that what you have available is what you work with. Tele 251, U-47, U-67, C-12 are wonderful. However, you might not find those mics everywhere, so you might end up with an 87, or AT 4060 or 4050 or 4033, or other mics in that style and price range. All of these can be great for a group.

I am a big fan of the API pres. So groups through an API pre are nice, but you might only have the console pres. And that is fine too.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

I don't feel it compromises the integrity. Editing and tuning are tools we use to make it better. Are they different than a compressor or equalizer. Both of those help how a singer sounds and sings. We used to record 10 tracks of a solo, comp it down to one track and then sing another 10 and comp those. And punch in 100's of times to get the best pitch and timing. That was no different. It is a lot easier on the singer now to get that great performance. Not all singers need to be tuned. I have worked with those. Not all drummers need their tracks to be quantized. I work with that kind of a drummer all the time. The difference is that today, if it needs that kind of fixing, you can do it. Before, you just did another 3 takes of the song. It just might not be perfectly on a grid.

6. What is your approach to recording vocal percussion?

I have not had to record any, so I have no answers here.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

The tools we use continue to get better as our electronics get better. The quest for that perfect recording of the perfect performance will always continue.

8. Is there anything else you would like to add?

A great singer is a great singer is a great singer. We do our best as engineers to capture all that the artist is giving us. We must not let technology get in the way.
I recorded a fantastic artist. We were at RCA A in Nashville. She was in the vocal booth. There were 55 musicians on the floor. Wonderful arrangements. We did 3 songs in three hours. The vocals on the album are those session vocals she sang. No punching, no sliding into the "pocket", no tuning necessary. The tools were there, but not needed. The trick is to NOT use them just because they are there. If it needs it, we can fix it. But you don't have to just because you can. If all those tools create and define your artist, then use them.

DAVID HALL – NASHVILLE, TENNESSEE – INTERVIEW
VIA EMAIL – January 22, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

I think that the vocal is the most important thing in the production. It is the delivery method for the story, that the singer/songwriter is trying to convey.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

That is an interesting question… By nature of what we are doing, it is "analog". I almost always start with several different flavors of mics and find what mic matches the voice. I do this with a clean signal chain. Hopefully making decisions based on the flavor of the mic.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

My most important thing that I do when recording vocals is, make sure that the singer is able to be intimate and confident with the delivery of the story-line… I do this by keeping the control room as "quiet" as possible. If you are in the control room and you are my assistant or intern, you won't be on your iphone, you will be paying eyes forward and no talking. No distractions. I want the singer to be able to look into the control room and see positive vibes from the people that are participating in his/her heart.

4. What is your favorite microphone set-up for group vocals?

If it is choral "SATB", I will use 251 or C-12 for soprano, U67's for alto and tenors, the bass gets either a U47 or an M-49

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

I personally like "blue-notes" When I do vocals, I will always keep the 1st pass. The artist is not under the pressure that we are "in the red". I will let the artist do 2 or 3 passes with minimum involvement. I will make notes as we go. Noting any big issues. I will then start adding my opinion and honing any issues that need to be addressed. Hopefully, I do 6-7 passes. If I feel like I have everything I need, I will dismiss the artist and entourage from the control room and comp the best of each pass. This generally takes about an hour to do. I will put a light chromatic tuner on the comp pass, put an effect on it and bring the artist back in to listen and make any corrections. As far as tuning, we have let the cat out of the bag. Depending on the genre, the listener is use to it. I don't personally like it but the market dictates it.

6. What is your approach to recording vocal percussion?

N/A

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

All I know is that I learned my craft by watching and observing. I learned from people that learned from people that… As we move further away from that structure, we start to introduce bad practices such as distortion, sibilance that could've been minimized

by proper mic selection or finding the right preamp that matches the voice AND mic. I believe that better A to D converters and clocks make recording in the digital domain way better sounding.

8. Is there anything else you would like to add?

I love to record vocals. I take this process very serious. Years ago, I had the opportunity to record Ronnie Milsap. The producer and I were diligent in keeping the talkback open when we were not in recording. We forgot to open the talkback a few times and Ronnie eventually said, "I feel a million miles away from you guys"… Valuable lesson #1 Another time, I was recording vocals with an artist and there was a party going on in the back of the control room, I had my attention on the artist and yet, I got pulled into the commotion of the party atmosphere. After the session, the client told me that it was very distracting to sing with the visual clutter that was going on. The next day, we corrected this. I tell my interns and any new assistant's that are with me, to either stay visually quiet or hide behind the speakers or stay out of the view of the artist. I want to create a safe environment that the artist can be vulnerable and "naked".

MATT PIERSON – NEW YORK, NEW YORK – INTERVIEW
VIA EMAIL – February 3, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

Essential. If you don't document the human voice in an honest and detailed way, you won't be working with something that contains all of the most important emotional elements of the artist's interpretation going forward. In terms of my favorite choices, I usually try a few microphone/pre-amp combinations and pick what will work best for that particular artist. However, I usually work at a studio called Sear Sound in NYC, and I inevitably end up choosing a Neumann M-49 through the Neve 1081 console mic pre. Sometimes I'll use an Avalon M3 mic pre. For male voices, I usually end up going with a Neumann U-47 "chrome top" through the Neve. There is also a rare Pultec mic pre that we use on occasion. The other two microphones I like for voices are the Telefunken C12 and 251, but when I compare them to the Neumanns, they usually lose out.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

I don't believe there is any difference. Sound is sound, and you're documenting something that is alive. When I used to record to tape, I needed to account for the changes in sound. We could record to tape, listen back, and then in some cases adjust the EQ or even the mic choice based on what the tape would do to the sound. However, when recording digitally, you get almost exactly the actual sound, so it's more honest than analog. However, there are elements of the voice that are usually identified with an "analog" sound that I will always focus on – warmth, a breath that doesn't have too much

of an edge, sibilance that isn't brittle, etc. So it's not about "analog" as much as it's about "live."

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

Everyone is different, and every single performance is different. The producer is many things, and being an effective combination of therapist and director is essential. I usually just develop a level of trust with an artist, and put them in a musical and physical environment where they are comfortable to open up emotionally, and then direct them as needed. There are many, many stories I could tell (but won't!) about what it takes to get some artists to reach a level of vulnerability and frank honesty. Again, it mainly has to do with preparation, trust, and environment. It also helps if they love what they sound like in their headphones – there is an addictive quality to hearing yourself sound great.

4. What is your favorite microphone set-up for group vocals?

We usually use a couple of Neumann U-67s. In some cases, we may put up a C-24 stereo tube mic or a AEA R-88 stereo ribbon mic to capture a group/room sound.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

I do all of my own vocal comping/editing and Melodyne work. I find it to be an essential element of vocal production. The recording studio is a very different environment than the stage, and a live performance and a recording are two very different art forms. The problems arise when the tools available to us are in the wrong hands and

are abused. If you were to look at the initial vocal tracks I've worked on, even with incredible singers, you would see a LOT of edits and crossfades. In addition, I have used either AutoTune or, exclusively over the past couple of years Melodyne, on every vocal performance I have produced over the past 5 years – certainly some more than others. Being a trumpet player, I've always been very attentive to pitch, and my attention to detail is very set, thanks in particular to working with Whit Sidener as a member of the UM Concert Jazz Band, and as time goes on, I'm more and more exacting. It's very important to know where to stop – you can never allow the process to remove any of the emotional impact of the performance. The skill is in to be sure that every single thing you do allows the deepest emotional impact, so removing distraction is key. But I defy anyone to listen to the recordings and point out where they hear any editing or tuning. My basic rule is, if my ear is drawn to something in a negative way, I fix it to allow the positive to shine.

6. What is your approach to recording vocal percussion?

I haven't done it much, and am generally not a big fan – I work very regularly with Bashiri Johnson, one of the greatest percussionists alive. If an artist has that particular skill in his/her bag of tricks, I've made sure to use it sparingly. In terms of recording, I would take the same as any vocal performance. If they use one setup for the lead, the same setup should be used for backgrounds, vocal percussion, scatting, rapping, etc. Same instrument, same mic I would say.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

I don't really see any changes taking place that could be seen as a significant improvement. I think a bigger concern is the fact that many producers/engineers are compromising based on budget limitations and/or ignorance. So it's more an issue of retaining the level of excellence that can currently be attained. Pro Tools and Melodyne are incredibly refined and user-friendly at this point. There are new versions of tube and ribbon microphones being made every day, but I haven't heard anything yet that can beat out my favorite vintage mics. I think the advancement can take place with the music itself, and an artist's ability to continue to document their art honestly while refining it through technology.

8. Is there anything else you would like to add?

The human voice is clearly the greatest communicator, and when recording, it's essential to document something deeply emotional, showing a strong connection between the performer and the material. It is important to see to it that through the technical recording process you are not only getting the raw material, but you have to have something to work with emotionally. The artistic vision, the connection the artist has with the material, how prepared and in synch the players are, the recording environment (setup, temperature, food & drink, who is in attendence, studio location, etc) are all important factors to consider.

TOM REEVES – NASHVILLE, TENNESSEE – INTERVIEW
VIA EMAIL – January 19, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

I started recording long before "digital workstations" so the importance of the recording chain is still of vital importance to me....although I'm always shocked to hear what many "hits" have been recorded on and through. For me, microphone, preamp, EQ, and compressor combinations is where vocal magic occurs even if I already know what plug- ins I'm turning to for a mix. I love an AKG C-12 through an API 512 or 312, then compressed with a Tube-Tech CL1a or LA2a. You can't go wrong on most vocal applications with that combination. If you don't have a $10,000 C-12 laying around then, depending on the vocalist, I would look for an Neumann 87, AKG 414, Telefunken U47 and any number of other choices. In the final analysis though, plug-in treatment can also play a major role in my final vocal sound. I guess I ride the fence...one foot in the analog vocal chain and the other foot inside the world of workstations and plug-ins. Both sides of the chain are extremely important and relevant by todays standards. I try to use the best of what each side brings to the table.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

I'm usually trying to make the vocal "pop" out of the mix so I'm rarely trying to

warm things up any further if I've already used a tube mic or mostly tube vocal chain when recording. For me a vocal recorded on 2" tape is and always will be a thing of beauty, however, in a dense Pro Tools mix that might not always be the sound I'm needing. The first plug-in I turn to is currently an SSL 4000 channel for minor EQ and additional compression. I might use a Waves Ren Vox plugin after the SSL just to make the vocal "stand up" a bit more. The SSL EQ section will cure just about any issue you have with the sound of a lead vocal after you're hearing everything in a complex mix. I never let any of this lessen the importance of choosing the right mic and preamp for a particular singer and song. Jazz trio, full orchestra or rocking rhythm section.....those situations require different lead vocal sounds.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

I've gone full circle with singers from punching every word or phrase 200 times to now letting them sing through a song 3 times and then wood-shedding a 4th take. I encourageartists to spend plenty of rehearsal time "out of the studio" so when we do record, they're ready to give it their best shot. After all that, I may likely "comp" or assemble the final vocal from those performances. Here's why, most singers voices change rapidly over the course of an hour. I really want to get 3 takes that sound the same "physically". Most great singers have about 4 good takes in them and then it gets counter-productive. After that, they're not only tired but have likely lost the "moment" and are thinking about things way to much. I try to be in "record" right away because the

best singer moments happen quickly and then they're gone. Now, this is not to say that occasionally it may take a while to get a certain phrase musically correct. The problem with most producers is that they're not singers themselves so they really can't relate to what a singer is going through. I'm a good example of that...but I've learned...the hard way. The producer that starts right in with a lot of suggestions regarding pitch and needs for things to be his/her way is, in most cases, NOT the producer that gets the best performance out of a singer. I handle the situation almost the same for the "pro" studio singers. They get tired too. I'll take "passion" in a vocal performance every time over perfection, particularly these days when I know that I have the technology to clean things up a little.

4. What is your favorite microphone set-up for group vocals?

I pretty much default to a semi-circle and 4 mics for SATB recording. I have the best mix results when I have blend control so I tend to choose separate mics for each stand or part (SATB). I can tell when it's right as the singers will relax and settle in to what they're hearing.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

I'm a huge fan of vocal editing/tuning as long as it's a spice and not the main ingredient. Unless the song calls for tuning/editing as an effect, I would never globally tune a vocal performance. I use Melodyne to make an already solid performance into a keeper by todays standards. I might be carefully affecting pitch, drift, or vibrato on any given

note. Any one of those things can improve or save a vocal phrase. The technology is incredible but the real issue comes down to making the correct musical decision while doing it. It can all be a bit subjective too. You have to ask yourself....is what I've done really making this "better"? Have I taken the "soul" out of this performance by choosing perfection? Sometimes the answer will surprise you. Many times the answer is somewhere in between the two extremes.

As far as musical integrity goes.....the question is almost laughable. I get paid to make things come out they way a client "wants" it to sound, not how it "actually" sounds. Further, understand that the expectation in the marketplace is different these days. Most

folks want it "perfect" so that's what we give them...perfection that sounds like it was done that way to start with. That's just the "gig"....get over the musical integrity question. You'll just be disappointed if you hang your hat on that one. I've chosen to die on a different battlefield.

