PROFESSIONAL AUDIO PRODUCTION, EDITING, AND MIXING

TECHNIQUES

by

Justin M. Dowse

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Professional Audio Production, Editing, and Mixing Techniques

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ABSTRACT

The purpose of this thesis is to demonstrate professional and efficient music production and music mixing techniques. This is not meant to be a definitive answer to the endless approaches that can be used in music production and mixing. Instead, it is intended to aid learning in a profession where there are no black and white answers since choices in music production and mixing are highly dependent upon the engineer's preferences. While it is true that music is an art and in the end not constrained by rules in the purist form of artistic expression, it is also true that certain practices produce results that are pleasing to a large portion of the population. In the beginning of my journey to become a professional audio engineer I became very frustrated with answers to questions like "How much compression should I use?" or "How do you get the guitars to sound like that?" Even though my questions were incredibly vague due to a lack of proper education and highly dependent on preference, application, and context, there is always at least one possible solution or piece of valuable information that can be conveyed. Throughout the thesis there will be specific explanations of production and mixing techniques/processes that I use in the hope that other engineers can find or begin to find a way to produce, record, and mix music at a professional level.

This abstract appropriately describes the candidate's thesis. I recommend its publication.

Approved

Lorne Bregitzer
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CHAPTER 1 PRODUCTION

Producer Philosophy

As Jack Joseph Puig said in an interview for the book Behind the Glass, “Production is an insanely all encompassing job (Massey).” He or she often acts as the liaison between a record label and the artist. It can be the producer's responsibility to decide which studio the music will be tracked in, which audio engineers will be hired to record the album, what microphone or preamplifier will be used, who will mix the album, who will master the album, what musicians will play on the record, and what instruments the musicians will play.

The producer can play a sort of big brother role and an almost temporary extra member of the band. They have an outsider’s perspective and a fresh ear on the music. They have the unique position of hearing the music without previous emotional attachments that can distort decision-making and some amount of authority to make decisions (Massey).

Yet, the amount of authority and the role a producer plays can change from album to album; artist to artist. So it is important that a producer is flexible in his or her abilities and has the personality to take the role of a leader or sit back and be an intermittent consultant. A producer must be a chameleon.

Because the form of a producer can be so incredibly vague, shapeless, and indefinable, I found myself looking for things that a producer should always look to do. Whether or not his or her advice will be part of the final product is dependent on the situation. But regardless, a producer should always have an open ear for certain things. My hunt for those things that always define a producer began with this project. Another producer might, and most likely, argue a different role but at some point every artist must find at least a small piece of solid ground to plant two feet.

At the beginning of this project I had a general grasp of how a song should develop and a natural inclination to suggest different ideas on how to improve a song. But as I debated how I should describe my role as a producer in the project, I realized that I had not solidified the concept of a producer in my mind. More importantly, I had not decided my personal philosophy as a producer.

So I began to recall the times when my suggestions to artist were accepted or rejected and when the outside listeners enjoyed the final product. Throughout the recording of this album, I presented ideas to Kevin and Graham, the artists. Some ideas were loved. Some were hated. And I began to notice that the ideas that were rejected were a part of my concept of how a song should sound. I was trying to inject my personal taste into the recipe. The ideas that were loved were the ideas that I made an attempt to look through the eyes of Kevin. As one might expect, this is extraordinarily difficult. So I began to devise ways to achieve this effectively. I cannot
say that I have learned to look through the eyes of the artist successfully without fail because that would be braggadocios - especially at such an early stage in my career. But I can say that I now have a clearer concept of how a producer should function on this type of project and a general concept of the role a producer should play in any musical project.

The first and foremost thing a producer should do is to decide whether or not the collaboration in question will be beneficial (Farinella, Producing Hit Records). Furthermore, the producer should urge the artist to be conscious of the producer they choose since their ideas of a good album could differ greatly. I usually like to begin to test the production waters with casual conversation. Most artists have music they can relate to or note as an influence. Sometimes it is fairly obvious when the collaboration would not work. For instance, a country producer probably would not be the best person for a death-metal band. Ultimately, the best litmus test will be to delve into pre-preproduction, recording, and mixing a song together. By the end of that production, both parties will have an idea of each person's work habits and creative direction.

Through the process of an album, from the first pre-production session to the cellophane wrap, the producer must have intensely focused concentration on what the artist is about and what they are trying to achieve. Will it be a pop record? Will it be avant-garde? Does the artist want thick arrangements with 100 tracks or will it be minimalistic and only feature one or two instruments. A great deal of information can be answer through healthy dialogue. Still, much of the artist's vision can be lost in this form of communication. There is also much to be learned by watching an artist. Just as in conversation, much of what an artist wants to communicate is expressed through body language while performing. Are the musician's body gestures aggressive, graceful, tense, calm, or suspenseful? There might be a quirky or unique body movements that only manifest on the stage. And sometimes an artist has not learned how to completely harness some of those unique behaviors that make them special so that they are consistently displayed through the music. The producer can take notes of these unique characteristics and help the artist develop them.

Now while all these things are important, the paramount aspect of making a record is to listen. Even though it may seem obvious I find that I must constantly remind myself to pay attention to the music. So often ego, politics, business practices, money, and all the other things that have nothing to do with quality music influences the direction of an album. Instead, the focus should be on the song at hand and what the artist is trying convey in the song. My job is to enhance that message and all of the best qualities of the music that are already there. I must find what is special about the artist and bring the focus of the recording to that special thing. It is not my job to bring something out of the artist that is not there.

**Focusing On Elements of Music In Production**

To find what an artist is trying to express, it is helpful to examine the music with the major elements of music. There are essentially seven elements of music. They are rhythm, dynamics, melody, harmony, timbre, texture, and structure. (Western Michigan University).
The first things I look at to make sure that the song is working in the intended capacity is the overall feeling that the song induces, what the lyrics are, the structure of the song, and how those aspects work together. When Kevin's music was presented to me I immediately sensed a feeling of melancholy slightly counteracted by a bit of hope. The first thing that came to mind to develop that feeling with some of the songs was to restructure them so that they were dynamically calm for most of the song and ended with loud and intense dynamics.

Much of the dramatic changes in dynamics were created with drums. Several of the pre-productions that Graham recorded hinted to dramatic changes in dynamics but they had not met their full potential. During the drum tracking I treated the situation as a sort of pep rally to really enhance the energy and intensity in Graham's playing. However, sometimes the perceived dynamics of a song section is due to what came before. In certain cases, it made more sense to reduce the volume of what Graham was playing in a verse rather than excessively push the volume of the chorus that was to follow. The perception of louder volume can also be altered by which part of the drum set is being played. Not only does a large crash cymbal naturally produce higher decibel levels than smaller closed high-hats, but it also has a longer sustain that humans interpret as louder than shorter sounds(Holman). This can be taken advantage of by moving a mellow verse from high-hats to intense choruses that ride on a crash cymbal.

Tempo is also important in the early stages. As one might expect, higher tempos create a sense of higher energy while lower tempos create a more relaxed environment. To find the right tempo it is always a good idea to perform with a metronome/ click. People naturally speed sections that are at the top of the song’s dynamics. On the other end of the spectrum, it is common to slow tempo during difficult instrumental sections or dense lyrics. A metronome is key in finding the consistent tempo that optimizes performances throughout the song. It may even reveal sections of a song that would benefit from increases or decreases of tempo. Occasionally, finding a comfortable tempo for the musicians and increasing the tempo by one or two beats per minute can amplify the energy of a song. In the right circumstances a sense of urgency and struggle could improve a song.

As an album continues to develop, I use the other main elements of music as a checklist for consistency with the song's meaning or feel, the album's overall feel and sound, the artist's original vision, and the intended style or genre.

Here are a variety of questions that I might ask the artist and myself to be sure each song and the album as a whole is on the right track.

**Rhythm**

- Are the drums playing the rhythms that best compliment the lyrics and vocal melody? Are the guitar's rhythms too complex or too simple for the song?
- Are there too many individual rhythms among each instrument or could there be more independent rhythms?
- Should the bass guitar rhythms follow the drums or the guitars?
- Are all of the instruments' rhythms locked together? In other words, is each instruments tempo consistent with each other?
If necessary, I will revisit tempo and meter. But that would usually call for a complete reinvention of the song since that is typically my foundation.

**Harmony**

- Are the vocal melodies, guitar melodies, and any other instrumental melodies consonant with the key and the chords/chord progressions?
- Could the song use some dissonance to create a bit of tension?
- Are the chord progressions right for the song and individual sections?

In this case, the bulk of chord progressions were already pretty well established when the songs were presented to me, but it is still important to be open to the idea that a section made need alterations to the progression or possibly an entirely different progression.

**Timbre**

- Are we capturing the right drum, guitar, and bass sounds for the style of the album?
- Could the album use a song or two that has less aggressive drum sounds?
- Could there be a song without drums?
- Do we have the right guitar and amplifier for the given section or song?
- Should there be a fairly consistent guitar sound throughout the album or will there be entirely different guitar sounds for each song?
- Will the bass guitar sound be predominately low frequencies or will it have more of a punk rock sound that features more present high frequencies?
- Could a different microphone or preamplifier be used to capture a brighter or darker sound for the given sound?
- Are there any other timbres/instruments that could be used to provide sonic variety throughout the album?

**Texture**

Separate from the primary melodic theme or song hook, it is important to be aware of the density of melodies occurring once.

- Are there too many melodies happening at one time for the given song section?
- Which melody is most important?
- Could the song benefit from more complexity?
- Could the song benefit from more simplicity?

For pop music I prefer to have one to three melodies happening at once. Three would be the absolute most and two would be somewhat common. Most of the time I prefer to have the main melody, usually vocal melody, and occasionally a supporting melody with guitar or some other instrument like piano.
Melody

Producer Butch Walker has the view that, “melodies are everything” (Metro Music Scene). A memorable melody is the most important aspect in popular music. In this situation, Kevin already had his vocal melodies worked through for the most part so the melodies were generally not in question. Instead, most questions throughout the production process were aimed to support the main melodies. We constantly put each instrument’s part into question to be sure that it was not lessening the importance and impact of the primary melody. Usually the electric guitar was most in question since it was the main source of supporting melodies.

Now even though the vocal melodies were fairly decided from the beginning there is always room for improvement. As the song’s production becomes more complete, the vision for the song becomes more tangible and the melody can be readdressed in more detail. This is when a producer’s ear, a fresh ear that does not have the emotional connection that is created through creation, is instrumental in pushing the envelope. A producer should not be afraid or too lazy to ask if there could be something to improve when everyone else has project fatigue and are ready to settle with what is comfortable and familiar. When I listened to the final melody that will most likely make or break the song, I ask if it could be better.