6. What is your approach to recording vocal percussion?

Mic placement and selection is more important than normal as there are enormous "plosives" and other artifacts to deal with when recording vocal effects. Tube compression with a medium fast attack and medium slow release is helpful. Also a mic pre with extremely fast transient response like the API 512 is helpful. In some cases, I've found that a dynamic mic is a better choice for vocal percussion.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

I think we will continue to see the advancement of tools that will make it easier to improve/edit vocal performances. More cost efficient microphones have been the rage in the past few years but I'm seeing a shift back to more high-end choices these days. I think the younger producers are trying to get it right with their greatly reduced budgets. They might not own or even use a real studio but their racks are filled with at least two channels of the very best recording chain possible. I would consider that behavior an "advancement".

8. Is there anything else you would like to add?

As a professional community I think we're in a pretty good place right now as I feel like we're still using and teaching the balance between proven, high end recording tools and the newest digital technology available to us. I'm hoping the young generation of producer/engineers will continue to be dedicated to that balance. The fact that a piece of technology is "old" or "new" does not alone determine it's value to us. Let's continue to embrace both the old and the new while striving for excellence in our craft.

DAVE SPERANDIO – LOS ANGELES, CALIFORNIA – INTERVIEW
VIA EMAIL – February 8, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

The most important factor is the talent of the performer. The vocal chain is also quite important, especially in achieving the last "30%" or so of "excellence." One can obtain a good – even very good – recording with fairly inexpensive equipment. In some cases, an ultra hi-fi sound is not desired, or needed.

My go-to vocal chain is a Telefunken AR51 mic through a Vintech 273 Preamp/EQ. I typically do not compress on the way in, but compress later using software emulations of hardware compressors.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

Yes. I might add harmonics, tube saturation, tape saturation, "color" compression, or various forms of "desirable" distortion to warm/smooth the sound.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

Each singer is different. Some call for a firm hand, some need a delicate touch. Some respond best to encouragement, others to humor, others still need anger or shame to produce their best results. Or bourbon.

4. What is your favorite microphone set-up for group vocals?

I don't typically record vocals in a group.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

It is no different than makeup and camera angles. If the tools are available, they will be used. And their use will alter the sound of music.

6. What is your approach to recording vocal percussion?

Record 2-4 live tracks, quantize these, and supplement this live kit with vocal percussion samples. Mix the two to taste, as the artist and particular song calls for (more organic or more produced).

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

I hope that we will return to a more organic sound, and have less tolerance for poor singers. But I doubt that will happen. ☺

8. Is there anything else you would like to add?

PAUL TAVENNER – LOS ANGELES, CALIFORNIA – INTERVIEW
VIA EMAIL – February 5, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound?

The microphone, mic cable, preamp, and other components are all, to varying degrees, important links in the vocal chain. However, I find proper mic placement and performance technique to be more important. For example, some singers read their lyrics during recordings. When they look down to read the lyrics the mic no longer picks up the optimal presence of the voice and the level may change. The tone change cannot fully be recovered by equalization - the optimal tone is simply lost.

Many engineers and producers discuss matching certain microphones with certain preamps. In my experience, I have found that microphones have more to do with the ultimate quality, or characteristic, of the vocal sound. Most professional mic preamps will transform the mic signal to a proper line level without adding noise and distortion. Professional microphones, on the other hand, can have completely different sound qualities when compared to each other and offer more in the way of ultimate vocal sound.

What are your favorite Vocal Chain Combinations?

I typically use a Neumann U87 microphone coupled with a Universal Audio 610 mic preamp. I often use a modified AKG 414BULS. I have also used Neumann U67 and U47 mics. On rare occasions, to get a gritty sound, I have had excellent results with a Shure SM58. We live in an era of great microphone choices - both vintage and contemporary as well as pricey and inexpensive - that all sound great. I often times set up several microphones all at once for a test recording and let the vocalist pick the one

they like. The results are often times very surprising. I always throw in an SM58 to keep everyone honest.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

No, not at all. I have never been a huge fan of analog tape recording. I learned my recording skills using tape and was constantly frustrated with the problems analog brings. Current digital technology, in my opinion, has completely overcome its own original shortcomings and surpassed the sound quality of analog.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

It's important for a producer to be empathetic to the singer during the recording process. That means understanding both how it feels to be under scrutiny during the recoding process and the actual recording process. Here are a few examples:

If a singer is having tuning problems, it's possible he/she is not hearing enough of themselves in the phones or too much (and not enough bass or other instrument as a pitch reference) and to help them adjust. An inexperienced engineer/producer may inform the singer of pitch issues without offering any help or guidance which can bring the creative process to a halt.

Another example is using compression while recording. If the engineer uses too much compression then the singer can never be fully expressive and will unknowingly

"fight" the compression effect. Some singers work well with compression while others work best without.

So, understanding the technical aspect of the recording process is one of the most important and useful ways to help a singer get the most out of the recorded performance.

Understanding the singer's artistic vision and goal is probably the most important aspect of getting a good recorded performance. The folks on each side of the glass must be on the same page. Once this is established, the rest is easy. The rest consists of communicating effectively throughout the recording - offering useful feedback, asking questions (are you hearing everything you need to hear in the phones? Are you comfortable?), offering encouragement and making sure the vocalist is part of the evaluation process. One way to do that is to play back the take and ask the vocalist what his/her opinion is.

In general, helping the vocalist feel empowered when he/she is behind the microphone will work wonders in getting the best performance.

I can give more detailed explanations of the above, if necessary.

4. What is your favorite microphone set-up for group vocals?

If it's a modern a cappella group, I like to put each vocalist on their own mic and preferably isolate them for mixing control later.

For multiple background vocals, I often times put several singers on one mic then double or triple track them. One trick to create a more "spacious" and realistic group vocal sound is to have different singers stand a varying distances from the mic on different passes.

For background vocals that are harmonies to a lead vocal, it's very important to have each singer on a separate mic for better mixing control later. Once a mix begins to take shape, the balances will most likely need to be adjusted. Putting harmonies on the same mic will make it almost impossible get the desired balance later.

5. How do you feel about vocal editing and vocal tuning?

I am more than willing to utilize vocal editing and tuning. The bottom line for me is getting a vocal track with energy and emotion. It must move the listener. Editing and tuning are simply tools to improve an existing track. Again, the "performance" must move the listener. Simply tuning and editing won't do much for it other than clean it up if the energy isn't there.

Tuning is a tricky process. I typically tune the worst offending notes, if any. The process becomes tricky when our ears become too "in-tune" with the "flaws" that all human performance contains. In other words, if one digs one will find flaws to correct. I think this sterilizes a track and kills its energy. Tuning a "money note" without changing it's overall inflection is totally acceptable.

In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

I don't think it's an extension of the performance so much as it is an extension of the artist's concept. I think of recordings like paintings. To me, there are no rules except that the listener must be moved by the performance. I don't care how that performance is achieved as long as I'm thinking of the recording as a piece of art.

I'm always in favor of performance excellence. I think reaching a high skill level as a performer trumps all the editing and tuning techniques available. This is perhaps

most true with jazz and other more traditional musical forms. For dance or rave tracks who cares - if the groove moves the listener, or dancer, then it's acceptable.

6. What is your approach to recording vocal percussion?

It depends on the style. A Shure SM58 and lots processing (EQ, compression, etc) are key to getting desired results.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

Smaller, audiophile quality headset microphones will undoubtedly be part of the future of vocal recording especially with the proliferation of micro and nano technologies. Current vocal recording tends to consist of using a bulky large diaphragm microphone and shock mount that is mounted on a stationary mic stand. This limits the expressiveness of many vocalists. To date, I am not aware of any small, lightweight headset microphones that have a sound quality worthy of recording for an album.

8. Is there anything else you would like to add?

Let me know if there is anything you'd like more clarification on.

PAUL WICKLIFFE – NEW YORK, NEW YORK – INTERVIEW
VIA EMAIL – January 20, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

By vocal chain combinations, I'm assuming you mean what gear do I use. My default set-up is a Neumann u47 (Sinatra / Beatle mic), a Hardy M-1 or API 512 mic pre-amp, a Massenburg GML 8200 EQ and a DBX 160X or UREI LA-4 compressor. I also use a Neumann u87, M49 or vintage ribbons depending on the singer. I normally record the vocal flat except for a rumble filter and use as little compression as necessary. I use a Stedman pop filter with an additional layer of panty hose for the plosives. I place the pop filter an inch from the mic and the singer's mouth 6 to 8 inches from the pop filter. If you relax your hand, put your thumb to your lip and your pinky should touch the pop filter. Good mic placement is as critical as which mic you use. I don't usually EQ the voice until the final mix, unless there is an obvious issue.

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

I generally add the analog processing in mixdown on the whole stereo buss. I have an analog buss that runs out of ProTools into some '60s transformers, an SSL or Neve stereo buss compressor and even a tape machine if requested. Having recorded with analog for over 25 years before going all digital, I know how it behaves and can get digital plug ins to simulate the analog effect.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

Every situation is different depending on the singer's level of experience and what they are singing. What they all have in common is that they must be at ease and able to render the lyric story in an authentic and emotionally compelling way. Pitch, diction and placement can all be worked out, but a technically perfect performance isn't worth much if the singer is not making personal connection with the audience. Singers should not "self edit" while performing, but should listen to playbacks of what they just did before doing another pass to study what works and what doesn't. Remember, story telling is the essence of all lasting entertainment and if you don't believe what you are singing, pick another song or sing about floor wax with a big smile for big money. As to the technical issues, I've had the good fortune of living with a great singer for thirty years and can offer pretty good advice on the technical vocal issues. In the event, you are having pitch problems, ask to have the mix adjusted or use one headphone on and one off so you can hear your self in the room. If the headphones are too loud or you are not mixed up enough, you will tend to push and sing too loud. If you are mixed too loud in the headphones, you will either pull back or have pitch issues against the band. It's also important to remember that what works live doesn't always work in the studio. In the studio, you are not singing to be heard in the second balcony or over the Margarita blender, you are singing right into someone's ear at six inches from the mic. It's like the difference between acting on stage and acting in film. The big gestures that work on stage look over the top in a close-up on a screen 20 feet high.

4. What is your favorite microphone set-up for group vocals?

Again, this depends on the application. If I'm recording Take Six, everyone gets their own mic as the balance of every chord is critical to the arrangement and the engineer needs individual control to pull that off. If it is backing vocals on a pop song, two or three singers around one mic then triple tracking each part is fairly standard. A classical choir would stand in an arc around an X-Y stereo pair of mics and the balance would be done by physically moving the singers closer or farther from the mic.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

If you mean Photoshop for pop music, if I hear it, I hate it. When overused, it makes anyone sound artificial and just like everyone else. It takes the humanity right out of any performance. Play any famous singer before Autotune and most musically aware people can name the artist. It's much more difficult now. I hope the fad will reach the cliché status of disco and remain as popular as plaid, polyester bell-bottoms. I realize millions of people like the taste of a Big Mac, but it's full of unnatural ingredients, always tastes predictably the same, and if eaten everyday will make you sick. I rarely use Autotune as I don't like how it sounds. I use Waves Tune. It is similar to Melodyne, but for me the editing is quicker in the ProTools and more transparent. I feel like if you can tell I that fixed the pitch, I'm not doing my job right. The bottom line is that I'm in a service business and I'm here to make the client happy and sound good. Pitch correction and vocal editing is good for my cash flow. That being said, it's bad for music in general if it is used as a crutch or a cheat. 50 years ago, if you weren't an exceptional talent, you

likely didn't get recorded or played on the radio. Now, studio deception has made it possible for singers to be signed based on their looks instead of their talent and virtuosity. Autotune and MTV have allowed people with little ability to make a fortune while genuine talents are ignored. By the same token, the opportunities for artists to play live venues to fine tune their craft are few and far between.

6. What is your approach to recording vocal percussion?

I use a dynamic mic like a Shure SM58.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

As previously mentioned, I'm concerned about technology being used as a crutch or a substitute for study and virtuosity. Between software deception, mediocrity marketing, and the streaming paradigm, I fear for the survival of the music business writ large.

8. Is there anything else you would like to add?

1. It's important to remember the entire picture when mixing the recording. You will want to feel the band as much as you hear yourself. Try nesting the vocal into the mix and turn down the control room playback volume until it is barely audible. If you can still understand the lyrics of the vocal above the band you have it right. If a few words disappear, make notes and make a volume adjustment for those phrases.
2. If the arrangement is too busy or playing counter lines in the same range as your melody, turning up the vocal isn't the cure. Thin out the arrangement. Counterpoint is supposed to enhance and renew the melody, not step on it.

3. NEVER go into the studio without a rehearsal. Work out your keys and tempos before you arrive. Better yet, book some low profile gigs to work out the material and take a pocket recorder with you to listen back.
4. NEVER cut your tracks without the vocal present. Players will unconsciously fill the space if the vocal is missing.
5. Don't psych yourself out with "red light syndrome." Relax, meditate, don't tense up, be yourself, let go, and don't question your instincts while recording. Save that for the playback and DO LISTEN BACK.
6. Don't self produce. You need another set of ears to be both advocate and critic in the control room.
7. Try to be dynamically consistent with the band by either directing them at the session or following them in the overdub. The engineer should not have to "ride" the vocal if you are properly controlling your dynamics. If you have to hit a note hard for effect or to hit the pitch, stand so you can comfortably rock back from the mic. This is sonically preferable to compression, though you shouldn't sound like you temporarily walked into an open manhole.
8. To taper a decrescendo without the note becoming unstable, slowly turn your head from the mic.
9. If you need a big breath, turn your head.
10. Record yourself outside the studio. Pocket digital recorders can be had for under $100 so there is no excuse not to record rehearsals, even vocal warm-ups and karaoke. Then listen to yourself to see what is and isn't working regarding your pitch, placement, diction, pocket, the whole nine yards.