- Could it be more memorable?
- If it is not the best it could be, what could change?
- Would more pitch movement help?
- Are there too many notes?
- Might a change in the melodies rhythm improve it?
- Is it in the right range to match the dynamic of the song section?

Throughout the tracking process, I tried to look back to the original form of the song to make sure that the moods of the new tracks were cohesive with the work tracks. If the production is done well, the song will still evoke the same emotion but more thoroughly. However, a song or album can sometimes make an unexpected creative turn. I think it is almost inevitable on at least a small level, though I do not yet have enough experience to say with certainty. In my experience in music, the ideas that were never intended are sometimes the best. And this is why it is a balancing act. This is the producer’s job. He is the father figure on the road trip to the other side of the country. He has the stability and mindset to always keep the general direction in mind. If the children want to take a few scenic routes, that should be encouraged because there are all kinds of different things to explore on the way. So long as they arrive on the other side in a place that the passengers can enjoy within a reasonable time frame and budget, the destination is a success.
Tracking Details and Techniques

Choosing Microphones

When one is deciding on instruments, amplifiers, microphones, preamps, compressors, equalizers and every other thing used in recording, one can be overwhelmed to the point of a nervous breakdown. Experimentation is always an admiral quality in the arts but it is sometimes wise to put limitations in place so that the process has a healthy rate of production. At the onset of this project, I had familiarity with a handful of instruments and recording equipment. So I chose mostly what had worked in the past and a few new pieces of recording gear to try.

The following is a detailed description of which instruments and pieces of recording equipment I chose, why I chose them, and how they were used. Some of the choices changed from song to song so there will be a description for each song. If a technique or process has already been explained, it will simply be listed and reference the original explanation.

The Drum-Set

We chose to use a custom drum kit called Masters of Maple along with Zildjan cymbals.

- Kick: 18”x22”
- Snare: 7”x14”
- Rack Tom: 7”x12”
- Floor Tom: 11”x5”
- High-Hats: K light 16
- Crash 1: K light medium thin 18”
- Crash 2: K light medium thin 19”
- Ride: Sweet ride 21”

Drum Microphones and Placement

Engineers use a variety of microphones to record kick drums. Microphones that were available to me and suited the bass drum were a Shure Beta52, an AKG D112, and an Audix D6. The AKG D112 is very popular with audio engineers, but I always felt that is lacked definition and required more EQ in a mix. The Shure Beta52 needed repairs so the Audix D6 was chosen for its high frequency clarity and low frequency impact. The microphones were place approximately 6-8” inside the bass drum. It is important to place the microphones the same distance inside the drum so that their signals are in phase and have the most low-end impact.
Possible.

For the snare drum a Shure Sm57 and Sennheiser 441 was used. The Sm57 was placed about 1" over the top rim of the drum and about 1" high above the snare head.

Figure 1.1 An Audix D6 on Kick Drum.

Figure 1.1 A Shure Sm57 on Top Snare.
On the bottom of the snare the Sennheiser 441 was pointing at the snares about 2-3" away. Since the two snare microphones were pointing in almost completely different directions at the same source, they were out of phase. To remedy this, a single band ProTools EQ was inserted with the phase flip engaged. This gives the snare a much more prominent low end.

Figure 1.2 A Sennheiser 441 on Bottom Snare.

In the past, I had used Sennheiser MD421s to record the toms. But, I was never satisfied because I felt that they had a lack of low frequency resonance and high frequency clarity. After experimenting with an AKG 414, AKG 451, and Shure Sm81, I found that the Akg 414 accented too much of the low frequency spectrum and the AKG 451 seemed overloaded by the attack of the sticks on the drumhead. The Sm81 suited the particular toms we were using best. The Sm81 placement for the rack tom and low tom was as similar as possible. They were 3-4" above the drumhead and 2-3" inside the rim of the drum.
The microphones were pointed in between the center of the drumhead and the rim of the drum so that they recorded a fairly equal balance of stick attack and low frequency resonance.

There were several options for overhead microphones. The top options were a pair of AKG C 451-EBs, AKG 414 B-XL-IIs, and Neumann KMI 83s. The AKG 414s have a larger diaphragm so they do not capture as much focus as the smaller diaphragm AKG 451s and Neumann KMI 83s (Holman). It seemed logical to assign the AKG 414s as room microphones and one of the smaller diaphragm microphones as overhead microphones. After comparing the Neumann KMI 83s and the AKG 451s, we found that the AKG 451s had a bit more sheen than the Neumann KMI 83s.

One AKG 451 was pointed straight down at the edge of the left crash (drummer's perspective) and hi-hat about 20-22" above the crash. On the ride side of the drum kit, the other AKG 451 was pointed straight down at the edge of the right crash and the ride cymbal. It was
also 20-22" above its respective cymbal.

As mentioned before, the AKG 414s were used as room microphones. There were several room microphone placements throughout the album but for most of the songs they were placed about 8' high and 4' behind the drummer. The microphones were about 6' apart to create a sort of spaced pair. To ensure that the phase between the microphones was accurate, a small bit of audio was recorded in Pro Tools and the waveform representation was looked at on a millisecond level. The microphones were in phase and the drums sounded clear, present, and powerful so this placement was used on several songs.

Figure 1.5 AKG 451s on Overheads.
Tracking Gear for Drums

The microphones went to a Neve Portico 5088 console. The console was equipped with Neve Portico 5032 microphone preamplifier/ EQs, which were used for all of the microphones. The console also had Neve Portico 5043 Duo compressors. However, they were not wired inline with the Pre Amp signal so after the signal was amplified it was routed via patch bay to the compressor inputs and finally into an Apogee Rosetta 800. The soft-limit function on the Apogee was very useful with the drums since they have such a large dynamic range. That being said, there must still be close attention on the pre-amplifiers to ensure that they are not peaking. The soft-limit simply serves as a safety net for unexpectedly loud peaks. Since the soft-limit is so transparent it was engaged at all times for all of the instruments that were routed into the Apogee. However, the Apogee only had eight inputs, so the remaining drum signals (room microphones) were routed to the Digidesign 192k interfaces. From there the signal that was being recorded into ProTools was routed back to the console so that the control room monitoring levels could be control with faders.

Gear Settings and Tracking Techniques for Drums

To capture an aggressive, punchy, rock drum sound the compressors were used fairly heavily. Gain reduction ranged from 6-10db so that the character of the compressor could be heard throughout the track. However, the rest of the compressor settings were rather mild. The ratio was 5:1 on the Sm91 kick microphone and 3:1 on the Audix D6 kick microphone, the Sm57 top snare microphone, the Sennheieser 441 bottom snare microphone, and the Sm81 rack tom microphone. The low tom Sm81 was about 4:1 along with the overheads and rooms. The attack

Figure 1.6 AKG 414s on Rooms.
was set to its slowest while the release was at its fastest. This accentuates the drums attack and tends to lengthen sustain.

In the tracking room Graham, the drummer, received a mix with a click track and the preproduction guide track, which was acoustic guitar and vocals, via aux sends on the console. The loudest track was the click so that the drummer focused on a consistent tempo, while the guide track was lower to serve as a cue in key changes of the song. He also received a low level of the drums to hear the relationship of the drums to the click. One relationship to pay particularly close attention to is the snare and click. When the snare and click fall on the same beat and one seems to disappear, it is a good indication that the timing of the drummer is very close to the timing of the click track (Holman).

When a good drummer records a drum track, it can usually be completed in two or three takes. Fortunately, Graham is an excellent drummer and once the song arrangement was settled he was able to play his part in no more than three takes. It was usually less. In some cases, we liked the overall drum take but we wanted something different in certain sections. A simple solution is a punch-in.

This process applies to the ProTools punch-in function. There are a few ways to punch-in, but I find this the easiest and most convenient. I used this punch-in technique for the entire tracking process.
1. Select Grid mode and set the grid to bars.
2. Click the cursor a bar before the section that needs to be re-tracked.
3. Hold "option" and click a bar or two before the cursor to set a pre-roll. This will allow the musician to hear some music before recording so he or she can find the tempo.
4. Hold "control" and click on the record enable button until it changes to "P" for "Punch Mode". In this recording mode, ProTools retains all the new audio as soon as playback begins even though the punch-in does not begin until later. This is great when the punch-in location is audibly noticeable. When the trim tool is selected, the new punch-in region can be dragged out to the left to the point where playback began.

4AM Production

Drum Tracking

Studio and session date: Studio H/ 295 on Jan 8th, 2010

Refer to Drum Microphones and Placement and Tracking Gear for Drums.

Bass Guitar Tracking

Studio and session date: Studio H/ 295 on April 19th, 2010

Kevin, Graham, nor I were not experienced bass players so there were a few musicians
that recorded bass tracks. On 4AM, Sam Neff played a Musicman Stingray into a SansAmp Direct Injection (D.I.) followed by a Neve Portico 5032. It is fairly common to record bass through a Direct Injection (D.I.) and the SansAmp along with the Neve pre-amplifier gave the bass enough character that I felt there was no need for microphones or a bass amplifier.

Next in the chain was an Empirical Distressor that was used fairly conservatively to add a bit more character and control the wide dynamics that a bass guitar has due to its pitch range (Holman). For tracking, the Distressor compression ratio was set at 6:1 while the attack was at 6 and the release was at 4. The maximum gain reduction was about 7db. The general intent of their compression settings is to keep the compressor working for very short times. This helps to maintain a transparent dynamic control that does not produce compression pumping. Compression pumping would be characterized as unnatural spikes and dips in volume that we would not normally associate with the natural decay of the instrument. While I aimed for the compression function to be transparent, I still looked to gain a bit of character from the Distressor. Some of the character was created from the long attack time, which allows more attack to pass through. Also, simply routing the bass signal through the Distressor alters the sound. It essentially turns the compressor into a pseudo pre-amplifier. The bass would later be compressed further after editing with the same Distressor settings and once again during mixing.

Finally, the bass signal was routed from the Distressor to a Rosetta 800 into ProTools. The soft-limit on the Rosetta 800 was engaged to avoid any unexpected peaks.

**Electric Guitar Tracking**

Studio and session date: Studio H/ 295 on April 5th and 10th, 2010

To create a comfortable environment for tracking guitar, the amplifier was placed in the control room while the guitar cabinet was in the attached tracking room. This serves as a far more efficient way of detailing guitar sounds. Instead of walking into the tracking to make changes to the amplifier setting and returning to the control room to hear the changes, I could sit in the control room, play guitar, and listen through the monitors as I turned the amplifier knobs. This is also especially helpful when recording loud rock guitar because I was able to listen to the guitar at a lower controlled volume rather than the high decibel output of the speaker cabinet. Yet another benefit is hearing the guitar in the mix, without cumbersome headphones. I am able to gain a better vision of what guitars will sound like in the final product.