Critiquing yourself AS you sing will only inhibit you. Be in the moment, then listen to the moment objectively and learn from it.

NATHAN ZWALD – NASHVILLE, TN – INTERVIEW
VIA EMAIL – January 18, 2014

1. How important do you consider the Vocal Chain Combination to the quality of the final recorded vocal sound? What are your favorite Vocal Chain Combinations?

Critical, with a focus on fidelity and natural capture of the performance.

Brauner VM1-KHE (microphone) > Gordon Model 5 (preamp) > GML 8200 (EQ) > Tube-Tech CL1B or Retro Instruments 176 (compressor) > UBK Fatso or Anamod ATS-1 (analog tape saturation) > PCM4222 (A/D converter). If possible the digital recording medium clocked to an atomic clock like the Antelope 10M.

Sometimes I will substitute different microphones - Telefunken 251 (vintage), Neumann 269 (vintage), AKG C24 (vintage) but 90% of the time the KHE wins.

It drives me a little crazy when professional gear actually subtracts from the sound. When using the Gordon Audio preamps, for example, when your ear gets used to them and then you switch to a lesser but still "pro" preamp you hear information being subtracted. I want to decide what gets added or subtracted. Concerning a vocal mic chain George Massengerg is quoted as saying his goal is "...to keep a signal as clean and transparent as possible for as long as possible." and I agree.

Also, matching microphone to preamp and knowing how the sound of each will interact and how they interact electrically, specifically as it relates to resistance, is important. This relationship effects the "EQ" of the microphone's response and ultimately the sound of the vocal being recorded.

Finally... junk in junk out. There is a lot you can compensate for in the mixing and editing process but there is a saying in the south that seems apropos and goes something like "putting lipstick on a pig".

2. If you record in a digital platform, do you find yourself trying to make the vocal sound more "analog?" And if so, what steps do you take in trying to achieve more of that "analog" sound?

It partially depends on the quality of the converters and clocking of the digital system. Sometimes the analog harmonics help to compensate for deficiencies in the digital conversion process and would be used more frequently if that were the case. In the vocal chain listed above I use the Texas Instruments PCM4222 converter chip. The sound of this converter is exemplary in my opinion. Harmonics and overtones that are often blurred by lesser converters are pure and audible. With a great converter I will use analog simulators for tape saturation or slight circuit overdrive to add harmonics to a vocal on a case by case basis with my ear as the final arbiter and when used are usually very subtle. In this day and age of great sounding plugins that can be added after the fact I like to keep the raw audio being recorded pure. I always want to be able to "get back to beautiful". You can always add coloration but you can't remove it from the original recording short of a time machine.

3. As a producer, what is your approach to working with a singer? Are there certain phrases you have learned to say that tend to work "magic" with the singer while recording the vocal?

I find getting a singer very comfortable with their headphones, mix, the vibe of the room, etc. goes a long way. Singing in the studio is 90% mental. Usually laughter, a relaxed atmosphere and calm demeanor foster creativity. Singers can feel very exposed standing behind the glass and need to feel they are in a safe non-judgemental place while still maintaining the highest standards for their performance. This way they trust you will

get the very best from them yet they feel free to stretch and go for those special performance moments that make a great vocal recording.

4. What is your favorite microphone set-up for group vocals?

I'm assuming group vocals is not SATB choral style vocals. For SATB I'd use separate mics for each section and maybe a room mic. For group/gang vocals I like a good stereo mic like the Neumann SM69, AKG C24 or a stereo set of DPA 3532s.

I like the vocalists to have eye contact with each other and a potential conductor or producer. A slight arc is usually a good way to go about it. I like to keep an ear on the blend and the position of certain voices in the stereo spectrum as we stack vocal takes.

For a choir with good blend I love a stereo XY mic setup coupled with spaced pair AB omni mics. Placement would depend on the room's sound. DPA would be my microphone choice for this application.

5. How do you feel about vocal editing and vocal tuning? In today's marketplace, do you feel it compromises the recordings' musical integrity or is it now a common modern-day extension of the singers' performance?

Poor vocal tuning, poor application and overuse of these techniques can certainly compromise musical integrity. Like any recording technique, properly applied it is simply a tool in the tool box to create great music. Modern listening audiences almost expect to hear an in tune vocal whether it sounds "tune-y" or simply naturally correct. They often hear an out of tune performance as sounding less than polished even though they may not understand why they feel that way.

6. What is your approach to recording vocal percussion?

Minimize artifacts such as plosives and sibilance in the recording process. Often audio quantization in a DAW can help in creating layered vocal percussion performances.

7. Do you have any thoughts about the future of vocal recording and any advancements that might take place?

The next generation, not yet on the market, of digital converters will finally bring digital recording to a level of maturity where when coupled with the analog harmonic simulators available, in both hardware and plugin form, we will see the slow and final end of true analog recording mediums. Digital will finally sound good.

Someday we will see the death of MP3s. It couldn't come soon enough. A return to a high fidelity song distribution format will hopefully reprogram the public's ears to demand it. The increases in internet bandwidth coupled with high density flash storage will keep the convenience of MP3s but allow for the larger file sizes necessary to hear what's really there. Just like our eyes now can't live without HD television I hope our ears are next in line for an upgrade.

8. Is there anything else you would like to add?

APPENDIX D

EMAIL QUESTIONNAIRE PARTICIPANT BIOGRAPHIES

ED BOYER – LOS ANGELES, CALIFORNIA

Ed Boyer makes his home in Los Angeles, CA and is known for his work on Pitch Perfect (2012), Glee (2009) and The Sing-Off (2009). He arranged/produced/performed vocals for Universal's Pitch Perfect with Deke Sharon and Ben Bram and has mixed vocals for all three seasons of NBC's The Sing Off. He mixed Pentatonix's debut EP (produced by Ben Bram), which topped iTunes' pop chart and debuted at #14 on the Billboard 200. Ed mixed tracks for The Backbeats' debut album (Sony/Arrival Records) and arranged/co-recorded a total of 12 songs for Bubs/Warblers on Glee. The Glee singles (9 of which charted on the Billboard Hot 100) collectively sold over a million copies. In 2011, his clients received 42 C.A.R.A. nominations and 22 awards, including first-place awards for Best Pop/Rock Album, Best Male Collegiate Album, Best Female Collegiate Album, and Best C.A.L. Album. He mixed and co-produced (with Deke Sharon) Committed's debut album for Sony/Epic Records. Ed rehearsed contestants and mixed audio for The Sing Off's Macy's Thanksgiving Day Parade performance. He also mixed and co-produced (with Deke Sharon) The Sing-Off's "Harmonies for the Holidays" and "The Best of Season 2″ albums for Sony/Epic Records. Micky Rapkin's book "Pitch Perfect: The Quest For Collegiate A Cappella Glory" documents Ed's recording of Pandaemonium as well as many other a cappella stories.

BOB CLARK – NASHVILLE, TENNESSEE

Bob Clark, recording engineer for thirty eight years, makes his home in Nashville, Tennessee. Prior to his arrival in Nashville, Bob studied film and television production at San Diego State University in San Diego, California. During his time in Nashville, Bob worked as chief engineer at the Benson Company for twelve years with producers Greg Nelson, Phil Johnson, Don Marsh, Lari Goss, and David Clydesdale. During his career he has received eight gold and two platinum records, working with artists Sandi Patty, Dallas Holm, Dino, Travis Tritt, Ricky Skaggs, Shelby Lynne, Faith Hill and the Nashville Symphony.

DAVID HALL – NASHVILLE, TENNESSEE

David Hall began his music career in Nashville in 1988 as an assistant engineer to Doug Sarrett. Soon after, he became a staff engineer at Quad Studios in Nashville and later became a freelance assistant engineer, working primarily in Nashville's major studios. His first big break came when he had the opportunity to work with Quincy Jones and Mervyn Warren on "Handel's Messiah: A Soulful Celebration" as an assistant engineer. He then had the opportunity to assist under influential Pop/Rock engineers, Csaba Petocz, Joe Chiccarelli, and Mick Gauzaki. His career project credits now include Frank Sinatra, Petra, Etta James, Kenny Chesney, NewFound Road, Fall Out Boy, Vince Gill, Jewel, and Carrie Underwood. Award credits include a GRAMMY® Award for Petra for Best Rock Gospel Album in 2000, a Latin GRAMMY® Award for Ricardo Arjona for Best Male Pop Vocal in 2006, as well as several nominations. David has also worked on numerous gold and multi-platinum selling projects.

MATT PIERSON – NEW YORK, NEW YORK

Matt Pierson has established himself as a true leader in the Jazz and Adult Music world with over twenty years of diversified music industry experience, both as a record producer and executive. In particular, through his work in the studio with artists ranging from instrumentalists Joshua Redman, Kirk Whalum, and Brad Mehldau to vocalists such as Sophie Milman, Jane Monheit, K.D. Lang, and more recent breakthrough artists like Becca Stevens and Nicole Henry, Pierson continues to put his creative stamp on some of the most important and successful projects in contemporary music.

TOM REEVES – NASHVILLE, TENNESSEE

Tom Reeves is a Nashville based musician, producer, recording engineer, and studio owner under the company name of Westpark Creative Group, Inc. Westpark Creative Group provides recording services for the local Nashville marketplace and also specializes in music production for themed entertainment companies associated with cruise ships, theme parks, corporate event planners, casinos, and other entertainment venues. Over a 33 year career, Tom has toured extensively as well as produced and/or engineered award winning projects for Word Music, Benson Company, Brentwood Music, Celebrity Cruises, Royal Carribean, Dollywood Entertainment, Opryland Entertainment, Disney Entertainment, The Ryman, Oceania Cruises, Norwegian Cruise Lines, Regents Cruise Line, POET Theatricals, Gary Musick Productions, Jean Ann Ryan Productions, Marguerite Scott Entertainment and numerous corporate industrial packages for a wide variety of clients.

DAVE SPERANDIO – LOS ANGELES, CALIFORNIA

Dave Sperandio, or "dio" has been working with a cappella music for 20 years as a performer, producer, and promoter. His work has helped to define and support the contemporary a cappella world for more than a decade. Dave is the founder of the Alliance for A Cappella Initiatives, and creator of the acclaimed SoJam A Cappella Festival, as well as the 'Sing' vocal compilation CD. Since founding the a cappella production company diovoce in 2000, Dave and his clients have made more than 100 appearances on numerous "best-of" compilations, and have earned hundreds of CARA nods, both as a performer and while "behind the glass". He has been called the "Timbaland of a cappella" in the Gotham-published aca-documentary "Pitch Perfect" (now a major motion picture), and has appeared in featured interviews with the Wall Street Journal, Mouth Off!, and Acatribe. Dave has performed with numerous ensembles, directed the UNC Clef Hangers, and has toured and recorded with vocal tonic, transit and Almost Recess. His work has afforded him the opportunity to work with many artists including Ben Folds, Pentatonix, Peter Hollens, Lindsey Stirling, Brendan James, ARORA, and Street Corner Symphony. In 2006 Dave partnered with fellow producers Ed Boyer, James Cannon, James Gammon, and Tat Tong to form VocalSource, which represents the most comprehensive network of vocal music producers and studios in the world. In 2008 Dave merged the AACI with CASA, joining the CASA Board of Directors as Director of Events, managing SoJam and adding LAAF, VoCALnation, BOSS, and Acappellafest to CASA's family of events. In 2011 Dave formed Vocal Mastering: the world's only dedicated mastering facility tailored specifically to the exacting and unique needs of vocal music.

PAUL TAVENNER – LOS ANGELES, CALIFORNIA

Paul Tavenner's career as a recording engineer, producer and musician has spanned more than 25 years and over 1,200 recordings. He has held positions at Capitol Records, CBS-TV, The Post Group and Kitchen Sync Studios. He started his first business, Man Alive Music Productions, in 1989 when he was only 25 years old and specialized in CD mastering and manufacturing. He currently owns Big City Recording Studios in Granada Hills, CA, which he acquired in 1999 where he specializes in recording and producing jazz, classical, pop and other acoustic styles of music. He has worked with a wide variety of world-class artists including Ernie Watts, Stanley Clarke, Peter Erskine, Jimmy Haslip, The Yellow Jackets, Bird York (Grammy-nominated song for the movie "Crash"), Kate Reid, Josh Nelson, Sara Gazarek, William Shatner (reached #1 album spot on Billboard), Bill Cunliffe, Alan Pasqua, Jeff Richman, "Fusion For Miles: A Guitar Tribute" featuring Mike Stern, Bill Frisell, Pat Martino and others; platinum-selling ska band, Sublime; Quiet Riot, Judy Tenuta (Grammy-nominated for best comedy album), Justo Almario, Jerry Vale, Glen Campbell, Andre Previn (restorations from original recordings), The Fifth Dimension, The Playboy Jazz Festival, Lew Del Gatto (Saturday Night Live) and many others. He also has extensive experience recording music and dialog for films, TV and radio.