There was quite a bit of time spent finding the distorted rhythm sound that would be used throughout the album. Over three recording sessions Kevin, Graham, and I searched for a sound that fit with the songs. We would record sections of songs and listen to the guitar sounds until the next recording session. Once we all felt confident with the guitar sound, we took pictures of the microphones and amplifier settings so that we could use the sound on other songs. I also taped the outline of the microphone onto the grill of the speaker cabinet so that the microphone placement could be found easily in following sessions. By the end of tracking the album, there was an assortment of guitar sounds but there were two predominant rhythm guitar sounds that we used. The set-up for this song was a Fender Telecaster to a Bogner Shiva amplifier combined with a 2x12 speaker cabinet.
A Shure 57 and a Neumann U87 was placed about two inches from the grill of the speaker cabinet. They pointed at the edges of the speaker cone so that there was a healthy balance of treble and bass frequencies. Both microphone signals were sent to independent Neve Portico 5032 preamplifiers, which were sent to an Apogee Rosetta 800 interfaced with ProTools.

To achieve and wide stereo field, the main rhythm guitar was tracked twice. Most of the
time players are naturally consistent in using the same rhythms over multiple takes. That being said, slight variations in rhythm and time (tempo) occur often. So, it is important to pay close attention when recording a double of an instrument. The closer the two tracks are in rhythm and tempo, the more impact they have. If the two tracks are played well enough they seem to turn into one take or one instrument while still having the excitement of stereo.

**Acoustic Guitars Tracking**

Studio and session dates: Studio H/ 295 on Feb 19th and Feb 27th, 2010

For acoustic guitar, Kevin played a Martin acoustic that was recorded with two Neumann U87s into two independent Neve 5032 preamplifiers followed by the Apogee Rosetta 800. One microphone was facing the guitar sound hole about eight to ten inches from the guitar strings. The other U87 faced the third fret also at eight to ten inches from the strings.

![Two Neumann U87s on a Martin Acoustic Guitar.](image)

I was looking for a dry (short room reflections) acoustic guitar sound so Kevin played in an isolation booth. This acoustic guitar-tracking set-up was the same for all but one song (Cast in Different Melodies).
The doubling technique used for electric rhythm guitar was also used for acoustic guitar.

_Vocal Tracking_

Studio and session dates: Studio H/ 295 on May 1st, 2010

The signal chain for the lead vocal was essentially the same as acoustic guitar. The U87 was routed to a Neve Portico 5032, which was followed by an Apogee Rosetta 800. A pop filter was placed in front of the microphone so that plosives did not peak the preamplifier signal.
The same punch-in technique that was used for all the instruments was used on vocals as well. However, the vocal is the most important aspect of this album so tracking was far more detailed. Even though the vocals would be pitch corrected during the editing process, we strived to record vocals that sounded great in their raw form. We also made an effort to ensure that all the lyrics were clear easily understood.

In the early stages of vocal tracking, I reserved outboard compression for post tracking since I was just starting to gain familiarity with the Empirical Labs Distressor. However, the compression settings were the same in both cases. Before compressing the vocal fades were put on each individual take to avoid pops and clicks. I also made sure that all the breaths that Kevin took were still audible and edited in a natural way. The breaths may not seem that important, but once the compression brings the volume up it is distracting when a breath has an unnatural edit. If a breath is missing, it removes part of the human quality.

In this style of music it is important that every word is heard so the Distressor was set to compress rather heavily. The ratio was set at 6:1 with the input at 5.5, attack at 6, and release at 4, and output at 8.5. With the Distressor, the knee hardens, as the ratio is set higher. In other words, the compression slope moves closer to a ninety-degree angle so that loud audio is more aggressively kept at the same volume as quieter audio. The threshold on the Distressor is effectively controlled the input. As the input is turned up, more gain reduction is applied.

I found the loudest part of the vocal track and turned the input up until the gain reduction was around fifteen to 17dB. That may seem extreme, but again, it only applies to the loudest point of the track. For the medium volume portions of the track, the gain reduction was only around 3dB. The gain reduction for the medium volume section is not entirely necessary for reducing volume since it is quieter than other section. Although, when the vocals are being
compressed, the accentuated attack gained from the Distressor gives the vocals a more aggressive sound.

Breaking Point Production

Acoustic Guitar Tracking

Studio and session dates: H/ 295 on Oct 18th, 2009

The initial tracking for Breaking Point was a bit out of the ordinary. Graham had broken his arm so drum tracking was on hold for about a month. Since the song had not changed from the preproduction track with regards to drums, I decided to move forward in tracking other instruments for the song and test the powers of Beat Detective (Beat Detective techniques will be detailed later). Fortunately, Graham still had the original pre-production ProTools session so we were able to mute the pre-production acoustic guitars and track new acoustic guitars to drums and click alone.

Refer to 4AM Acoustic Guitar Tracking for equipment details.

Drum Tracking

Studio and session dates: Studio H/ 295 on Oct 31st, 2009

Refer to Drum Microphones and Placement and Tracking Gear for Drums.

Bass Guitar Tracking

Studio and session dates: Pentavarit on May 29th, 2011

The bass was originally tracked on the same day as the bass for 4AM with the exact same equipment. However, after listening to the song over time I felt the arrangement was lacking in energy. After several attempts to persuade myself otherwise, I decided to re-track bass and guitars.

For the alternate bass track that is heard on the album, I played a Fender Jazz Bass with flat-wound sounds, which have a very smooth sound that minimizes string noise. The bass was played directly into a Pacifica A-design preamp and into Pro Tools.

Electric Guitar Tracking


Note: Electric guitar track names such as Gt1 or Cl1 always include two microphones for
one take unless otherwise specified.

The guitars were also tracked twice. Some of the original melody guitars were kept but the majority of the guitars were recorded with more aggressive playing and heavier amplifier distortion. All of the original guitars were played with a Fender Telecaster and a Bogner Shiva amplifier. The guitar signal chain was the same as the 4AM signal chain. Out of the original guitar tracks, "VsMldy1" and "VsRhy2" were kept. VsMldy1 (Gt6) played a melody on the pre-chorus and bridge while VsRhy2 (Gt7) played a repeating melody that started at an almost inaudible volume and progressed louder into the third chorus began.

For the second round of guitar tracking I switched the U87 microphone out with a Royer 121 ribbon microphone. I originally wanted to use the Royer 121 for the guitar sound but it was not available at the time.

![Figure 1.12 A Shure Sm57 and Royer 121 on a Bogner 2x12 Speaker Cabinet.](image)

The U87 has a bright and clear sound that is pleasant, but the Royer captures more of the low frequency spectrum. I continued to use the microphone tape outlines from the beginning of the project along with the same approximate distance of 2 inches from the grill. The microphone preamplifiers were also change from the Neve 5032s to Neve 1272. All of changes produced quite a different sound, but it captured more of the aggressive qualities of the amplifier that I was looking for. To add to the edgy sound on the main rhythm guitars (Gt1-Gt5) I used a Paul Reed Smith (PRS) CE 24.
Another guitar amplifier called a Badcat New Cub II 1x12 was introduced on this song. The Bogner 2x12 cabinet was used with the Badcat amplifier. The microphone setup was essentially the same as prior guitar amplifier microphone placements. Just like the other setups, a Shure Sm57 pointed at the edge of the right side of the speaker cone about 2 inches away. However, the second microphone was a ribbon microphone called a Cascade Fathead. This microphone was approximately 2 inches away and pointing at the edge of the left side of the speaker cone. The BadCat was used with a Gibson Les Paul set on the neck/rhythm pickup to record all of the lead lines (Gt12- Gt17).
Piano Tracking

Studio and session dates: Pentavarit on May 22nd, 2011

Breaking Point did not originally have piano but I thought it would be a good support to the vocal melody. I worked with pianist Chris Hennig to create the parts. At first the piano part was played on a Yamaha Motif ES 8 midi keyboard so that there was complete control of edits. The piano samples in the Motif sound great independently, but they lack higher harmonics to help it fit in a mix. The Motif tends to sound dull regardless of how much the higher frequencies are boosted. Fortunately, access to an early 1900s Howard upright piano became available. Two Audio Technica AT4047s and a rack combo Pacifica A-design equipped with two preamplifiers were used to record the piano. The two microphones were roughly equal distances from the left and right sides of the piano to achieve an equally balanced stereo image. The microphone covering the upper register was about a foot from the strings and the microphone covering the lower register had a few more inches of distance to reduce some excessive low frequencies.
**Vocal Tracking**

Studio and session dates: H/ 295 on Oct 18th, 2009

Refer to 4 AM Vocal Tracking.

**Burning All The Pages Production**

**Drum Tracking**

Studio and session dates: Studio H/ 295 on January 31st, 2010 (went unused)

Burning All The Pages was the most conservative song on the album with regards to instrumentation. The song originally had a drum track that we all liked very much. After countless attempts to write guitar parts that all turned into distractions from the vocal melody, I stripped away the drums and listened to simply acoustic guitars and the lead vocal. Suddenly I realized we were moving in the wrong direction. We decided to record piano and some background vocals, but beyond that we left the song to breathe on its own.

**Acoustic Guitars Tracking**

Studio and session dates: H/ 295 on Feb 13th and 20th, 2010

Refer to 4AM Acoustic Guitar Tracking.
**Piano Tracking**

Studio and session dates: Pentavarit on May 22nd

Refer to Breaking Point Piano Tracking.

**Vocal Tracking**

Studio and session dates: H/ 295 on May 25th, 2010

Refer to 4 AM Vocal Tracking.

**Background Vocal Tracking**

Studio and session dates: Pentavarit on August 6th, 2011

For background vocals, we recruited a singer named Kelsey Harmon. I gave her a rough mix of the song so that she could familiarize herself with the melodies and lyrics. Spending time with the rough mixes also gave her an opportunity to begin developing her own ideas for harmonies. When she came to the studio, she had a fairly solid idea of what she would sing. From there, we made a few adjustments to harmonies and fine-tuned the direction and development of the background vocals to fit the song.

The microphone used was a Miktek Cv4 sent into a Universal Audio LA-610 preamplifier. The LA-610 is also equipped with an EQ and compressor. Only the compressor was used. I enjoyed the sound of the LA-610 compressor but the activation of the compressor was more noticeable than the Distressor. To avoid an unnatural sound, the LA-610 was set to a more conservative gain reduction that did not exceed 6db.
Cast In Different Melodies Production

Cast In Different Melodies had quite a few different instruments and equipment than the other songs. It was also one of the more difficult songs to develop. The song originally came to me with drums, piano, and vocals. All three aspects changed rather drastically and several other aspects were added to the production.

Drum Tracking


The original approach to the drums was much more mellow. While the drum track did not sound completely out of character, it seemed like the vocal track was calling for a harder hitting drum track that strongly accentuated the down beats. In musician's terms, I wanted the drums to really sit in the pocket.