PAUL WICKLIFFE – NEW YORK, NEW YORK

Paul Wickliffe is the sole proprietor of Skyline Productions, Inc. a New York / New Jersey corporation providing professional audio engineering and production services for established record labels in the jazz and world music genres, i.e. Arcadia, BMG, Blue Note, Concord Jazz, Evidence, GRP, Green Linnet, JVC, Narada, N2K, Shanachie,

Telarc Jazz and Verve / Forecast. Mr. Wickliffe's experience also includes designing, building and operating seven professional recording studios. His client list includes: B-52's, Living Colour, Mariah Carey, C+C Music Factory, Eric Clapton, Miles Davis, Duran Duran, Roberta Flack, Dizzy Gillespie, Al Jarreau, Jodeci, Carol King, Queen Latifah, Melissa Manchester, Bobby McFerrin, David Liebman, John Fedchock, Betty Buckley, Ray Vega, Kevin Eubanks, Ric Ocasek, Puff Daddy, REM, Diana Ross, Richie Sambora, James Taylor, Stevie Ray Vaughan, Suzanne Vega and Paul Young. He has also provided recording services for the following producers: Clivilles and Cole, Neil Dorfsman, Kevin Killen, Arif Mardin, Hugh Padgham, Nile Rodgers and Don Was. Skyline Studios was nominated for a TEC award by the readers of Mix Magazine as top five studio in the world for five years in a row (1989 - 1993). During his first decade of credits after 1975, he recorded the gamut of pop artists from Sammy Davis, Jr. to Meatloaf, from Judy Collins to Grace Jones. Since being nominated for a Grammy Award in 1986 for engineering a GRP jazz record, he made jazz and world music his primary focus, engineering and / or producing an average of two dozen releases per year. From 1973-1975, Paul attended New York University, School of the Arts, Institute of Film and Television studying cinematography and film editing. Worked with classmates Joel Silver, Amy Heckerling and Bob Colesbury among others. From 1972-1973, he attended Ithaca College Film and Television program primarily to study screen writing with Rod Serling.

NATHAN ZWALD – NASHVILLE, TN

Nathan Zwald is a very bright “up and coming” audio engineer from Nashville, TN working for various record labels and artists. His credits include Rascal Flatts, Scotty

McCreery, Steven Curtis Chapman, Avalon, Richard Marx, Rick Elias, Gaither Vocal Band, Michael W. Smith, CeCe Winans, Kenny Rogers, Willie Nelson, Smokey Robinson and Harry Belafonte. He is also a very "in demand" tracking engineer in the Nashville area, working at studios such as Ocean Way, Blackbird, Soundstage, The Tracking Room, Sony/ATV, Masterfonics, Loud, Emerald, Quad Studios and Sound Kitchen.

APPENDIX E

RECORDED SAMPLES

Soundfile #	Voice-type	Microphone	Pre-Amp	Compressor
Soundfile 1.1a	Soprano >	Korby U-47 >	API 512c	
Soundfile 1.1b	Soprano >	Korby U-47 >	API 512c >	Tube Tech CL-1B
Soundfile 1.2a	Soprano >	Korby 251 >	API 512c	
Soundfile 1.2b	Soprano >	Korby 251 >	API 512c >	Tube Tech CL-1B
Soundfile 1.3a	Soprano >	Korby U-67 >	API 512c	
Soundfile 1.3b	Soprano >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 1.4a	Soprano >	Rode NT1-A >	API 512c	
Soundfile 1.4b	Soprano >	Rode NT1-A >	API 512c >	Tube Tech CL-1B
Soundfile 2.1a	Alto >	Korby U-47 >	API 512c	
Soundfile 2.1b	Alto >	Korby U-47 >	API 512c >	Tube Tech CL-1B
Soundfile 2.2a	Alto >	Korby 251 >	API 512c	
Soundfile 2.2b	Alto >	Korby 251 >	API 512c >	Tube Tech CL-1B
Soundfile 2.3a	Alto >	Korby U-67 >	API 512c	
Soundfile 2.3b	Alto >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 2.4a	Alto >	Rode NT1-A >	API 512c	
Soundfile 2.4b	Alto >	Rode NT1-A >	API 512c >	Tube Tech CL-1B
Soundfile 3.1a	Tenor >	Korby U-47 >	API 512c	
Soundfile 3.1b	Tenor >	Korby U-47 >	API 512c >	Tube Tech CL-1B
Soundfile 3.2a	Tenor >	Korby 251 >	API 512c	
Soundfile 3.2b	Tenor >	Korby 251 >	API 512c >	Tube Tech CL-1B
Soundfile 3.3a	Tenor >	Korby U-67 >	API 512c	

Soundfile 3.3b	Tenor >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 3.4a	Tenor >	Rode NT1-A >	API 512c	
Soundfile 3.4b	Tenor >	Rode NT1-A >	API 512c >	Tube Tech CL-1B
Soundfile 4.1a	Bass >	Korby U-47 >	API 512c	
Soundfile 4.1b	Bass >	Korby U-47 >	API 512c >	Tube Tech CL-1B
Soundfile 4.2a	Bass >	Korby 251 >	API 512c	
Soundfile 4.2b	Bass >	Korby 251 >	API 512c >	Tube Tech CL-1B
Soundfile 4.3a	Bass >	Korby U-67 >	API 512c	
Soundfile 4.3b	Bass >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 4.4a	Bass >	Rode NT1-A >	API 512c	
Soundfile 4.4b	Bass >	Rode NT1-A >	API 512c >	Tube Tech CL-1B
Soundfile 5.1a	Soprano >	Korby U-47 >	SH MG	
Soundfile 5.1b	Soprano >	Korby U-47 >	SH MG >	Tube Tech CL-1B
Soundfile 5.2a	Soprano >	Korby 251 >	SH MG	
Soundfile 5.2b	Soprano >	Korby 251 >	SH MG >	Tube Tech CL-1B
Soundfile 5.3a	Soprano >	Korby U-67 >	SH MG	
Soundfile 5.3b	Soprano >	Korby U-67 >	SH MG >	Tube Tech CL-1B
Soundfile 5.4a	Soprano >	Rode NT1-A >	SH MG	
Soundfile 5.4b	Soprano >	Rode NT1-A >	SH MG >	Tube Tech CL-1B
Soundfile 6.1a	Alto >	Korby U-47 >	SH MG	
Soundfile 6.1b	Alto >	Korby U-47 >	SH MG >	Tube Tech CL-1B
Soundfile 6.2a	Alto >	Korby 251 >	SH MG	
Soundfile 6.2b	Alto >	Korby 251 >	SH MG >	Tube Tech CL-1B
Soundfile 6.3a	Alto >	Korby U-67 >	SH MG	
Soundfile 6.3b	Alto >	Korby U-67 >	SH MG >	Tube Tech CL-1B
Soundfile 6.4a	Alto >	Rode NT1-A >	SH MG	

Soundfile 6.4b	Alto >	Rode NT1-A >	SH MG >	Tube Tech CL-1B
Soundfile 7.1a	Tenor >	Korby U-47 >	SH MG	
Soundfile 7.1b	Tenor >	Korby U-47 >	SH MG >	Tube Tech CL-1B
Soundfile 7.2a	Tenor >	Korby 251 >	SH MG	
Soundfile 7.2b	Tenor >	Korby 251 >	SH MG >	Tube Tech CL-1B
Soundfile 7.3a	Tenor >	Korby U-67 >	SH MG	
Soundfile 7.3b	Tenor >	Korby U-67 >	SH MG >	Tube Tech CL-1B
Soundfile 7.4a	Tenor >	Rode NT1-A >	SH MG	
Soundfile 7.4b	Tenor >	Rode NT1-A >	SH MG >	Tube Tech CL-1B
Soundfile 8.1a	Bass >	Korby U-47 >	SH MG	
Soundfile 8.1b	Bass >	Korby U-47 >	SH MG >	Tube Tech CL-1B
Soundfile 8.2a	Bass >	Korby 251 >	SH MG	
Soundfile 8.2b	Bass >	Korby 251 >	SH MG >	Tube Tech CL-1B
Soundfile 8.3a	Bass >	Korby U-67 >	SH MG	
Soundfile 8.3b	Bass >	Korby U-67 >	SH MG >	Tube Tech CL-1B
Soundfile 8.4a	Bass >	Rode NT1-A >	SH MG	
Soundfile 8.4b	Bass >	Rode NT1-A >	SH MG >	Tube Tech CL-1B
Soundfile 9.1a	Soprano >	Korby U-47 >	MSS-10	
Soundfile 9.1b	Soprano >	Korby U-47 >	MSS-10 >	Tube Tech CL-1B
Soundfile 9.2a	Soprano >	Korby 251 >	MSS-10	
Soundfile 9.2b	Soprano >	Korby 251 >	MSS-10 >	Tube Tech CL-1B
Soundfile 9.3a	Soprano >	Korby U-67 >	MSS-10	
Soundfile 9.3b	Soprano >	Korby U-67 >	MSS-10 >	Tube Tech CL-1B
Soundfile 9.4a	Soprano >	Rode NT1-A >	MSS-10	
Soundfile 9.4b	Soprano >	Rode NT1-A >	MSS-10 >	Tube Tech CL-1B
Soundfile 10.1a	Alto >	Korby U-47 >	MSS-10	

Soundfile 10.1b	Alto >	Korby U-47 >	MSS-10 >	Tube Tech CL-1B
Soundfile 10.2a	Alto >	Korby 251 >	MSS-10	
Soundfile 10.2b	Alto >	Korby 251 >	MSS-10 >	Tube Tech CL-1B
Soundfile 10.3a	Alto >	Korby U-67 >	MSS-10	
Soundfile 10.3b	Alto >	Korby U-67 >	MSS-10 >	Tube Tech CL-1B
Soundfile 10.4a	Alto >	Rode NT1-A >	MSS-10	
Soundfile 10.4b	Alto >	Rode NT1-A >	MSS-10 >	Tube Tech CL-1B
Soundfile 11.1a	Tenor >	Korby U-47 >	MSS-10	
Soundfile 11.1b	Tenor >	Korby U-47 >	MSS-10 >	Tube Tech CL-1B
Soundfile 11.2a	Tenor >	Korby 251 >	MSS-10	
Soundfile 11.2b	Tenor >	Korby 251 >	MSS-10 >	Tube Tech CL-1B
Soundfile 11.3a	Tenor >	Korby U-67 >	MSS-10	
Soundfile 11.3b	Tenor >	Korby U-67 >	MSS-10 >	Tube Tech CL-1B
Soundfile 11.4a	Tenor >	Rode NT1-A >	MSS-10	
Soundfile 11.4b	Tenor >	Rode NT1-A >	MSS-10 >	Tube Tech CL-1B
Soundfile 12.1a	Bass >	Korby U-47 >	MSS-10	
Soundfile 12.1b	Bass >	Korby U-47 >	MSS-10 >	Tube Tech CL-1B
Soundfile 12.2a	Bass >	Korby 251 >	MSS-10	
Soundfile 12.2b	Bass >	Korby 251 >	MSS-10 >	Tube Tech CL-1B
Soundfile 12.3a	Bass >	Korby U-67 >	MSS-10	
Soundfile 12.3b	Bass >	Korby U-67 >	MSS-10 >	Tube Tech CL-1B
Soundfile 12.4a	Bass >	Rode NT1-A >	MSS-10	
Soundfile 12.4b	Bass >	Rode NT1-A >	MSS-10 >	Tube Tech CL-1B
Soundfile 13.1a	Soprano >	Korby U-47 >	1272	
Soundfile 13.1b	Soprano >	Korby U-47 >	1272 >	Tube Tech CL-1B
Soundfile 13.2a	Soprano >	Korby 251 >	1272	

Soundfile 13.2b	Soprano >	Korby 251 >	1272 >	Tube Tech CL-1B
Soundfile 13.3a	Soprano >	Korby U-67 >	1272	
Soundfile 13.3b	Soprano >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 13.4a	Soprano >	Rode NT1-A >	1272	
Soundfile 13.4b	Soprano >	Rode NT1-A >	1272 >	Tube Tech CL-1B
Soundfile 14.1a	Alto >	Korby U-47 >	1272	
Soundfile 14.1b	Alto >	Korby U-47 >	1272 >	Tube Tech CL-1B
Soundfile 14.2a	Alto >	Korby 251 >	1272	
Soundfile 14.2b	Alto >	Korby 251 >	1272 >	Tube Tech CL-1B
Soundfile 14.3a	Alto >	Korby U-67 >	1272	
Soundfile 14.3b	Alto >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 14.4a	Alto >	Rode NT1-A >	1272	
Soundfile 14.4b	Alto >	Rode NT1-A >	1272 >	Tube Tech CL-1B
Soundfile 15.1a	Tenor >	Korby U-47 >	1272	
Soundfile 15.1b	Tenor >	Korby U-47 >	1272 >	Tube Tech CL-1B
Soundfile 15.2a	Tenor >	Korby 251 >	1272	
Soundfile 15.2b	Tenor >	Korby 251 >	1272 >	Tube Tech CL-1B
Soundfile 15.3a	Tenor >	Korby U-67 >	1272	
Soundfile 15.3b	Tenor >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 15.4a	Tenor >	Rode NT1-A >	1272	
Soundfile 15.4b	Tenor >	Rode NT1-A >	1272 >	Tube Tech CL-1B
Soundfile 16.1a	Bass >	Korby U-47 >	1272	
Soundfile 16.1b	Bass >	Korby U-47 >	1272 >	Tube Tech CL-1B
Soundfile 16.2a	Bass >	Korby 251 >	1272	
Soundfile 16.2b	Bass >	Korby 251 >	1272 >	Tube Tech CL-1B
Soundfile 16.3a	Bass >	Korby U-67 >	1272	

Soundfile 16.3b	Bass >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 16.4a	Bass >	Rode NT1-A >	1272	
Soundfile 16.4b	Bass >	Rode NT1-A >	1272 >	Tube Tech CL-1B
Soundfile 17.1a	VP Kick >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 17.1b	VP Kick >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 17.2a	VP Kick >	Senn e935 >	API 512c >	Tube Tech CL-1B
Soundfile 17.2b	VP Kick >	Senn e935 >	1272 >	Tube Tech CL-1B
Soundfile 18.1a	VP Snare >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 18.1b	VP Snare >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 18.2a	VP Snare >	Senn e935 >	API 512c >	Tube Tech CL-1B
Soundfile 18.2b	VP Snare >	Senn e935 >	1272 >	Tube Tech CL-1B
Soundfile 19.1a	VP Hat/Cym >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 19.1b	VP Hat/Cym >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 19.2a	VP Hat/Cym >	Senn e935 >	API 512c >	Tube Tech CL-1B
Soundfile 19.2b	VP Hat/Cym >	Senn e935 >	1272 >	Tube Tech CL-1B
Soundfile 20.1a	VP Osc/Pitch >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 20.1b	VP Osc/Pitch >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 20.2a	VP Osc/Pitch >	Senn e935 >	API 512c >	Tube Tech CL-1B
Soundfile 20.2b	VP Osc/Pitch >	Senn e935 >	1272 >	Tube Tech CL-1B
Soundfile 21.1a	VP Full Kit >	Korby U-67 >	API 512c >	Tube Tech CL-1B
Soundfile 21.1b	VP Full Kit >	Korby U-67 >	1272 >	Tube Tech CL-1B
Soundfile 21.2a	VP Full Kit >	Senn e935 >	API 512c >	Tube Tech CL-1B
Soundfile 21.2b	VP Full Kit >	Senn e935 >	1272 >	Tube Tech CL-1B
Soundfile 22.1	6-Voice Group Vocals (unedited)			
Soundfile 22.2	6-Voice Group Vocals (edited)			
Soundfile 22.3	6-Voice Group Vocals (edited and mixed with track)			

Soundfile 23.1	Vocal Improv Solo (unedited)
Soundfile 23.2	Vocal Improv Solo (edited)
Soundfile 23.3	Vocal Improv Solo (edited and mixed with track)

APPENDIX F

GLOSSARY

A cappella Sung without instrumental accompaniment

Acoustic Of or relating to sound, the sense of hearing, or the science of sound.

Acoustical Traps Attenuates airborne sound waves by increasing air resistance, thus reducing the amplitude of the waves. The energy is dissipated as heat. The purpose is to reduce, but not entirely eliminate resonance within the room. Acoustic foam deals more with the mid and high frequencies. To deal with lower frequencies, much thicker pieces of acoustic foam are needed; large pieces of acoustic foam are often placed in the corners of a room and are called acoustic foam corner bass traps.

Algorithm A step-by-step problem-solving procedure, especially an established, recursive computational procedure for solving a problem in a finite number of steps.

Amplification The process of increasing the magnitude of a variable quantity, especially the magnitude of voltage, power, or current, without altering any other quality. The increase in strength of an electrical signal by means of an amplifier.

Amplitude On a graph of a sound wave, the sound pressure of the waveform peak. On a graph of an electrical signal, the voltage of the waveform peak. The amplitude of a sound wave or signal as measured on a meter is 0.707 times the peak amplitude.

Analog Measuring or representing data by means of one or more physical properties that can express any value along a continuous scale. For example, the position of the hands of a clock is an analog representation of time.

Analog filter A basic building block of signal processing much used in electronics. Amongst their many applications are the separation of an audio signal before application to bass, mid-range and tweeter loudspeakers.

Analog Mixing Bus A external piece of hardware that receives 16 analog outputs from any DAW interface and combines them into a stereo mix.

Analog Signal Any continuous signal.

Analog Synthesizer A synthesizer that uses analog circuits and analog computer techniques to generate sound electronically. The earliest analog synthesizers in

the 1920s and 1930s such as the Trautonium were built with a variety of vacuum-tube (thermionic valve) and electro-mechanical technologies. After the 1960s, analog synthesizers were built with a variety of operational amplifier (op-amp) integrated circuits, along with potentiometer (pot, or variable resistor) to adjust the traits of the sound that is produced. Analog synthesizers also use low-pass filters and high-pass filters to modify the sound. While 1960s-era analog synthesizers such as the Moog used a number of independent electronic modules connected by patch cables, later analog synthesizers such as the Minimoog integrated them into single units, eliminating patch cords in favor of integrated signal routing systems.

Analog-to-digital (A/D) Converter A circuit that converts an analog audio signal into a stream of digital data (bitstream).

Anti-aliasing filter In an A/D converter, a lowpass filter that removes all frequencies above 20kHz before sampling to prevent audio artifacts called aliasing.

Anti-imaging filter In a D/A converter, a lowpass filter that smooths the voltage steps in the analog signal that was generated by translating digital numbers into analog voltages. The anti-imaging filter recovers the waveform of the original analog signal.

API Automated Processes Incorporated

Aspiration noise Noise from the act of breathing

Atmospheric pressure All microphones work by sensing the pressure difference on either side of a thin sheet known as a diaphragm. In a pressure-operated mic, one side of the diaphragm is open to the atmosphere and is able to respond to the microscopic changes in pressure representing sound. In a pressure-gradient mic, the diaphragm is still sensitive to the difference in pressure on either side, but this time both sides are exposed to the atmosphere, and therefore to the changing pressure caused by passing sound waves. A sound arriving in the plane of the diaphragm will present identical pressures on both sides and, consequently, there will be no net movement. There is no pressure gradient across the diaphragm and so the microphone is deaf to sounds on this axis. In contrast, sounds arriving perpendicular to the diaphragm will create a large pressure difference between front and rear, and it will be moved a maximum amount as a result.

Attack time In a compressor, the time it takes for gain reduction to occur in response to a musical attack. The 'attack phase' is the period when the compressor is decreasing gain to reach the level that is determined by the ratio. The 'release phase' is the period when the compressor is increasing gain to the level determined by the ratio, or, to zero dB, once the level has fallen below the threshold.

Attenuator In a mixer (or mixing console) input module, an adjustable resistive network that reduces the microphone signal level to prevent overloading of the input transformer and mic preamplifier

Audio Engineer Is concerned with the recording, manipulation, mixing and reproduction of sound. Many audio engineers creatively use technologies to produce sound for film, radio, television, music, electronic products and computer games. Alternatively, the term audio engineer can refer to a scientist or engineer who develops new audio technologies working within the field of acoustical engineering. Audio engineering concerns the creative and practical aspects of sounds including speech and music, as well as the development of new audio technologies and advancing scientific understanding of audible sound.

Auto-tune An audio processor created by Antares Audio Technologies, which uses a proprietary device to measure and alter pitch in vocal and instrumental music recording and performances through use of a phase vocoder. Auto-Tune was initially created by Andy Hildebrand, an engineer working for Exxon. Hildebrand developed methods for interpreting seismic data and subsequently realized that the technology could be used to detect, analyze, and modify the pitch in audio files.

Auxiliary Track A track created in DAW to hold an additive effect for other tracks to send/bus signal over to (i.e. reverb). A certain amount of signal is bussed from other tracks to add the quality of the particular effect housed on an insert channel of the auxiliary track.

Baffles Any object designed to reduce airborne sound.

Bandwidth The difference between the upper and lower frequencies in a continuous set of frequencies. It is typically measured in hertz.

Beat Detective Digidesign Pro Tools's proprietary means of analyzing, editing and manipulating audio or MIDI material that is rhythmic in nature.

Bias Electret microphone elements typically include a junction field-effect transistor as an impedance converter to drive other electronics within a few meters of the microphone. The operating current of this JFET is typically 0.1 to 0.5 mA and is often referred to as bias, which is different from the phantom power interface which supplies 48 volts to operate the backplate of a traditional condenser microphone. Electret microphone bias is sometimes supplied on a separate conductor.

Binary number A base-2 numeral system, represents numeric values using two symbols: typically 0 and 1.

Bit depth In digital audio using pulse-code modulation (PCM), bit depth is the number of bits of information in each sample, and it directly corresponds to the resolution of each sample. Examples of bit depth include Compact Disc Digital Audio, which uses 16 bits per sample, and DVD-Audio and Blu-ray Disc which can support up to 24 bits per sample. Bit depth is only meaningful in reference to a PCM digital signal. Non-PCM formats, such as lossy compression formats like MP3, AAC and Vorbis, do not have associated bit depths.

Blocking Using noise barriers to reflect or absorb the energy of the sound waves.

Bussing Relates to how separate audio channels are routed through a mixing desk, either real or virtual

Buttery A term used by audio engineers to mean a smoothness of sound, usually referring to the sound quality in a microphone or microphone pre-amplifier.

Capacitor A capacitor (originally known as a condenser) is a passive two-terminal electrical component used to store energy electrostatically in an electric field. A microphone consisting of a capacitor with one plate fixed and the other forming the diaphragm moved by sound waves.

Cardioid Commonly used as vocal or speech microphones, since they are good at rejecting sounds from other directions. In three dimensions, the cardioid is shaped like an apple centered around the microphone which is the "stalk" of the apple. The cardioid response reduces pickup from the side and rear, helping to avoid feedback from the monitors.

Clean signal Containing no noise or distortion to the input signal

Compression ratio The amount of gain reduction is determined by ratio: a ratio of 4:1 means that if input level is 4 dB over the threshold, the output signal level will be 1 dB over the threshold. The gain (level) has been reduced by 3 dB. The highest ratio of ∞:1 is often known as 'limiting'. It is commonly achieved using a ratio of 60:1, and effectively denotes that any signal above the threshold will be brought down to the threshold level (except briefly after a sudden increase in input loudness, known as an "attack").

Compressor Either an external hardware or plug-in where the dynamic range, the difference between loud and quiet, of an audio waveform is reduced

Condenser Microphone The condenser microphone, invented at Bell Labs in 1916 by E. C. Wente is also called a capacitor microphone or electrostatic microphone—capacitors were historically called condensers. Here, the diaphragm acts as one plate of a capacitor, and the vibrations produce changes in the distance between the plates.

Control Room The room in the studio occupied by the engineer and producer, also housing the DAW and other outboard gear

Cross-fading used in audio engineering as a mixing technique. A mix engineer will often record two or more takes of a vocal or instrumental part and create a final version which is a composite of the best passages of these takes by crossfading between each track. Usually the crossfader keeps a constant output level.

Cue Station/Box The control box in the vocal booth where the singers can control their own headphone mix with potentiometers.

Cylinder recordings Phonograph cylinders were the earliest commercial medium for recording and reproducing sound. Commonly known simply as "records" in their era of greatest popularity (c. 1888–1915), these cylinder shaped objects had an audio recording engraved on the outside surface which could be reproduced when the cylinder was played on a mechanical phonograph. The commercial mass production of phonograph cylinders ended in 1929.

Damping Damping is an influence within or upon an oscillatory system that has the effect of reducing, restricting or preventing its oscillations. In physical systems, damping is produced by processes that dissipate the energy stored in the oscillation.

Decibel (dB) The decibel (dB) is a logarithmic unit used to express the ratio between two values of a physical quantity (usually measured in units of power or intensity). One of these quantities is often a reference value, and in this case the dB can be used to express the absolute level of the physical quantity. The decibel is also commonly used as a measure of gain or attenuation, the ratio of input and output powers of a system, or of individual factors that contribute to such ratios.

Diaphragm A thin, semi-rigid membrane that vibrates to produce or transmit sound waves

Digital Data that is represented using discrete (discontinuous) values.

Digital Audio Workstation (DAW) A digital audio workstation (DAW) is an electronic system designed solely or primarily for recording, editing and playing back digital audio. DAWs were originally tape-less, microprocessor-based systems such as the Synclavier. Modern DAWs are software running on computers with audio interface hardware. A computer-based DAW has four basic components: a computer, a sound card (also called a sound converter or audio interface), a digital audio editor software, and at least one input device for adding or modifying musical note data.

Digital Click An undesirable sound created by the sudden interruption of the waveform in Digital Audio Workstation platform, happening most often when a

recorded region is not faded in or out or crossfade with another adjacent recorded region.

Digital Pop An undesirable sound created by the sudden interruption of the waveform in Digital Audio Workstation platform, happening most often when a recorded region is not faded in or out or crossfade with another adjacent recorded region.

Digital signal A discrete-time signal for which not only the time but also the amplitude has been made discrete; in other words, its samples take on only values from a discrete set (a countable set that can be mapped one-to-one to a subset of integers). If that discrete set is finite, the discrete values can be represented with digital words of a finite width. The process of converting a continuous-valued discrete-time signal to a digital (discrete-valued discrete-time) signal is known as analog-to-digital conversion. It usually proceeds by replacing each original sample value by an approximation selected from a given discrete set (for example by truncating or rounding, but much more sophisticated methods exist), a process known as quantization. This process loses information, and so discrete-valued signals are only an approximation of the converted continuous-valued discrete-time signal, itself only an approximation of the original continuous-valued continuous-time signal.

Diodes In electronics, a diode is a two-terminal electronic component with asymmetric conductance, it has low (ideally zero) resistance to current flow in one direction, and high (ideally infinite) resistance in the other. A vacuum tube diode has two electrodes, a plate (anode) and a heated cathode.

Discreet-time signal A discrete signal or discrete-time signal is a time series consisting of a sequence of quantities. In other words, it is a time series that is a function over a domain of integers. Unlike a continuous-time signal, a discrete-time signal is not a function of a continuous argument; however, it may have been obtained by sampling from a continuous-time signal, and then each value in the sequence is called a sample. When a discrete-time signal obtained by sampling a sequence corresponding to uniformly spaced times, it has an associated sampling rate; the sampling rate is not apparent in the data sequence, and so needs to be associated as a characteristic unit of the system.