The recording process was essentially the same as the previous songs. The only difference was the placement of the room microphones. I changed the placement from a spaced pair to a sort of X/Y stereo technique configuration. However, there were some differences from a traditional X/Y configuration. Usually one small diaphragm condenser microphone is on top of another microphone of the same model so that they are essentially in the same place. The two microphones make an angle of 90 degrees to 135 degrees by facing in opposite directions.

The microphone technique I used was different in several ways. To start, the two microphones were AKG 414 large diaphragm condensers.

Figure 1.16 A Miktek Cv4 on Background Vocals.
The shape and size of these microphones prevent them from setting on top of each other in the same space. Instead, the microphones were about five to six inches from each other and side-by-side. They still faced opposite directions like an X/Y configuration so that the microphone on the right faced the left side of the drums and the microphone on the left faced the right side of the drums.

For other drum tracking details, refer to Drum Microphones and Placement and Tracking Gear for Drums.

_Bass Guitar Tracking_

Studio and session dates: Graham’s home studio on December 17th, 2010
This song introduced yet another bass player, Jeff Wiencrot. He played a five string Music Man Stingray into a Mark Bass amplifier and speaker cabinet with a Radial Pro Di in parallel. A Mojave Audio 201 microphone into an A Design P-1 preamplifier was used to capture the bass cabinet sound while the Radial Pro Di went into a Great River preamplifier. The Mojave Audio microphone was placed about two inches from the speaker cabinet grill and pointed at the edge of the speaker cone.

**Piano Tracking**

Studio and session dates: Pentavarit on May 22nd, 2011

Chris Hennig recorded piano for this song as well. All of the tracking techniques and equipment was identical to the setup for Breaking Point. However, there were significant changes to the piano part. Kevin originally wrote the song on piano as opposed to his usual instrument of acoustic guitar. The track was acceptable but it was obvious that Kevin was not a pianist. When Chris and I arranged the new piano part, we started with a clean slate. The chord changes remained the same, but we used different inversions and melodies. In general, we aimed for a jazz-blues sound imposed on a pop song rather than an entirely pop sound. With the new piano part, I was able to break through my lack of writing inspiration and complete the guitar arrangement.

**Electric Guitar Tracking**

Studio and session dates: Pentavarit on June 18th, 2011

The guitar sounds on this song was quite a departure from the guitar sounds on the other songs. The primary amplifier used was the Badcat New Cub II 1x12 combo that was used for the lead guitar lines on Breaking Point. The guitar I played was a Fender Stratocaster. The same microphones and preamplifiers that were used to record the Badcat on Breaking Point were used on this song.

The core of the guitar sound remained the same throughout most of the song, but several effects pedals were introduced. In the verses, there is a guitar on the left, right, and center. The left guitar is the Stratocaster set on the neck/rhythm pickup. On the right side, the Stratocaster was set to the bridge/ treble pickup. Different pickups were used because the parts were fairly similar and needed some sonically distinguishing characteristics. There was also a single note being sustained with vibrato panned in the center. This track had all the same guitar and amplifier settings as the left guitar.

When the song progresses to the chorus, there were several effects engaged. On the left side, the Stratocaster was set to the neck pickup with a Verbzilla effects pedal on its Leslie simulator preset and a Tremovibe effects pedal. The second chorus guitar panned to the right utilizes all of the same pedals and settings except that it used the bridge pickup.

For the end of the song, the Tremovibe is turned off, a Fulldrive pedal is turned on, and the Stratocaster pickup selector is set to one away from the neck pickup. The pickup choice and
Fulldrive pedal created what some would call a signature blues guitar sound.

**Acoustic Guitar Tracking**

Studio and session dates: Pentavarit on July 8th, 2011

This was the only song that I played acoustic guitar. Kevin never wrote an acoustic guitar part since the song was written on piano. So, I went ahead and arranged and recorded a simple acoustic guitar track. I played a Guild acoustic in front of a Miktek CV4 microphone—the same microphone that was used to record many of the vocals. As I did with the other songs, I put the microphone in front of the acoustic guitar’s sound hole and brought it as close to the guitar as possible without impeding on my strumming. The microphone went into a LA-610 with the compression and EQ on bypass.

**Lead Vocal Tracking**

Studio and session dates: Pentavarit on June 26th, 2011

Tracking equipment for the lead vocals was the same as the tracking equipment used for the background vocals of Burning All The Pages. A Miktek CV4 tube microphone went into a LA-610 preamplifier with the compressor on. The gain reduction did not exceed 6db.

**Background Vocal Tracking**

Studio and session dates: Pentavarit on August 6th and 7th, 2011

Kelsey Harmon was featured on background vocals again. The background vocal recording chain was identical to the lead vocal recording chain. However, creating a suitable background vocal arrangement was rather challenging. The lead vocal chorus is quick and has little melodic movement, which does not lend to many options for harmonies. After countless attempts to find a backing vocal that fit with the lead vocal, I decided to move in the direction of an almost independent melodic line that only accented the lead vocal line at certain points. To achieve this, I turned off the lead vocal and had Kelsey sing the lyrics at her own pace with her own melody. We loved the first thing that she sang. I had her record a double and moved on to the grandiose ending.

The goal for the end was to create a choir sound. Kelsey had a few bits of melody that worked well, but I was having trouble developing harmonies that sat well in the mix. We struggled with quite a few different options and finally decided to come back to it the next day. I took some time away from the song and came back to it with a simple three-part harmony. The following day we tried the three-part harmony and it fell right into place.
Lust or Love went through several transformations. In the original demo version, the song began with an acoustic guitar introduction followed by a high-energy instrumental interlude with the drums playing loudly. The song then decreased in dynamics for a verse, which was followed by a pre-chorus and then a chorus. The chorus was exited with a small instrumental section and entered into the second verse. Once again, a pre-chorus and then a chorus came. From there, the song drastically decreased in dynamics and the drums cease playing for the bridge. The song gradually built in dynamics by increasing the acoustic guitar volume and slowly reintroducing the drums with snare roll until the dynamics reached a peak and the song entered the final chorus.

When we began tracking electric guitars we felt that the song was too long and did not have a natural flow. We decided to remove the first pre-chorus, the dynamic building bridge, and the final chorus so that the song structure was acoustic introduction, instrumental interlude (with all instruments), verse, chorus, instrumental chorus-out, verse, pre-chorus, chorus, and then one more chorus with variations. We all agreed that the structure was improved but we still were not completely satisfied. In particular, we had trouble writing a melody for the instrumental interlude in the beginning of the song. After returning to the song several times over the next few months with no success, I decided to restructure the song again.

However, since I did not have a drummer or a tracking space for a drummer, I was somewhat limited and had to be creative with the recorded material I had. The acoustic introduction was removed and the high-energy instrumental interlude in the beginning of the song was replaced with a more subdued and spacious song introduction. The second chorus was then shortened to one progression, a bridge was added, and a chorus and a chorus with variations were added to the end of the song.

The drums for the new intro were copied from the verse drums and modified to fit a newly recorded piano melody. I also reversed each drum hit and piano note and pasted them in front of their corresponding hits and notes so that the attacks met each other at the same time. The bridge was manufactured from sections of the final chorus variation. The guitar line was created first and then the drums were modified so that the rhythm of the drums matched. It was necessary to pay very close attention to how the cymbal was being hit because each beat had a different accent and intensity. Each drum hit was carefully chosen so that the section did not have any hint that it was anything but a human drummer playing. Finally, we were satisfied.

Drums Tracking

Studio and session dates: Studio H and G on Jan 9th, 2010

Refer to Drum Microphones and Placement and Tracking Gear for Drums.
Acoustic Guitars Tracking

Studio and session dates: Studio H/ 295 on March 6th, 2010 and March 13th, 2010

Refer to 4Am Acoustic Guitar Tracking.

Bass Guitar Tracking

Studio and session dates: Pentavarit on June 10th, 2011

Refer to Breaking Point Bass Guitar Tracking.

Electric Guitar Tracking

Studio and session dates: Studio H/ 295 May 21\textsuperscript{st}, 22\textsuperscript{nd}, and 24\textsuperscript{th}, 2010; Pentavarit on August 27th, 2011

The tracking was essentially done in two phases. The first phase of tracking was done with a Fender Telecaster with a Bogner Shiva. As in 4AM, a Sm57 and a Neumann U87 into two Neve 5032 preamplifiers were used to record the guitar sound. Originally, all of the guitar tracks were recorded with the Telecaster and Bogner combination. A fair amount of the tracking was kept from the original sessions, but the main rhythm guitars for the choruses were re-tracked.

All of the original guitar tracks for the verses were kept and unmodified. For the first verse the main guitar part was made up of two guitars set to the clean channel with the amplifier reverb set at one forth of the maximum setting. I chose the bridge/treble pickup on the Telecaster to produce a brighter sound. Another guitar track, a clean intermittent melody panned to the left, was recorded with the Telecaster set to the neck/rhythm pickup. An additional verse guitar panned to the right was recorded with the Bogner Shiva switched to the overdrive channel and the Telecaster on the bridge/ treble pickup. In the second verse, the two main guitars were replaced with two guitar tracks that used the overdrive channel and the bridge/ treble pickup of the Telecaster. Before the second phase of guitar tracking, the second verse main guitar sound was used for the main rhythm guitar sound on the choruses.

As I mentioned before, the rhythm guitars for the choruses were re-tracked because the same sound could not be recreated without the Neumann U87. The rhythm guitars were recorded with a PRS CE 24, a Bogner Shiva amplifier, Bogner 2x12 cabinet, an Sm57 microphone, and a Royer 121 microphone. Refer to Breaking Point for the specific placement of the Sm57 and Royer 121.
*Lead Vocal Tracking*

Studio and session dates: Pentavarit on June 27th and 28th, 2011

Refer to Cast In Different Melodies Vocal Tracking.

*Palm Trees In The Mountains Production*

*Drum Tracking*

Studio and session dates: Studio H and G on Jan 11th, 2010

Refer to Drum Microphones and Placement and Tracking Gear for Drums.

*Bass Guitar Tracking*

Studio and session dates: Pentavarit on June 17th, 2011

Refer to Breaking Point Bass Guitar Tracking.

*Acoustic Guitar Tracking*

Studio and session dates: Studio H/295 on Feb 13th, 2010 and Feb 27th, 2010

Refer to 4AM Acoustic Guitar Tracking.

*Electric Guitar Tracking*

Studio and session dates: Pentavarit on July 7th, July 8th, July 9th, and July 13th. Home studio on Oct 27th, 2011

I experimented with a few different sounds in this song. The verses were kept simple with only one guitar that played harmonics. I used a Fender Stratocaster set to the bridge/treble pickup. The Badcat amplifier was used with a Shure Sm57 and Cascade Fathead. The microphones were routed to two Neve 1272 preamplifiers. For the pre-chorus two clean guitars were recorded. Panned to the left is a Fender Stratocaster with the second bridge/treble pickup selected. While I was writing the part, I was trying to create different types of chords by putting a capo at the 6th fret. In the end, the chords I played were not directly changed by the capo. But, it did create an overall brighter tone that I liked. Again, I used the Badcat amplifier and all of the equipment just mentioned. The second pre-chorus guitar also used the same equipment except a Gretsch Historic Series guitar was used instead of a Stratocaster. The Gretsch was set to the middle pickup.

When the song moves into the choruses and the bridge, the guitars switch to an
overdriven sound. At first the chorus and bridge guitars were recorded with a Fender Stratocaster and the Badcat amplifier. But once I began the mix I noticed that the guitar was out of tune at certain points. I hoped to use Melodyne to adjust the out of tune chords, but I was unsatisfied with the way the sound of the guitar was affected. I finally decided that the best option was to re-record the chorus and bridge guitars. Unfortunately, the tracking space previously used was unavailable. I decided to make an attempt to record with the equipment I had available at home. I used a PRS CE24 with my Bogner Shiva amplifier. I placed a Shure Sm57 2 inches away from the speaker cabinet grill and pointed it at the edge of the speaker cone (just as I had done with the other microphone placements). I was concerned that my Digi 002 preamplifier would be inadequate, but the results were surprisingly good.

There was also a melodic arpeggio throughout the chorus. Once again a Fender Stratocaster was used with the bridge/treble pickup selected and a capo at the 6th fret. Just as the bridge, a Badcat amplifier was recorded with a Shure Sm57 and a Cascade Fathead.

**Lead Vocal Tracking**

Studio and session dates: Pentavarit on June 25th, 2011

Refer to Cast In Different Melodies Lead Vocal tracking.

**Perfect Remedy Production**

**Drum Tracking**

Studio and session dates: Studio H/ 295 on Sept 6th, 2009

Refer to Drum Microphones and Placement and Tracking Gear for Drums.

**Bass Guitar Tracking**

Studio and session dates: Studio H/ 295 on Sept 6th, 2009

The bass guitar tracking for this song was very different than the rest of the album. Chris Cash played a Warwick bass with an SWR amplifier and a 4x12 Ampeg speaker cabinet. The bass signal was routed out of the SWR amplifier into a Radial Pro DI and then a Neve 5032. A Sennheisser MD421, a Neumann U87, and an Akg D112 were used to capture the speaker cabinet sound. All three microphones were an inch away from the cabinet grill and pointing at different sections of the speaker cone edge. However, only the Sennheisser MD421 was used in the mix.
Acoustic Guitars Tracking

Studio and session dates: Studio J and G on Sept 20th, 2009

The same Martin acoustic that was used on the majority of the songs was used on this song as well. An AKG 414 was placed about six inches in front of the sound hole right were the guitars neck ends. The signal went directly to a Sony DMX R100 console/ preamplifier and then into Pro Tools. Just as the other songs, the acoustic guitar track was recorded twice.

Electric Guitar Tracking

Studio and session dates: H/295 on Sept 25th, 2009; Pentavarit on July 14th, 2011

This song was recorded before I had access to the Bogner Shiva or the Badcat, so we used Kevin's Fender Bassman to record all of the electric guitars. We still used an Sm57 microphone but we felt that an AKG 414 microphone was more suited for the Fender Bassman than a Neumann U87. For the verses and pre-choruses, we used a Fender Telecaster set on the middle pickup. On the amplifier we kept the gain low to produce a clean and bright sound. On the choruses the Telecaster pickup was switched to the bridge/ treble pickup and an OCD overdrive pedal was turned on.

Later on in the project I decided to address an amp hum that was most notable in the bridge. The same Telecaster and Fender Bassman amplifier was not available so a similar sound was created with a Fender Stratocaster set on the Neck/Rhythm pickup and a Badcat amplifier. Fortunately, there was only overdriven guitar sounds before and after the bridge, so the difference in the clean guitar sound was not very noticeable or much of an issue.
**Lead Vocals Tracking**

Studio and session dates: Studio H/ 295 on Sept 26th, 2009

Refer to 4Am Vocal Tracking.

**Take Your Pride And Dance Production**

**Drum Tracking**

Studio and session dates: Studio H/ 295 on December 14th, 2009

Refer to Drum Microphones and Placement and Tracking Gear for Drums.

**Percussion Tracking**

Studio and session dates: Studio H/295 on May 19th, 2010

Various sounds were recorded to create a percussion section during the bridge. Bt1.A and Bt1.B was a guitar scratch recorded with a PRS CE-24 and a Bogner Shiva, which was then pitched down an octave. Then the sound was copied and pasted on the grid in Pro Tools to create a beat pattern. Several other sounds were recorded and incorporated into the beat pattern. Bt.D112 was an Akg two feet away from Graham and Kevin sitting in chairs and stopping on a hollow stair platform. Chair1 and chair 2 was Graham and Kevin sitting in chairs while jumping up and down with them. Chair1 was recorded with a Neumann U87 close and Chair 2 was recorded with a Neumann U87 about five feet away. A Neumman U87 was placed two feet away from Graham hitting a pallet with drumsticks to record Bt.87. A Neumman U87 was also used to record a box of microphone clips and microphone stand parts being dropped on the ground. The box sound was labeled Bt2.87. Wood1 was a box of parts and a pallet being jumped on and broken in reverse. The sounds were recorded with a Neumann U87 three feet away. Wood2 was another pallet being broken, which was recorded with a U87.

**Bass Tracking**

Studio and session dates: Graham’s home studio on Dec 17th, 2010

Refer to Cast In Different Melodies Bass Tracking.

**Acoustic Guitar Tracking**

Studio and session dates: Studio H/295 on January 9th and Feb 13th, 2010

Refer to 4Am Acoustic Guitar Tracking.
Electric Guitar Tracking

Studio and session dates: Studio H/295 on May 17th, 2010

The general setup for electric guitar was a Fender Telecaster and a PRS CE 24 with a Bogner Shiva amplifier and Bogner 2x12 speaker cabinet. For this song, a Sm57 and Neumann U87 into two Neve 5032 preamplifiers were used to record the guitar. Refer to 4AM Electric Guitar tracking for microphone placement specifics.

The volume swell sound in the verses was created with the PRS CE 24 into a Line 6 Delay modular. To create more sustain, the Bogner's gain control was turned up extremely high. A second verse guitar track that was primarily harmonics with a clean tone was played with the Fender Telecaster set to the bridge/treble pickup. Another verse guitar that serves as the main rhythm was played with the Fender Telecaster on the middle pickup. A very small amount of gain was applied on the Bogner to give the guitar a bit more attack, but still remain relatively clean sounding. The track continues into the first two choruses with the same sound.

The first two choruses also have a lead melody guitar that was played on the Fender Telecaster with the bridge/treble pickup selected. This track also includes a simple melody that was played with the same settings. However, it has a slightly different sound because it was played by hammering on and off with the right hand.

As the song moves into the choruses there is a guitar that plays a sort of blues lick to help the transition develop. On that same track, there is an arpeggiated melody that helps the song move from the choruses back to the verses and a repetitive picking melody that gradually increases in volume to help the bridge dynamics build. This track was played with the Fender Telecaster set to the middle pickup with the same amplifier setting as the main verse rhythm guitar.

Two additional bridge guitars that play harmonies with each other were recorded with the Fender Telecaster set to the bridge/treble pickup. I then used the audio suite reverse plug-in on each individual note and arranged the notes in the order I desired.

The end of the song has rather different guitar sounds than the rest of the song. The main rhythm is the PRS CE24 panned to the left and the Fender Telecaster panned to the right. The gain was turned up to around 10 o'clock. A slide guitar is also introduced for the final chorus. I used the PRS CE24 with the amplifier reverb turned on and the amplifier gain turned up about half way. Another lead line that matches the slide guitar melody at an octave higher was played on the Fender Telecaster set to the bridge/treble pickup and the same amount of amplifier gain. I also used the Line 6 Delay modeler to create a long repeating delay. Finally, the same lead melody from the first two choruses was played again but with the PRS CE24 and the amplifier gain the same as the other final chorus guitar melodies.
**Vocal Tracking**

Studio and session dates: Studio H/295 on May 26th, 2010

Refer to 4Am Vocal Tracking.

**The Fall Production**

**Drum Tracking**

Studio and session dates: Studio H/295 on January 31st, 2010 and February 6th, 2010

Drum tracking was split between two sessions because the majority of the song up until the end was played with brushes. Graham used regular sticks for the end of the song. The brush portion of the song was tracked in studio H with the same recording chains as the drum tracking on 4AM. The end of the song was tracked in studio G. Refer to Cast In Different Melodies Drum Tracking for different room microphone placement.

**Bass Guitar Tracking**

Studio and session dates: Pentavarit on June 3rd, 2011

Refer to Breaking Point Bass Tracking.

**Acoustic Guitar Tracking**

Studio and session dates: Studio H/295 on Feb 20th, 2010 and March 6th, 2010

Refer to 4AM Acoustic Guitar Tracking.

**Electric Guitar Tracking**

Studio and session dates: Studio H/295 on May 7th, 2010; March 26th, 2010; April 3rd, 2010; April 12th, 2010, and August 27th, 2011

Most of the guitars had a clean sound until the final portion of the song where the drums switched to regular sticks. For the verses, the first clean guitar was played with a Fender Telecaster on the middle pickup into a Line 6 Delay Modeler. The delay modeler was set to a volume swell setting. Again, the Bogner Shiva amplifier and Bogner 2x12 speaker cabinet was used with a Shure 57, a Neumann U87, and two Neve 5032 preamplifiers. Refer to 4AM electric guitar tracking for specific microphone placement. The other clean guitar in the verses had basically the same guitar sound as the first, except the Line 6 Delay Modeler was changed from the volume swell to a short delay to give the guitar a bit of a room like sound. The delay is rather short so it tended to sound somewhat like reverb.
In the first two choruses, the guitar panned to the left is the same exact sound as the second verse guitar as well as the two guitars panned hard left and right in the bridge following the second chorus. Following the first two choruses, there is an instrumental chorus that has two overdriven melodic lead guitars panned hard left and right. These lead parts were recorded with a Fender Telecaster on the rhythm/neck pickup with the same amplifier and microphone setup as the rest of the song’s guitars. The only difference was that the amplifier was switched to the overdrive channel.

For the final choruses after the bridge, I kept the amplifier and microphones the same but used a PRS instead of a Telecaster to get a more overdriven sound. I recorded two of the same rhythm tracks so that I could pan them hard left and right and achieve a large and powerful stereo sound. However, once I reached the mix stage I realized that the guitars could have more impact and low-end. I kept the rhythms that I had but recorded two additional rhythms. The amplifier setting and microphone placement was the same but I used a Royer 121 ribbon microphone instead of a Neumann U87 to capture more low-end. I also used Neve 1272 preamplifiers rather than the Neve 5032.