Distortion The alteration of the original shape (or other characteristic) of something, such as an object, image, sound or waveform. Distortion is usually unwanted, and so engineers strive to eliminate distortion, or minimize it. In some situations, however, distortion may be desirable. The important signal processing operation of heterodyning is based on nonlinear mixing of signals to cause intermodulation. Distortion is also used as a musical effect, particularly with electric guitars.

Dither An intentionally applied form of noise used to randomize quantization error, preventing large-scale patterns such as color banding in images. Dither is

routinely used in processing of both digital audio and digital video data, and is often one of the last stages of audio production to compact disc.

Dynamic Microphone Moving-coil microphones use the same dynamic principle as in a loudspeaker, only reversed. A small movable induction coil, positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters through the windscreen of the microphone, the sound wave moves the diaphragm. When the diaphragm vibrates, the coil moves in the magnetic field, producing a varying current in the coil through electromagnetic induction.

Dynamic range (DR or DNR) The ratio between the largest and smallest possible values of a changeable quantity, such as in signals like sound and light. It is measured as a ratio, or as a base-10 (decibel) or base-2 (doublings, bits or stops) logarithmic value.

Dynamics A variation and contrast in force or intensity.

Elastic Time An option in Pro Tools of tightening up a multi-track drum performance or snapping whole bass line into place.

Electret Condenser An electret microphone is a type of capacitor microphone invented by Gerhard Sessler and Jim West at Bell laboratories in 1962. The externally applied charge described above under condenser microphones is replaced by a permanent charge in an electret material. An electret is a ferroelectric material that has been permanently electrically charged or polarized. The name comes from electrostatic and magnet; a static charge is embedded in an electret by alignment of the static charges in the material. Unlike other capacitor microphones, they require no polarizing voltage, but often contain an integrated preamplifier that does require power (often incorrectly called polarizing power or bias). This preamplifier is frequently phantom powered in sound reinforcement and studio applications.

Electro-optical A branch of technology involving components, devices and systems which operate by modification of the optical properties of a material by an electric field.

Equalization (EQ) Is the process of adjusting the balance between frequency components within an electronic signal.

Equalizer Strengthen (boost) or weaken (cut) the energy of specific frequency bands.

Fader A device for gradually increasing or decreasing the level of an audio signal.

Fading In An editing technique of raising the volume level of a track from minus infinity to the desired dB level over a particular period of time.

Fading Out An editing technique of lowering the volume level of a track from the desired dB level to minus infinity over a particular period of time.

Field-Effect Transistor Microphone (FET) A microphone in which a membrane is used as the gate to a field-effect transistor (FET) located just below it, and motion of the membrane modulates the current between the source and drain of the transistor.

Formants The spectral peaks of the sound spectrum of the voice or an acoustic resonance of the human vocal tract. Formants are the distinguishing or meaningful frequency components of human speech and of singing. The formant with the lowest frequency is called f_1, the second f_2, and the third f_3. Most often the two first formants, f_1 and f_2, are enough to disambiguate the vowel.

Frequency Response The quantitative measure of the output spectrum of a system or device in response to a stimulus, and is used to characterize the dynamics of the system. It is a measure of magnitude and phase of the output as a function of frequency, in comparison to the input.

Frequency Spectrum Any signal that can be represented as an amplitude that varies with time has a corresponding frequency spectrum. Often, the frequency spectrum clearly shows harmonics, visible as distinct spikes or lines at particular frequencies, that provide insight into the mechanisms that generate the entire signal.

Fundamental Frequency Often referred to simply as the fundamental, is defined as the lowest frequency of a periodic waveform.

GAMA Golden Age of Microphone Amplifier

Hard-clipping Clipping may be described as hard, in cases where the signal is strictly limited at the threshold, producing a flat cutoff. Hard clipping results in many high frequency harmonics. In the audio domain, clipping may be heard as general distortion or as pops. In digital signal processing, clipping occurs when the signal is restricted by the range of a chosen representation. Clipping is preferable to the alternative in digital systems, wrapping, which occurs if the digital hardware is allowed to "overflow", ignoring the most significant bits of the magnitude, and sometimes even the sign of the sample value, resulting in gross distortion of the signal.

Harmonic Distortion The components in a loudspeaker, amplifier or microphone or other equipment produce a more accurate reproduction by reducing harmonics added by electronics and audio media.

Harmonic Frequency A multiple of a fundamental frequency. A fundamental frequency of 500Hz has a first harmonic frequency of 1000Hz, double the

fundamental frequency. Its second harmonic is 1500Hz, the third harmonic is 2000Hz and so on. A musical instrument produces both fundamental and harmonic frequencies, which allows the human ear to discern the differences between instruments even if they are playing the same note.

Harmonic A component frequency of the signal that is an integer multiple of the fundamental frequency, i.e. if the fundamental frequency is f, the harmonics have frequencies 2f, 3f, 4f, etc. The harmonics have the property that they are all periodic at the fundamental frequency, therefore the sum of harmonics is also periodic at that frequency. Harmonic frequencies are equally spaced by the width of the fundamental frequency and can be found by repeatedly adding that frequency. For example, if the fundamental frequency (first harmonic) is 25 Hz, the frequencies of the next harmonics are: 50 Hz (2nd harmonic), 75 Hz (3rd harmonic), 100 Hz (4th harmonic) etc.

Head Register The higher ranges of the voice in speaking or singing; the vibrations of sung notes are felt in the head.

Headroom The amount by which the signal-handling capabilities of an audio system exceed a designated level known as Permitted Maximum Level (PML). Headroom can be thought of as a safety zone allowing transient audio peaks to exceed the PML without exceeding the signal capabilities of an audio system (digital clipping, for example). Various standards bodies recommend various levels as Permitted Maximum Level.

Hertz The unit of frequency in the International System of Units (SI). It is defined as the number of cycles per second of a periodic phenomenon.[1] One of its most common uses is the description of the sine wave, particularly those used in radio and audio applications, such as the frequency of musical tones. The unit is named for Heinrich Rudolf Hertz, who was the first to conclusively prove the existence of electromagnetic waves.

Heterodyning A radio signal processing technique invented in 1901 by Canadian inventor-engineer Reginald Fessenden, in which new frequencies are created by combining or mixing two frequencies. Heterodyning is useful for frequency shifting signals into a new frequency range, and is also involved in the processes of modulation and demodulation. The two frequencies are combined in a nonlinear signal-processing device such as a vacuum tube, transistor, or diode, usually called a mixer. In the most common application, two signals at frequencies f1 and f2 are mixed, creating two new signals, one at the sum f1 + f2 of the two frequencies, and the other at the difference f1 − f2. These new frequencies are called heterodynes. Typically only one of the new frequencies is desired, and the other signal is filtered out of the output of the mixer. Heterodynes are related to the phenomenon of "beats" in acoustics.

High Pass Filter An electronic filter that passes high-frequency signals but attenuates (reduces the amplitude of) signals with frequencies lower than the cutoff frequency.

Intonation Is a musician's realization of pitch accuracy, or the pitch accuracy of a musical instrument. Intonation may be flat, sharp.

Interpolation A method of constructing new data points within the range of a discrete set of known data points.

kHz 1000 Hertz

Lead-in The spot in the session where the track is started before punching in

Long-term average spectrum analysis **O**ffers representative information on voice timbre providing spectral information averaged over time. It is particularly useful when persistent spectral features are under investigation. LTAS quantifies the quality of voices, pointing differences between gender, age, professional - spoken and sang - and dysphonic voices. The LTAS has been used a lot in researches that investigate voice. As it evidences the contribution of the glottic source and of resonance to the quality of voice, it provides objective parameters for the evaluation of this aspect which usually depends on our auditive perception.

Low Pass Filter A filter that passes low-frequency signals and attenuates (reduces the amplitude of) signals with frequencies higher than the cutoff frequency.

Leveling amplifier A simple compressor that provides gain reduction and output increase, but has fixed levels of threshold, attack, release & ratio.

Linear regions The linear operating region of a device, for example a transistor, is where a dependent variable (such as the transistor collector current) is directly proportional to an independent variable (such as the base current). This ensures that an analog output is an accurate representation of an input, typically with higher amplitude (amplified). A typical example of linear equipment is a high fidelity audio amplifier, which must amplify a signal without changing its waveform.

Low-level noise The residual low level sound (usually hiss and hum) that is heard in quiet periods of program. In audio engineering it can also refer to the unwanted residual electronic noise signal that gives rise to acoustic noise heard as hiss.

Magnetic Field A mathematical description of the magnetic influence of electric currents and magnetic materials. The magnetic field at any given point is specified by both a direction and a magnitude (or strength); as such it is a vector field. All moving charged particles produce magnetic fields. Moving point

charges, such as electrons, produce complicated but well known magnetic fields that depend on the charge, velocity, and acceleration of the particles.

Magnetic tape A medium for magnetic recording, made of a thin magnetizable coating on a long, narrow strip of plastic film. It was developed in Germany, based on magnetic wire recording. Magnetic tape was invented for recording sound by Fritz Pfleumer in 1928 in Germany, based on the invention of magnetic wire recording by Valdemar Poulsen in 1898. Pfleumer's invention used a ferric oxide (Fe2O3) powder coating on a long strip of paper. This invention was further developed by the German electronics company AEG, which manufactured the recording machines and BASF, which manufactured the tape. In 1933, working for AEG, Eduard Schuller developed the ring-shaped tape head. Previous head designs were needle-shaped and tended to shred the tape. An important discovery made in this period was the technique of AC biasing which improved the fidelity of the recorded audio signal by increasing the effective linearity of the recording medium.

Mask Area of the face below the eyes and between the cheeks and upper lip where voice resonance is placed and felt.

Mastering A form of audio post-production, is the process of preparing and transferring recorded audio from a source containing the final mix to a data storage device (the master); the source from which all copies will be produced (via methods such as pressing, duplication or replication).

Melodyne A program for OS X or Windows that offers truly extraordinary possibilities for the editing of audio. Melodyne recognizes the notes that are sung or played in your recording and displays them for editing.

Microphone Pre-Amplifier A sound engineering device that prepares a microphone signal to be processed by other equipment. Microphone signals are often too weak to be transmitted to units such as mixing consoles and recording devices with adequate quality. Preamplifiers increase a microphone signal to line level (i.e. the level of signal strength required by such devices) by providing stable gain while preventing induced noise that would otherwise distort the signal.

Microphone Proximity The distance the singer is to the microphone.

Mixed Voice Blending all three resonators in your voice: chest resonance, oral resonance (mouth tone), and sinus cavity resonance (head voice). These three resonances working together constitutes mixed voice.

Mixing The process by which multiple recorded sounds are combined into one or more channels, for instance 2-channel stereo. In the process, the source signals' level, frequency content, dynamics, and panoramic position are manipulated and effects

such as reverb may be added. This practical, aesthetic, or otherwise creative treatment is done in order to produce a mix that is more appealing to listeners.

Modulated square wave A modulation technique that conforms the width of the pulse, formally the pulse duration, based on modulator signal information. Although this modulation technique can be used to encode information for transmission, its main use is to allow the control of the power supplied to electrical devices, especially to inertial loads such as motors. The average value of voltage (and current) fed to the load is controlled by turning the switch between supply and load on and off at a fast pace.

Modulation noise Noise which is present only in company with a signal. In analog recorders the recording process has a certain "granularity" due to the fact that the magnetic characteristics of the tape are not completely uniform which causes an irregularity in the recorded signal that sounds like noise. In digital audio systems there is also an "uncertainty" in the level of the signal because of quantization in the A/D converter. This uncertainty sounds like added noise and is not present if the signal is not present.

Mono channel A single channel, as opposed to stereo two channel.

"More Me" Potentiometer The knob on the cue box that allows the singer to hear more of themselves.

Moving coil microphone (Dynamic Microphone) Uses the same dynamic principle as in a loudspeaker, only reversed. A small movable induction coil, positioned in the magnetic field of a permanent magnet, is attached to the diaphragm. When sound enters through the windscreen of the microphone, the sound wave moves the diaphragm. When the diaphragm vibrates, the coil moves in the magnetic field, producing a varying current in the coil through electromagnetic induction.

Noise Any unwanted sound. Noise is not necessarily random. Sounds, particularly loud ones, that disturb people or make it difficult to hear wanted sounds, are noise.

Noise Floor The measure of the signal created from the sum of all the noise sources and unwanted signals within a measurement system, where noise is defined as any signal other than the one being monitored. A common way to lower the noise floor in electronics systems is to cool the system to reduce thermal noise, when this is the major noise source. In special circumstances, the noise floor can also be artificially lowered with digital signal processing techniques.

Noise Gate An electronic device or software that is used to control the volume of an audio signal. Noise gates attenuate signals by a fixed amount, known as the range. In its most simple form, a noise gate allows a signal to pass through only when it is above a set threshold: the gate is 'open'. If the signal falls below the threshold no signal is allowed to pass (or the signal is substantially attenuated):

the gate is 'closed'. A noise gate is used when the level of the 'signal' is above the level of the 'noise'. The threshold is set above the level of the 'noise' and so when there is no 'signal' the gate is closed. A noise gate does not remove noise from the signal. When the gate is open both the signal and the noise will pass through.

Nyquist frequency Named after electronic engineer Harry Nyquist, the Nyquist frequency is ½ of the sampling rate of a discrete signal processing system. It is sometimes known as the folding frequency of a sampling system.

Nyquist-Shannon Sampling Theory Named after Harry Nyquist and Claude Shannon, the Nyquist sampling theorem or simply known as the sampling theorem, is a fundamental result in the field of information theory, in particular telecommunications and signal processing. Sampling is the process of converting a signal (for example, a function of continuous time or space) into a numeric sequence (a function of discrete time or space). Shannon's version of the theorem states: If a function x(t) contains no frequencies higher than B hertz, it is completely determined by giving its ordinates at a series of points spaced 1/(2B) seconds apart. In other words, a bandlimited function can be perfectly reconstructed from a countable sequence of samples if the bandlimit, B, is no greater than half the sampling rate (samples per second). The theorem also leads to a formula for reconstruction of the original function from its samples. When the bandlimit is too high (or there is no bandlimit), the reconstruction exhibits imperfections known as aliasing. The Poisson summation formula provides a graphic understanding of aliasing and an alternative derivation of the theorem, using the perspective of the function's Fourier transform. In practice, infinite sequences, perfect sampling, and perfect interpolation are all replaced by approximations, deviating from the ideal mathematical reconstruction. Modern statements of the theorem are sometimes carefully, explicitly stating that x(t) must contain no sinusoidal component at exactly frequency B, or that B must be strictly less than ½ the sample rate. The theorem does not preclude the possibility of perfect reconstruction under special circumstances that do not satisfy the sample-rate criterion. It is a sufficient, but not necessary, condition.

Offset When a recorded track is offset intentionally to account for a delay that is acquired from a processing insert.

On-axis The microphone's response to sound coming directly on-axis towards its diaphragm (0°).

One take The first pass or attempt ("take") of recording a track.

Optimized To make as perfect or effective as possible.

Oscillating The repetitive variation, typically in time, of some measure about a central value (often a point of equilibrium) or between two or more different states. Familiar examples include a swinging pendulum and AC power. The term

vibration is sometimes used more narrowly to mean a mechanical oscillation but is sometimes used as a synonym of "oscillation".

Outboard gear External effects units that can be used either during a live performance or in the recording studio. These are separate from the effects that may be applied by using a mixing console or a digital audio workstation. Some outboard effects units and digital signal processing (DSP) boxes commonly found in a studio are: analog-to-digital and digital-to-analog converters, musical instrument digital interfaces (MIDIs), microphone preamplifiers ("microphone preamp", "mic preamps" or "preamps"), equalizers ("EQs"), dynamics effects units: compressors/limiters and noise gates, time-based effects units: reverb, flanging, delay, echo, chorus etc.

Oversampling The process of sampling a signal with a sampling frequency significantly higher than the Nyquist frequency. Theoretically a bandwidth-limited signal can be perfectly reconstructed if sampled at or above the Nyquist frequency. Oversampling improves resolution, reduces noise and helps avoid aliasing and phase distortion by relaxing anti-aliasing filter performance requirements.

Panning The spread of a sound signal (either monaural or stereophonic pairs) into a new stereo or multi-channel sound field. A typical physical recording console pan control is a knob with a pointer which can be placed from the 8 o'clock dial position fully left to the 4 o'clock position fully right. Audio mixing software replaces the knob with an on-screen "virtual knob" or slider for each audio source track which functions identically to its counterpart on a physical mix console.

Parametric equalization Multi-band variable equalizers which allow users to control the three primary parameters: amplitude, center frequency and bandwidth. The amplitude of each band can be controlled, and the center frequency can be shifted, and bandwidth ("Q") can be widened or narrowed. Parametric equalizers are capable of making much more precise adjustments to sound than other equalizers, and are commonly used in sound recording and live sound reinforcement.

PCI Computer Card A local computer bus for attaching hardware devices in a computer. Attached devices can take either the form of an integrated circuit fitted onto the motherboard itself, (called a planar device in the PCI specification) or an expansion card that fits into a slot.

Pentode An electronic device having five active electrodes. The term most commonly applies to a three-grid amplifying vacuum tube (thermionic valve), which was invented by the Dutchman Bernhard D.H. Tellegen in 1926. The pentode consists of an evacuated glass envelope containing five electrodes in this order: a cathode heated by a filament, a control grid, a screen grid, a suppressor grid, and a plate (anode). The obsolete consumer tubes are still used in a few legacy and specialty vacuum tube audio devices.

Permutation Relates to the act of permuting (rearranging) objects or values. Informally, a permutation of a set of objects is an arrangement of those objects into a particular order.

Phantom power A method for transmitting DC electric power through microphone cables to operate microphones that contain active electronic circuitry. It is best known as a convenient power source for condenser microphones, though many active direct boxes also use it. The technique is also used in other applications where power supply and signal communication take place over the same wires. Phantom power supplies are often built into mixing desks, microphone preamplifiers and similar equipment.

Phase shift If you gradually delay the audio going through a second channel, the peaks and troughs of the two sine waves shift out of alignment. Because of the unique properties of sine waves, the combination of the two channels will now still produce a sine wave of the same frequency, but its level will be lower than if the two channels were in phase, and we say that partial phase cancellation has occurred. When the second channel is delayed such that its peaks coincide exactly with the first channel's troughs (and vice versa), the two waveforms will combine to produce silence. At this point we say that the waveforms are completely 'out of phase' with each other and that total phase cancellation has occurred.

Phonograph Also called gramophone or record player, is a device introduced in 1877 for the recording and reproduction of sound recordings. The recordings played on such a device consist of waveforms that are engraved onto a rotating cylinder or disc. As the cylinder or disc rotates, a stylus or needle traces the waveforms and vibrates to reproduce the recorded sound waves. The phonograph was invented in 1877 by Thomas Edison.

Pitch Drift line The line that represents slow fluctuations in pitch in the Melodyne program of the kind that are usually unintentional and symptomatic of poor technique.

Playlist In recording multiple takes, provides a good way of keeping track of all the takes. Before the studio recording session, create template Sessions with the appropriate number of tracks and set up the input and output routing, track names, grouping and so on. Group all of the tracks and create a new Playlist by selecting New Playlist in one of the track name pull-down menus in the Edit window. This automatically attaches a '.01' suffix to the end of each individual track name in the group, clearly indicating take one of the recording process. The New Playlist pop-up on a track creates new Playlists for all tracks in the same active group. When you record subsequent takes, click the pull-down menu in one of the group of tracks and choose New Playlist again. Pro Tools now advances all of the playlists in that group to '.02', which becomes your take two, and so on. This way, you always have a numerical ident associated with the Playlist and the audio for each take, because the track names together with their suffixes are transferred

automatically to the audio files as you record. The 'helpful advice' in the control room will refer to performances by their take numbers, so you can very quickly tab through to the desired takes and assemble a composite track in the original Playlist (the one without a numerical suffix).

Plosives An oral stop of a consonant that is made by blocking a part of the mouth so that no air can pass through, and the pressure increases behind the place where it is blocked, and when the air is allowed to pass through again, this sound is created. This sound is the plosive consonant. The blocking is usually done using the tongue, the lips or the throat. Plosives can be voiced or voiceless.

Plug-in A software version of external hardware that can be inserted on a track as either an Instrument or an Effect.

Pre-recorded To record at an earlier time for later presentation or use.

Pop-screen (Pop-filter/Pop-sheild) An anti-pop noise protection filter for microphones, typically used in a recording studio. It serves to reduce or eliminate 'popping' sounds caused by the mechanical impact of fast moving air on the microphone during recorded speech and singing. It can also protect against the accumulation of saliva on the microphone element. Salts in human saliva are corrosive and thus use of a pop filter may prolong the life of the microphone.

Potentiometer (pot) A three-terminal resistor with a sliding contact that forms an adjustable voltage divider. If only two terminals are used, one end and the wiper, it acts as a variable resistor or rheostat. A potentiometer measuring instrument is essentially a voltage divider used for measuring electric potential (voltage); the component is an implementation of the same principle, hence its name.

Presence The area that is close to someone or the part of space within one's immediate vicinity.

Print through (bleed-through) A generally undesirable effect that arises in the use of magnetic tape for storing analogue information, in particular music. Print-through is a category of noise caused by contact transfer of signal patterns from one layer of tape to another. Print can take two forms: 1) thermo-remanent magnetization induced by temperature, and 2) anhysteretic magnetization caused by an external magnetic field. The former is unstable over time and can be easily erased by rewinding a tape and letting it sit so that the patterns formed by the contact of upper and lower layers begin to erase each other and form new patterns with the repositioning of upper/lower layers after rewinding. This type of contact printing begins immediately after a recording and increases over time at a rate dependent on the temperature of the storage conditions.

Producer An individual working within the music industry, whose job is to oversee and manage the recording (i.e. "production") of an artist's music.

Punch in In modern, Digital Audio Workstation-based recording environments, punching in and out can be done automatically by pre-selecting the in and out points on the timeline of the DAW. When the record button is pressed, the DAW software will play back the previously recorded track outside of these points. As soon as the playhead reaches the in-point, the recording begins and the previously recorded material is muted. At the out-point, recording stops and the software reverts to playback. When done "on the fly," punch-in is technically known as "quick punch."

Punch out In modern, Digital Audio Workstation-based recording environments, punching in and out can be done automatically by pre-selecting the in and out points on the timeline of the DAW. When the record button is pressed, the DAW software will play back the previously recorded track outside of these points. As soon as the playhead reaches the in-point, the recording begins and the previously recorded material is muted. At the out-point, recording stops and the software reverts to playback. When done "on the fly," punch-in is technically known as "quick punch."

Quantization In analog-to-digital conversion, the difference between the actual analog value and quantized digital value is called quantization error or quantization distortion. This error is either due to rounding or truncation. The error signal is sometimes modeled as an additional random signal called quantization noise because of its stochastic behaviour. Quantization is involved to some degree in nearly all digital signal processing, as the process of representing a signal in digital form ordinarily involves rounding. Quantization also forms the core of essentially all lossy compression algorithms.

Raw Vocals Vocals recorded but untouched by editing or any other kind of processing

Reconstruction filter In a mixed-signal system (analog and digital), a reconstruction filter (or anti-imaging filter) is used to construct a smooth analogue signal from a digital input, as in the case of a digital to analog converter (DAC) or other sampled data output device.

Recorder head bumps Fluctuations in low-frequency response caused by Higher speeds used in professional analog tape recorders.

Reed-Solomon Coding System Non-binary cyclic error-correcting codes invented by Irving S. Reed and Gustave Solomon. They described a systematic way of building codes that could detect and correct multiple random symbol errors. By adding t check symbols to the data, an RS code can detect any combination of up to t erroneous symbols, or correct up to [t/2] symbols. As an erasure code, it can correct up to t known erasures, or it can detect and correct combinations of errors

and erasures. In Reed–Solomon coding, source symbols are viewed as coefficients of a polynomial p(x) over a finite field. The original idea was to create n code symbols from k source symbols by oversampling p(x) at n > k distinct points, transmit the sampled points, and use interpolation techniques at the receiver to recover the original message.

Reel-to-reel Recorder The form of magnetic tape audio recording in which the recording medium is held on a reel, rather than being securely contained within a cassette. In use, the supply reel or feed reel containing the tape is mounted on a spindle; the end of the tape is manually pulled out of the reel, threaded through mechanical guides and a tape head assembly, and attached by friction to the hub of a second, initially empty takeup reel.

Reference Mixing A rough mix post recording with the intention to allow the producer to hear the quality of the recording. Mixing techniques used in this process are minimal usually on including minimal reverb, compression, and EQ with slight level adjustments.

Region An area in which recorded material or a recorded waveform exists which looks like a shaded area of block within Digital Audio Workstation software.

Release The period when the compressor is increasing gain to the level determined by the ratio, or, to zero dB, once the level has fallen below the threshold.

Resolution A measure of digital audio quality.

Resonating Vocal Cavities The process by which the basic product of phonation is enhanced in timbre and/or intensity by the air-filled cavities through which it passes on its way to the outside air. The result of resonation is to make a better sound.

Reverberation The persistence of sound in a particular space after the original sound is produced. A reverberation, or reverb, is created when a sound is produced in an enclosed space causing a large number of echoes to build up and then slowly decay as the sound is absorbed by the walls and air. This is most noticeable when the sound source stops but the reflections continue, decreasing in amplitude, until they can no longer be heard. The length of this sound decay, or reverberation time, receives special consideration in the architectural design of large chambers, which need to have specific reverberation times to achieve optimum performance for their intended activity. In comparison to a distinct echo that is 50 to 100 ms after the initial sound, reverberation is many thousands of echoes that arrive in very quick succession (.01 – 1 ms between echoes). As time passes, the volume of the many echoes is reduced until the echoes cannot be heard at all.

Ribbon Microphone (ribbon velocity microphone) A type of microphone that uses a thin aluminum, duraluminum or nanofilm electrically conductive ribbon placed

between the poles of a magnet to generate voltages by electromagnetic induction. Ribbon microphones are typically bidirectional, meaning they pick up sounds equally well from either side of the microphone.

Ride the level A technique where the engineer keeps his finger on the mixing console fader while recording in order to attenuate some of the volume changes by the singer.

Rolled off in the mix Refers to an EQ technique of diminishing the lower frequencies below a certain point. Usually below 150, 100 or 50 Hz.

Rotary encoder (shaft encoder) An electro-mechanical device that converts the angular position or motion of a shaft or axle to an analog or digital code. It provides cyclical outputs (only) when the encoder is rotated. They can be either mechanical or optical. The mechanical type requires debouncing and is typically used as digital potentiometers on equipment including consumer devices.