**Piano Tracking**

Studio and session dates: Pentavarit on May 22nd, 2011

Refer to Breaking Point Piano Tracking.

**Lead Vocals Tracking**

Studio and session dates: Studio H/295 on May 27th, 2010

Refer to 4Am Lead Vocal Tracking.

**Background Vocal Tracking**

Studio and session dates: Pentavarit on June 12th, 2011

Mike Williamson was featured as a background vocalist on this song. Refer to Burning All The Pages for background vocal tracking specifics. Since the beginning of production for this song, I had envisioned a choir singing a melody I stumbled on. I also wanted to hear some harmonies during the verses. Fortunately, Mike was very experienced with creating background vocals. We played through the song a few times and Mike presented different ideas for the verses until we had something that we were excited about. He also created a few harmonies for the choir section that created a denser and more believable choir sound. Once we had the verse and choir vocals, we felt that the continuity of the song could benefit by creating counter melodies for the chorus and chorus-outs. For every background melody or harmony, we recorded a double so that the tracks could be panned hard left and right and give the lead vocals and background vocals their own space.
Drum Sampling Session

To capture extremely clean recordings of individual drums without bleed or resonance from other drums, two days were spent sampling Graham's Master Of Maple drum set along with his Drum Workshop (DW) drum set.

DW Drum sizes:
Kick: 18X22
Snare: 5X14
Rack Tom: 8X12
Floor Tom: 16x16

New drumheads were purchased specifically for the sampling session since they are expensive and loose resonance and high-end through the process of tracking an album. Sampling sessions also create an opportunity to try different drum tunings and amounts of resonances for each drum. For the most part, the same microphones, preamplifiers, and compressors were used as on the drum tracking sessions with the exception of a few additional microphones on toms.

Kick

For the DW kick, the front drumhead was removed so the drum had a more open and resonant sound. All microphones, microphone placements, and equipment used apply to both the Master of Maple and DW drum-set.

Refer to 4AM Drum Tracking for specific microphone placement and compressor settings.

Snare

Just as the kick drum, the snare was sampled with the same microphones, microphone placements, preamplifiers, and compressors that were used during the albums drum tracking sessions.

Refer to 4AM Drum Tracking for specific microphone placement and compressor settings.

Toms

As I said in the drum tracking description, I experimented with an AKG 414, AKG 451, and Shure Sm81 on toms. For the purpose of this record the Sm81 was used. But for the drum sampling a variety of tom sounds were recorded for future projects. The AKG 414 was underneath the tom about three to four inches from the drumhead. It was pointed at the outer portion of the drumhead. On the top of the tom the AKG 451 and Shure Sm81 were placed side-
by-side (parallel) and about 2-3 inches from the drumhead. They were pointed at the outer portion of the drumhead as well.

To achieve different resonances, the toms were recorded without various amounts of dampening. First they were recorded without any sort of dampening tools, then with moon gel at the edges of the drumhead, and finally with toilet paper taped to the bottom drumheads. Initially, the toilet paper was stretched across the entire diameter of the drum and taped to the rims so that toilet paper lightly touched the drumhead.

It was very similar to how the snare on a snare drum is placed. However, as the drum was hit more, the toilet began to break. The sound we enjoyed most was right before the toilet paper broke off completely. The microphones, microphone placements, and all equipment apply to both the rack tom and floor on both the Master of Maple drum-set and DW drum-set.

Figure 1.20 Shure Sm81, AKG 451, and AKG 414 on Sample Toms.
Refer to 4AM drum tracking for specific recording chains and settings.

Figure 1.21 Toilet Paper on Tom Bottom.
CHAPTER 2 EDITING

Beat Detective

Beat Detective is a powerful time editing tool that was used on every song. Since drums are the cornerstones of rhythm in this style of music, I edited the drums with Beat Detective before recording any other instruments for the given song. It can also be utilized on guitars, piano, bass guitar, and anything else that has a strong transient and attack. However, the user must pay close attention to the edits Beat Detective is creating because it is not always accurate and can create artifacts. Also, when beginning editing with Beat Detective the user must also decide how accurate or lenient the instruments timing should be. In some styles of music listeners have come to expect absolutely perfect timing and in other styles listeners expect to hear instruments playing a little before or after the beat. For this project, the drums were edited as close to the beat as possible, which was not far from Graham's original playing since he has remarkable timing. Still, in this style of music, timing perfection with regards to the drum is expected. Bass guitar also has high expectations for timing but because it is harder to avoid editing artifacts there was much more time spent punching in parts and ensuring that timing was accurate when it was recorded. The same approach was taken with guitar.

Note: Beat detective can be used to extract a tempo if the track was not originally played to a click. However, editing becomes more complicated so I find it easiest to record to the Pro Tools internal click so that the instruments’ rhythm is reasonably in sync with the Pro Tools grid.

**Beat Detective for Drums**

Before any editing is done, make a group for the drums so that all editing actions are performed on each individual drum track. Also make sure that Grid mode is selected. Grid mode will be used until the end of the editing process when fades are applied. Next create a duplicate playlist so that the drum tracks are saved in their original state and all editing can be done to the copy. Add ".edit" or something similar to the playlist name to signify that it will be the edited playlist. Having this backup puts my mind at ease since I can always return to the original drum track if a mistake is made. With that confidence I find I have less of an urge to over check my work and I am able to edit with more speed and efficiency. I make the edit, listen to it once, and then move on to the next edit. Once the entire track has been edited I listen to it from start to finish. Again, if I find a major edit error I can return to the original playlist, make a copy of the errant section, paste it to the edit playlist, and edit it correctly.

When beginning editing, take note of where the first down beat of the recorded track is in relationship to the grid. In Pro Tools, the darkest grid lines are the first beats of a bar. Beat Detective will work best if the drummer’s downbeats/bars are in line with the Pro Tools grid downbeats/bars. When the drummer's downbeats are out of line with Pro Tools downbeats, Beat
Detective will see downbeats as upbeat, or vise-versa, and try to move edits to the beats that it assumes are correct. So if the drummer did not begin playing at the beginning of the bar turn on the tab-to-transient option and click on the waveform representation so that cursor is before the first beat that the drum plays. Hit tab and the cursor will move to the very beginning of the first transient. Then hit Apple + E to cut the region. Be sure that grid mode is selected and all of the drum tracks are in a group. Then drag the drums to the beginning of a bar. For consistency, the same thing can be done to the original drum track but any further editing should be avoided. To avoid any confusion and eliminate the steps just explained, I have the drummer take note of the beginning of the bar during tracking. This is not always obvious because some drummers prefer to play to a click with no accents, which is what Graham did. To remedy this, I simply made sure that the click began it's count-off at the beginning of a bar.

Now that any grid alignment issues have been addressed bring up the Beat Detective window with the key command Apple + 8 (on the numeric keypad). If a numeric keypad is unavailable the Beat Detective window can be found under Event. There are several detection and time correction options in Beat Detective so it is important to select the optimal options for the job at hand. For drums I use most of the default settings.

Next to "Operation" the "Audio" option should be selected. Beneath Operation, click on Region Separation to view the detection settings. Next to "Detection" make sure that "Normal" is chosen. The other Detection option, "Collection", is meant to account for the natural time differences that each drum microphone with have. For instance, the overhead microphones will
record the kick drum later than the kick drum microphone. In Collection mode each drum track can be analyzed, cut and moved according to the beginning of the their individual transient beginnings. I prefer to keep the time differences between microphones as they were recorded because it is in part what gives the drums a sense of space.

For "Analysis" the options are Low Emphasis, High Emphasis, and Enhanced Resolution. "Low Emphasis" is best for low frequency and/or harmonic material like bass guitar or kick drum. "High Emphasis" is for high frequency and/or inharmonic material like cymbals. I use Enhanced Resolution since it analyzes both types of material. Under Selection there are several note divisions that can be chosen. If a quarter note is selected, Beat Detective will search and mark transients that are closest to the quarter note in relation to the grid. In the case of drums, I usually edit where the kick and snare plays. Most of the time, the kick and snare plays eighth notes so I do most of the detection with the eighth note selected. When a drum fill comes along I often change the note division to 16th notes since drum fills are so commonly 16th notes.

After the appropriate note division is selected, click "Capture Selection" and then click "Analyze". Increase the sensitivity until markers appear on the waveform. As I said before, I primarily edit where the kick and snare falls so I increase the sensitivity until all of the kick and snare hits have markers. Sometimes Beat Detective will miss a transient and not place a marker even at 100% sensitivity. If this occurs, select the hand tool with the key command Apple + 4 (top of the main keyboard) and click where a marker needs to be. Zoom in with Apple + ] to the sample/single line waveform to place the marker specifically at the beginning of the transient. If an unwanted marker appears, hold option and click on it until it disappears.

Beat Detective can sometimes analyze a transient as an incorrect beat. When the marker is darker, it has been analyzed as a downbeat and Beat Detective will move it to the closest downbeat when conformed. When a marker is lighter it has been analyzed as an up beat and will be moved to the closest upbeat. To change the markers beat destination, double click on it. The "Identify Trigger" dialog will come up. Type in the desired beat destination and click OK. When the drum downbeats correspond correctly with the grid downbeats, like mentioned before, the amount of marker errors is greatly reduced. Still, occasional errors are bound to occur. Once the markers are on the appropriate transients and have the correct beat destination, click separate. The region will be cut at each marker location.
Next, change the operation to Region Conform. On the top right there is two ways to conform. If "Groove" is chosen there are preset rhythms that the audio at hand can be conformed to. A custom groove can also be created. For instance, if a song needs a swing or blues feel that does not fall directly on the beat, then a template groove can be created that conforms audio a little ahead or behind the grid. For this project, I set conform to standard. With this setting there is also an option to adjust the swing of the audio being edited. However, it does not conform to a preset rhythm template. Strength effects how accurate the transient will be in regards to the grid. At 100%, the transient will be moved directly on the grid. As I had mention at the beginning of the Beat Detective explanation, I edited the drums as close to the grid as possible (100% strength). However, in some cases Beat Detective will make slight errors and place a transient a few milliseconds ahead or behind the grid. If it is not a blatant rhythmic error I will leave it since it will not make a noticeable difference. In some cases I might argue that those subtle differences help to maintain the human feel that separates live drums from drum machines. If the conform operation moves a transient to a beat that is obviously wrong, undo the action (Apple + z), switch back to the region separation operation, check that the beat markers have the correct beat assignment, and then repeat the conform process.

Figure 2.1 Beat Detective Region Separation Function.
For each four to eight bars I use the Edit Smoothing function. There are two options for smoothing. Fill Gaps simply pulls the ends of regions out to fill spaces between regions that have occurred due to editing. The Fill and Crossfade function does the same as Fill Gaps but it also crossfades where the edit was made. I always use Fill Gaps because the Fill and Crossfade function almost always has errors.
The usual error is the transient occurring twice since the space between transients is being lengthened. Instead, I use a short process that ensures that all of the edits are made cleanly.

I begin by setting Nudge at the top middle of the edit window to 10 milliseconds. Again, be sure that the drums or any set of tracks that are being edited together are in a group. At this point, the Grid mode can be changed to Slip mode since lengthening of the regions will need to fall outside of the grid. Once I have the correct settings, I hit return to move to the very beginning of the session and then hit Control + Tab. This key command will select the first region and each following region every time it is hit. After each Control + Tab, I hit Option +(-) on the numeric keypad. This key command pulls the front of the region out to the left however much the nudge is set to. Be careful not to miss the option key alone and only hit the (-) key, because this will move the entire region to the left as opposed to increasing the length of the region. I usually hit Option + (-) two times to make sure that the unneeded transient copy is covered. Even though there can be thousands of edits/regions when editing drums, with a little practice this nudge procedure can be completed in five to ten minutes. I then select all of the edited tracks and change the editing smoothing to Fill And Crossfade. The length of the region will not be changed but fades will be placed on every edit that was made so that there are not any pops or clicks. I listen to the entire drum track and if there are not any errors I consolidate with
Shift + Option + 3. To free up computer processing resources, click on the down arrow at the top of the regions bin (clips bin is ProTools 10). Drag down to "Select" and then "Unused Audio Except Whole Files". Then hit Shift + Apple + b to remove the unused files from the session. I do not remove all unused files (whole files/ original unedited files) until I have completed all of the edited and prepare for mix.

**Melodyne For Vocals**

Melodyne is a powerful plugin that has several editing parameters. Its primary function is pitch correction. But it can also be used to alter the length of audio and do similar things to Beat Detective like change the timing of audio. However, Melodyne time editing has a very different approach than Beat Detective. It can also be used to create harmonies from a single note melody. I only used Melodyne for vocal pitch correction.

I start by inserting the Melodyne plugin on the vocal track to be tuned. I then select the section in the ProTools edit window that I will be working with. I typically select the entire length of the track. In Melodyne, hit the transfer but and then playback the audio. Melodyne will record all of the audio playing back on the given track. Once playback is stopped, Melodyne will take a moment to analyze the material. The audio can now be manipulated within Melodyne. But first set the song's key in Melodyne by clicking on the down arrow in the upper left corner of the plugin. Navigate to Pitch Grid and then Select Scale to choose the key appropriate for the song. Also under Pitch Grid I usually choose Scale Snap. If I am working with a song that has quite a few accidentals (notes outside of the key), I choose Semitone Snap.

Next, choose the arrow tool within the Melodyne plugin, and then drag over a small section to select it and correct the pitch. Melodyne correction accuracy works the same on small sections as large sections. Still, I like to work in small melodic phrases so I can have a fresh look at the original Melodyne pitch interpretations prior to correction. This helps making decisions on how to correct a vocal so that it is near perfect as possible but still human-sounding and conveying the original idea of the performer.

Now that a section is selected for tuning, hit correct pitch in the upper right corner of Melodyne and choose a percentage. At 0% the original note is not moved and at 100% the note is moved to the center of the given pitch without any cent deviation. However, one of the great features of Melodyne is that it does not flatten performed note into a sine tone type sound. Instead, Melodyne searches for the overall intended note and maintains all of the vibratos and different inflections that are natural when singing in something they call a "blob". Since all of the things that make a note sound human move together, there is more allowance for very strict pitch correction without as much risk of sounding robotic or too artificial. But, just like Beat Detective, it is important to understand the expectations of the project at hand and choose an appropriate editing direction. Since this project is most closely associated with pop rock, vocals were corrected at 100% in most cases. If artifacts appeared at 100%, a lower pitch correction percentage was used.

While Melodyne is rather simple and this is the bulk of the vocal tuning process, there are some pitfalls to listen for. When tuning vocals, much of the tuning process is spent deciding
whether or not Melodyne has corrected a note to the correct pitch. Another large portion of time is spent separating two notes that Melodyne has detected as one note. Fortunately, these things are relatively simple to fix.

The simplest and most common error is when Melodyne moves a note to an incorrect pitch. When this happens, select the arrow and simply drag it to the correct pitch. In most cases, the note is only a semitone away from the correct pitch. What makes this difficult is deciding what the singer originally intended to sing. I find it helpful to have a guitar or piano nearby to determine the melody.

Melodyne also occasionally misinterprets a note due to different harmonics in the voice and assigns it to an octave to high or too low. This will become apparent when the note is tuned and high or low pitch octaves result. It is usually fairly obvious. To assign the note to the correct octave, undo the pitch correction that resulted in the artifact and choose the arrow with (+/-) near it. This will show an alternate view that displays possible octaves that the note belongs in. Just drag that blob into the correct octave. This will not tune the note. Instead, it will be re-evaluated and processed differently when tuned. Once the note is assigned to the appropriate octave, correct the note's pitch as usual.

Sometimes Melodyne will interpret two adjacent notes as one note. Select the Note Separation tool and double click on the blob when the two notes most notably divide. This does not necessarily have to be precise because once the two notes are divided, the separation point can still be adjusted. Listen to the separated notes and decide if the edit sounds natural. If not, use the note separation tool to clip on the separation line that was created when dividing the two notes and drag it to the left or right. This usually takes a bit of experimentation to find the separation point that reflects the performer's original intent. Again, once the error has been corrected do the usual pitch correction.

One other common problem that Melodyne has is analyzing a singer’s breath as a note. If a breath is pitch corrected it tends to have high pitch harmonics accented making it harsh. There are two simple ways to avoid this. One way is to not select breathes when highlighting a section of notes for pitch correction so that the breath goes unaffected. If a breath is accidentally affected and it is not noticed until later, copy and paste the breath from the original vocal track.
CHAPTER 3 MIXING

Mixing Philosophy

A mix should be treated as a continuation of the entire record production process. Just as a producer should search to find what an artist is trying to express, a mixer should listen to a song to find the theme or core of the song. Many mixing engineers share a similar view. Joe Chicarelli argues that, “It’s a matter of finding out what the center of the song is or what makes the song tick (Owsinski, The Mixing Engineer's Handbook ).” Similarly, Allen Sides, says, “I really want to understand what they want so I can make that a part of the picture that I draw (Owsinski, The Mixing Engineer's Handbook ).” There will inevitably be characteristics of someone’s style since everyone has a unique perspective, but the objective and mindset should be to listen for the intention of the artist/ producer rather than try to superimpose a personal agenda.

I have found that because it is part of a bigger picture it makes sense to follow the major musical elements that were used in the beginning of the production. However, even though mixing can be viewed as a continuation of producing philosophies and utilize musical elements, it is still a unique process and has its own purpose. During the pre-production and producing process the musical elements can be used to decide the best structure of parts to convey a particular message. A mixer can use those same musical elements to refine the message in the best possible way.

The refinement of the message could also be described by comparing a mixer to an actor just as a producer and/ or artist could be compared to a playwright. In a play all of the words are already decided, but the way each word is presented individually and in relation to each other effects how the message is received. The actor's job is to look for the story that the writer was trying to tell. A bad actor misses or ignores the story and leaves the audience in confusion as to what the writer was trying to say. The best actor tells the story so that the audience understands and is affected as the writer intended.

There is also a deconstruction and reconstruction process that takes place in a mix. This refining process occurs all around us. When a car is restored it is best to dismantle it entirely, remove rust and dirt, and reassemble it with slight adjustments to make it operate at it's best. When we lift weights our muscles are broken down and torn apart for the sole purpose of rebuilding a stronger and more efficient muscle. In the same way, a mix begins by breaking a song down into its individual parts. Before a mix is complete, the individual parts may collide and compete with each other so that the message is foggy and unclear. Each aspect of the mix must be adjusted and rebuilt so all the pieces work together to clearly speak the message.
Focusing On Music Elements In Mixing

A mix can be examined through the lens of the major musical elements, broken down, and reassembled in an exceptional form.

Rhythm

Just as rhythm builds the foundation for most songs in popular western music, it also often builds the foundation of the mix. Whichever instrument plays the main rhythmic idea, usually percussion of some sort, should be one of the main aspects of the mix. Harmonic structures will move according to the rhythmic motif just like the phrasing of a melody will be based off of that same motif. If the instrument plays consistent down beats, then that instrument should be manipulated in a way to make the down beats as strong as possible. However, since percussion is typically very loud and present, a space appropriate to the mix must be found so it does not dominant over melodic ideas.

In music that does not usually have percussion or consistent strong down beats, like music from the classical or romantic period, special care must be taken to not diminish the rhythmic aspects. Overuse of effects and an unbalanced frequency spectrum can smear rhythmic ideas.

Harmony

Since most popular western music is homophonic, almost all tracks in a mix, excluding the primary melody and the rhythmic support (usually percussion) would be considered as harmony(Western Michigan University). In a typical rock band, the bass guitar, electric or acoustic guitars, and keyboards would fall into the harmonic support category. When they are added to a mix, they should feel as if they are intertwined with the rhythm section. For a dance mix, the harmonic support might be considered as secondary. In rock, country, or general pop, it might take precedence over the rhythm section. Whatever the case may be, instruments play harmony should be considered as support to the primary melody.

Timbre

At the mixing stage, the major characteristics of timbres have already been decided. Aside from severe manipulation, a Fender Stratocaster will always sound like a Fender Stratocaster. However, a large part of mixing is making slight alterations to each individual timbre so that all of them work together. The majority of microphones do not capture sounds as bright as many mixers prefer. Consequently, there is quite a bit of work dedicated to brightening instruments so that they meet the standard that the public has come to expect.

Texture

Because most popular western music is essentially homophonic, it is sometimes
necessary to downplay certain aspects. It is somewhat common that each instrument in a band plays variations of the same idea. If too many ideas standout, the focus of the mix can be unclear. It is best to find the primary melody, usually the lead vocalist, and make that the focus. The rhythmic/percussion section should be treated as another texture. Out of the other instruments that create a harmonic texture, one might be featured to serve as a sort of secondary melody.

**Structure**

The structure of a song is highly dependent on changes in chord progression. If the chord progression is the same throughout the song, then introducing instruments, removing instruments, or changing dynamics, can define the song. Either way, since the structure is already decided, the mixer should focus on giving each section it's own special characteristics. Creating subtle contrast between sections with changes in space, focus on different instruments, and dynamics help develop the direction of the song. All of the other elements should be used to enhance the structure that is already there.