Runway Refers to where the engineer starts the session prior to recording a track.

Samples The reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal). A sample refers to a value or set of values at a point in time and/or space.

Sampling rate The number of samples per unit of time (usually seconds) taken from a continuous signal to make a discrete signal. For time-domain signals, the unit for sampling rate is hertz. The reciprocal of the sampling frequency is the sampling period or sampling interval, which is the time between samples.

Scratch vocal track A vocal track that is recorded during instrumental tracking that is intended only as a guide to be thrown away, but can be kept if wanted.

Sends A place on the mixer where individual tracks can be routed to another track such as an auxiliary track.

Sheen A high frequency gloss or air on the vocal

Shelf mode A technique used in equalization where the highest or lowest frequency bands are shelved either up or down to infinity.

Sibilance Of, characterized by, or producing a hissing sound like that of (s) or (sh): the sibilant consonants.

Signal Path The path an audio signal follows, through several pieces of gear chained together.

Slap-back Undesirable decaying echo picked up by the microphone on the recording stage usually cause by parallel walls or reflection from glass or other hard surfaces.

Smoothing filter A Savitzky–Golay filter is a digital filter that can be applied to a set of digital data points for the purpose of smoothing the data, that is, to increase the signal-to-noise ratio without greatly distorting the signal. This is achieved, in a process known as convolution, by fitting successive sub-sets of adjacent data points with a low-degree polynomial by the method of linear least squares. It is inevitable that the signal will be distorted in the convolution process. From property 3 above, when data which has a peak is smoothed the peak height will be reduced and the half-width will be increased. Both the extent of the distortion and S/N (signal-to-noise ratio) improvement: decrease as the degree of the polynomial increases and increase as the width, m of the convolution function increases

Soft-clipping A soft knee slowly increases the compression ratio as the level increases and eventually reaches the compression ratio set by the user. A soft knee reduces the audible change from uncompressed to compressed, especially for higher ratios where a hard knee changeover would be more noticeable. It may be described as soft, in cases where the clipped signal continues to follow the original at a reduced gain. Soft clipping results in fewer higher order harmonics and intermodulation distortion components. In the audio domain, clipping may be heard as general distortion or as pops. In digital signal processing, clipping occurs when the signal is restricted by the range of a chosen representation. Clipping is preferable to the alternative in digital systems, wrapping, which occurs if the digital hardware is allowed to "overflow", ignoring the most significant bits of the magnitude, and sometimes even the sign of the sample value, resulting in gross distortion of the signal.

Solid-state microphone Solid-state electronics are those circuits or devices built entirely from solid materials and in which the electrons, or other charge carriers, are confined entirely within the solid material. In a solid-state component, the current is confined to solid elements and compounds engineered specifically to switch and amplify it. Current flow can be understood in two forms: as negatively charged electrons, and as positively charged electron deficiencies called holes. The solid-state device came into its own with the invention of the transistor in 1947.

Sound Replacer Allows you to replace or mix an existing audio track with new samples from your sound library.

Spectra Are conditions or values that vary over a continuum.

Spectrograph An instrument that separates an incoming wave into a frequency spectrum.

Spectrograph Analysis The use of spectroscopes to analyze spectra.

Standing Waves This phenomenon can occur because the medium is moving in the opposite direction to the wave, or it can arise in a stationary medium as a result of interference between two waves traveling in opposite directions.

Stereo Imaging An audio jargon term used for the aspect of sound recording and reproduction concerning the perceived spatial locations of the sound source(s), both laterally and in depth. An image is 'good' if the performers can be effortlessly located; 'bad' if there is no hope of doing so. A well-made stereo recording, properly reproduced, can provide good imaging within the front quadrant; a well-made Ambisonic recording, properly reproduced, can offer good imaging all around the listener and even including height information.

Stereo Mix Simultaneous processing of both left and right audio channels.

Stereo Spectrum The way in which instruments are panned in a stereo mix.

Stereo Track Usually used when a stereo signal has been recorded using 2 microphones per instrument.

Take A term referring to one recorded pass, one whole recorded region or recorded performance on a particular track.

Tape compression A warm analog tape sound that happens when running the level slightly into distortion.

Tape recorder An audio storage device that records and plays back sounds, including articulated voices, usually using magnetic tape, either wound on a reel or in a cassette, for storage. In its present day form, it records a fluctuating signal by moving the tape across a tape head that polarizes the magnetic domains in the tape in proportion to the audio signal. Tape-recording devices include reel-to-reel tape deck and the cassette deck.

Tape Simulation An analog processor that recreates the sonic imprint of vintage tape recorders. The effect provides the roundness, the punch, the compression and the saturation of magnetic tape.

Tape stretching When the analog tape is stretched or crinkled causing a distortion of pitch.

Third harmonic distortion A measurement of the amplitude of the third harmonic of the input frequency and is the most prominent distortion component in analog magnetic recording systems. The third-harmonic level was used as a convenient figure-of-merit because the 2nd harmonic is difficult to hear, since it tends to reinforce the pitch of the fundamental. The 3rd harmonic is easy to detect on pure tones (although less so on music), thus it makes a good benchmark for comparing

sound “off tape” with the original. The distorted tone has an edge to it, containing a component one octave and a fifth above the fundamental. For this reason the third-harmonic is also called a musical twelfth (Octave + fifth).

Threshhold A compressor reduces the level of an audio signal if its amplitude exceeds a certain threshold. It is commonly set in dB, where a lower threshold (e.g. -60 dB) means a larger portion of the signal will be treated (compared to a higher threshold of −5 dB).

Timbral The combination of qualities of a sound that distinguishes it from other sounds of the same pitch and volume.

Time Stretching An editing process in Pro Tools that allows a region of recorded material to be stretched over time to lengthen or shorten a note.

Track bleed Other unwanted sounds recorded to a track in the background (i.e. other singers, click track or track from headphones).

Tracking Room The room, usually large, where a group of singers or other instruments are recorded.

Tracking Stage The stage in the recording process when the instrumental tracks alone are being recorded.

Transducers A device that converts a signal in one form of energy to another form of energy.

Transient In acoustics and audio, a transient is a high amplitude, short-duration sound at the beginning of a waveform that occurs in phenomena such as musical sounds, noises or speech. It can sometimes contain a high degree of non-periodic components and a higher magnitude of high frequencies than the harmonic content of that sound

Transistors A semiconductor device used to amplify and switch electronic signals and electrical power. It is composed of semiconductor material with at least three terminals for connection to an external circuit. A voltage or current applied to one pair of the transistor's terminals changes the current through another pair of terminals. Because the controlled (output) power can be higher than the controlling (input) power, a transistor can amplify a signal.

Tube Pre-amp A microphone pre-amplifier powered by a vacuum tube.

Two-Bus Analog Summing Mixer Can sum 16 channels (in eight left/right pairs), and has controls for adding 6dB of gain to each pair, or mono-summing each pair before it hits the mix bus. Other control is a master-level trim pot.

Ultrasonic frequencies Of or relating to acoustic frequencies above the range audible to the human ear, or above approximately 20,000 hertz.

Vacuum tube microphone A device controlling electric current through a vacuum in a sealed container. The container is often thin transparent glass in a roughly cylindrical shape. The simplest vacuum tube, the diode, is similar to an incandescent light bulb with an added electrode inside. When the bulb's filament is heated red-hot, electrons are "boiled" off its surface and into the vacuum inside the bulb. If the electrode—called a "plate" or "anode"—is made more positive than the hot filament, a direct current flows through the vacuum to the electrode (a demonstration of the Edison effect). As the current only flows in one direction, it makes it possible to convert an alternating current applied to the filament to direct current. The introduction of a third electrode, a grid between the filament and the plate, yields another function. A voltage applied to the grid controls the current flowing from the filament to the plate. Thus, it allows the device to be used as an electronic amplifier.

Valve Microphone The British equavalent to a vacuum tube.

Vocal Also referred to as “a vocal” or “the vocal.” This is a term used by audio engineers to refer to a single recorded vocal track.

Vocals This is a term used by audio engineers to refer to the entire group of vocals that are being recorded or edited. It can also mean the stage in which the entire recording project is in currently (i.e. We are currently recording vocals).

Vocal Booth A small booth designed within which a singer sings where their vocal is recorded.

Vocal Recording Chain Usually refers to the outboard gear used in succession for recording the vocal.

Vocal Comp (Composite) A composite of or “the best of” the total number of vocal takes.

Vocal Folds (Cords) Are composed of twin infoldings of mucous membrane stretched horizontally, from back to front, across the larynx. They vibrate, modulating the flow of air being expelled from the lungs during phonation.

Vocal Percussion Percussion sounds produced by the mouth and voice.

Vocal Percussionist A person who can produce vocal percussion sounds.

Vocal Track A recorded track containing a waveform of a vocalist singing, sometimes referred to by audio engineers as a “vocal.”

Vocal Tuning Pitch correcting the recorded vocal track.

Vocalign A software plug-in that automatically aligns notes within group vocals

Waveforms The shape and form of a signal such as a wave moving in a physical medium or an abstract representation.

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VITA

Kelly Garner's career encompasses a wide range of professional activities as a working member of the music industry for the past twenty-two years. She recorded her first 4-song demo on analog reel-to-reel, two-inch, 24-track tape in a Nashville recording studio in 1985, then spent the next five years as a college student,-earning a Bachelor of Science in Secondary Math Education from Auburn University in 1990.

Upon deciding to continue to pursue a career in music, Garner relocated to Nashville, Tennessee in order to earn a degree in music at Belmont University. After graduating with a Bachelor of Music in Commercial Vocal Performance in 1992, Kelly was offered a recording contract, beginning her career in Nashville as a nationally distributed recording artist/writer in the Gospel Music market. Kelly wrote and co-produced her debut release for the record label, Brentwood-Benson**,** and has been singing, writing and producing ever since.

From 1998-2003, Garner was signed as an Exclusive Staffwriter for Centergy Music Group, garnering Dove Award and BMI Award nominations for Song of The Year and Most Performed Gospel Song "I Stand Redeemed**"** As a session singer, Kelly was awarded two RIAA Certified Gold Records in 2001 for her work as a background vocalist on major artist recordings in Nashville.

That same year, she agreed to accept a full-time position as a member of the Commercial Voice Faculty at Belmont University while pursuing a Master of Arts degree in Jazz Studies at Middle Tennessee State University, graduating in 2004. Garner remained on Belmont's tenure-track faculty until she resigned in 2006 to manage the recording facility Big Dog Studios in Franklin, TN. She also founded Kelly Garner

Productions, LLC, where she continued to develop and produce independent artists covering a variety of styles, from Jazz to R&B, Gospel, Country and Rock.

The list of artists for whom Garner has served as Recording Engineer or Vocal Editor include Sunny Wilkinson, Lauren Kinhan, Kate Reid, Cindy Morgan, Ronnie Freeman, Jim Van Cleave & Mountain Heart, Avalon, Broken Wire, Jonathan Taylor Martin, Sarah Peacock, Jessa Anderson, Jordan Anderson, Hunter Redmon, Maury Davis, Dakota Green, Mike Speck Trio, Greater Vision, Legacy Five, The Booth Brothers, Buddy Mullins, The Keffers, JP Murie and Kara Reynolds.

Garner has also performed as a Back-Up Singer, sharing the stage with a variety of artists, including Gloria Estefan, New York Voices, George Benson, Steve Miller, Marie Osmond, Ray Stevens, Roseanna Vitro, Karrin Alyson, Kate Reid, Michael W. Smith, Billy Ray Cyrus, Michael McDonald, Shaun Groves, Mac Powell (Third Day), Russ Taff, Wendy Foy Green and Kathy Triccoli.

Kelly Garner was a 2013 Student Downbeat Award winner as a member of the Jazz Vocal Ensemble, "Extensions," a two-time 2012 Student Downbeat Award recipient for Outstanding Jazz Vocal Soloist (graduate level) as well as a member of UM's Downbeat Award-Winning JV1 while pursuing a Doctor of Music Arts degree in Jazz Voice Performance with a cognate in Music Technology at the University of Miami in Coral Gables, FL. She was a Graduate Teaching Assistant and Mancini Fellow within the Jazz Vocal Program, teaching Jazz Voice, Jazz Vocal Ensemble II and Jazz Skills. She also taught Contemporary Voice and Contemporary Skills within the Creative American Music Program, as well as other courses within UM's *Experiential Music Curriculum.* Kelly was a featured Clinician at the 2013 Jazz Education Network

conference in Atlanta, Georgia presenting a course entitled "Studio Recording Techniques for Jazz Vocal Ensembles: Getting the Most out of your Vocals without Compromising Musical Integrity."

Garner has recorded three of her own albums, "Confession of Love," "Undivided Heart," and "I Stand Redeemed," and appears on the movie compilation recording "Left Behind II – Adult Contemporary Collection" as a featured artist/writer/producer alongside fellow producer, Bob Carlisle (Butterfly Kisses). She currently owns and operates Nashville-based, Yellow Tree Music Group, a music publishing company that has seen over 65 songs recorded by various artists since 2010. Her published oeuvre includes over 125 songs recorded by major gospel artists and over 75 original songs and arrangements published in print. Because of her work as a publisher, Kelly is an active member of Nashville's BMI Songwriter Showcase Committee. She continues to work as a Recording Artist, Session Singer, Studio Engineer, Producer and Songwriter in a variety of musical genres.

Kelly grew up in Piedmont, AL as the only child of Martha Y. and James R. Garner.

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