**Dynamic**

All musical elements have an effect on each other. But structure and dynamics are particularly connected. Dynamics is one of the key factors to defining of section of a song from another. It is also essential to building the overall climax of the song. In popular music a typical structure is verse, chorus, verse, chorus, bridge, and chorus. A common approach would be increasing the dynamics on the first two choruses, dip down on the verse and bridge, and reach the highest dynamic range in the final course. When mixing, the structure should be quickly identified because such a large portion of mixing is bringing out the contrasting dynamics that were built in the song during tracking by choosing each instrument for the appropriate section.

**Melody**

The primary melody is fairly simple to identify in popular western music. As I mentioned before, it is most often the lead vocalist. But it might be played by a guitar, keyboard, or violin. Regardless of the style of music or instrument, the melody should always be at the forefront of the mix. Every phrase and inflection of a melody should be clear and featured above all else. The melody should be given its own space and unique sound. If all other aspects of a mix fail, a well-treated melody will keep the mix alive.

**Forms of Audio Manipulation**

During my internship with mixing engineer Craig Alvin, he shared a simplified way of approaching a mix.

The musical elements will define each aspect of a song and help to identify issues that need to be addressed in each aspect. How those aspects can be manipulated to better work
together can be split into three groups. One approach to a mix would be to look at each musical element of a mix and think about how a form of manipulation would improve the element. When one type of manipulation does not achieve a result that entirely fulfills the need of the mix, experimentation with another type of manipulation can be done to see if that renders the desired results.

Note: Since there is an abundance of material explaining audio effects/ processing the below will not go into great explanation of how the effects/ processing work. Instead, this is simply an explanation of how each type of effect falls into a group of audio manipulation.

**Balance (Loudness)**

**Loudness/ Balance.**

The most straightforward way to affect the power of a track is by increasing or decreasing the volume with the fader. However, to give a mix more life and interest the tracks will often need some variation in volume. Things like a drum set and bass guitar can often stay static in volume since they are so dynamic, but instruments like electric guitar are naturally compressed by the guitar amplifier and need help creating dynamics. As the song moves to a louder section like a chorus, the volume for the guitar can be boosted. When these volume changes are built into a variety of tracks, the songs dynamic range becomes larger and much more exciting.

**Compression.**

As I mentioned above, some instruments are extremely dynamic and need their levels controlled. But compression must be use appropriately. Unlike volume/ volume automation that keeps the audio in the original form, compression alters the characteristics and timbre of a sound.

**Frequency**

**Equalization.**

EQ can be used to boost or cut specific frequencies of a sound and essentially change the timbre. To attack specific troublesome frequencies in a sound, very small narrow EQs can be used to significantly reduce the volume of a frequency. Although, the most effective way to change the overall timbre of a sound is use wide EQs. Using wide EQs can give the impression that the instrument itself has changed in someway.

**Amplifier Simulators.**

Amplifier Simulators like AmpFarm and SansAmp are generally intended for use on guitars or bass guitars. Just as an actual amplifier, they tend to have broadband EQs to alter the instrument being amplified. But amp simulators also add harmonics to a sound to emulate the amplifiers they are modeled after. Some do better than others. However, they do not have to be solely used for emulation. Instead, their harmonic adding properties can be used to improve a sound. Increasing the gain on an amp simulator has a particularly profound effect since more gain produces distortion, which introduces more harmonics.
**Time**

**Reverb.**
Reverb will create the feeling that a sound source has been recorded in a particular room. In general, a sound becomes more distant as more reverb is introduced. Reverb achieves this effect by introducing distant reflections that cannot be heard individually (Holman). Reverbs can also include pre-delays that usually are not heard as discrete reflections (Holman). A reverb sounds best when it is timed to the tempo of the song, particularly the pre-delays (Owsinski, The Mixing Engineer's Handbook).

**Delays.**
Delays work by using discrete delays. Like reverb, delays work best when they are timed to the tempo of the song (Owsinski, The Mixing Engineer's Handbook). They tend to keep a sound more direct and forward. But, delay also smoothes the sound out, cures some harshness, and adds some depth and wideness.

**Side-effects of Audio Manipulation**

Some effects will have a side effect that could fall into another group of manipulation.

**Volume.**
According to the loudness curve, low frequencies in a sound will be heard less as the volume decreases. So if brighter sounds overshadow low frequency sounds, it may be difficult to turn a mix up to a listening level that has sufficient low end without the brighter sounds becoming bothersome.

**Compression.**
When a compressor controls an instrument’s dynamic range, other characteristics are inevitably changed. For instance, changing the attack and release controls on a compressor can affect an instrument's attack and sustain. Using slow compression attack can sound as if EQ has been used to boost high or high middle frequencies. Fast compression attack can give the opposite impression and make a sound seem to be dull. Space perception can also be effected with varying amounts of gain reduction. On drum room microphones, extreme gain reduction can bring up the volume of room reflections in relation to direct sounds that usually results in a larger room sound. It is as if a reverb has been added to the drum sound.

**Equalization**
EQ will affect the volume of a track. Since EQ is simply boosting or cutting certain frequencies of a sound, in effect those portions of the sound are being turned up or down. But, the inclination to change a track's volume as a result of EQ may not always be justified. Instead, it could be an indicator that the equalization is too extreme.
Mixing Preparation

Once the mixing stage is reached, it is wise to optimize the session at hand so that is taxes
the computer being used as little as possible. Also, when working with a record label there are
turn-in requirements that are generally the responsibility of the mixer. The following is the
procedure I use to address both needs. This process is somewhat tedious and many of the details
that I will describe seem minute. But when all the small details to the procedure are used they
can greatly increase workflow and save an enormous amount of time.

I began mix session preparation by doing a "save copy in" onto a separate hard-drive. If I
have an issue with an edit or need an alternate take I can also draw from the original tracking
sessions. (If I am working on a project solely as a mixer, I do not bother with doing a "save copy
in" since it is the producer or artist's responsibility to keep a backup of work up to that point.

Track Naming

To have an efficient mixing workflow, it is important to make each session as familiar as
possible. One way to make a session familiar is to have a consistent track-naming scheme. But
it is also important to keep the essence of the track name intact since clients will refer to a track
name that they are familiar with.

This is a general naming scheme that shortens track names but keeps the essence of the
name. This also serves as a key for the track names in these mix sessions. In this instance, I
named the tracks consistently with this name scheme so it was not necessary to change the
individual track names. But when both microphones for a single guitar take were bussed to a
single auxiliary, a simple name that identifies the theme of the tracks was given to the auxiliary.

Kick In = Kin  Kick Out = KOut
Sample Kick = SamK  Snare Top = SnTp
Snare Bottom = SnBtm  Sample Snare = SamSn
Rack Tom and Floor Tom = Toms (Both tom tracks are put on a single stereo track)
Drum Overheads = OH
Drum Room microphones = Rm
Sample Snare Room = SnRm
Tambourine = Tambo
Bass guitar D.I. = BsDI
Bass guitar amp = BsA

*All of the electric guitars on this album were recorded with two microphones with the exception
of two guitar tracks in Palm Trees In The Mountains. The dynamic microphone is always
labeled with an “A” and the condenser or ribbon is always labeled with a “B”. When a guitar
track is reference with a name like “Clean guitar 1” or “Cl1”, it will be assumed that there are two microphones for the take unless otherwise specified.

Clean guitar 1 Sm57 = Cl1A or Cl1.57
Clean guitar 1 Neumann U87 = Cl1B or Cl1.U87
Cl guitar 1 Royer 121 = Cl1B or Cl1.121
Cl guitar 1 Cascade Fathead = Cl1B or Cl1.Fat

Distorted guitar 1 Sm57 = Dr1A or Dr1.57
Distorted guitar 1 Neumann U87 = Dr1B or Dr1.87
Distorted guitar 1 Royer 121 = Dr1B or Dr1.121
Distorted guitar 1 Cascade Fathead = Dr1B or Dr1.Fat

Main rhythm guitars/ Power chords = Pwr
Verse guitar = VsGt
Chorus guitar = ChGt
Lead Guitar = LdGt
Lead Vocals = Vx
Lead Vocal Double = Dbl
Background vocal = Bg
Choir = Chr

File Organization

Next I check that all regions have fades or crossfades and that all edits sound natural and do not have any artifacts. Regions on individual tracks are selected along with any space that exist between the very beginning of the Pro Tools session and the first region. The multiple regions on that track are then consolidated into a single file by hitting Shift + Option + 3. This is done for each track in the session. Multiple microphone sources, like a drum set or a guitar amplifier that used two microphones, are consolidate together. I avoid consolidating all of the tracks in the session at the same time since some tracks carryout longer in the song than others. If all of the tracks are consolidated at once, the shorter tracks will have region space added to the end without any audio. This is not necessarily the worst thing but it does take up valuable hard-drive space.

To clearly identify the newly consolidated files from old files they are renamed by putting the initials of the song at the front of the file name. If the name of the song is one word then the entire word is used. The most efficient way to rename files is done by hitting return followed by Apple + a. This will selected all of the files in the edit window. Command shift + apple + r will bring up the file renaming window. The number at the end of the filename that Pro Tools adds is deleted and the song initials with a hyphen is typed at the front of the filename. It is not uncommon for a session to have fifty to a hundred tracks so it is helpful to copy the song initials and hyphen and continue the renaming process with a sequence of keys that makes the process relatively quick and painless. Hit enter to go to the next file, hit the right arrow key, delete the numbers at the end of the filename, hit the up arrow, then paste with apple + v. With
practice this become fluid and the whole process will be done in a few minutes.

Now all of the unused files can be select by hitting Shift + Apple + u and then removed with Apple + Shift + b. Pro Tools will ask if you would like to delete the files from the hard-drive or remove them from the session. I prefer to first remove them from the session since renaming errors can occur and Pro Tools does not always accurately remove all the unused files. Close session, go to the audio files folder for that song, and trash all audio files that do not have initials of the song at the front. Do not empty trash the can. First, reopen the session. If no files are missing go ahead and empty trashcan. If files are missing, close session without saving, move trashed audio files back into the audio files folder and reopen the session. Search for a consolidated file in the edit window that was not renamed or small bits of regions that were not deleted. Once the file has been renamed or deleted, repeat the above steps.

At this point, it is wise to do a "save as". When doing mix preparation and mixing for others, it is always good to have their working mix/rough mix to gain a sense of the direction they are looking for. For my own projects, I like to look back to mix ideas that I may have been experimenting with in the production process.

Additional Workflow Strategies

Efficiency can be improved by moving any left/ right mono files like drum overhead, or stacked background vocals to a single stereo track. (Stacked refers to a track recorded twice exactly the same way typically so it creates a large stereo sound. This would also be known as a double.)

As I said before, familiarity is key to an efficient workflow. That is furthered by always using the same preferences. For color coordination, the kick and snare is bright blue, the rest of the drums and percussion are dark blue, and bass guitar is pink. Electric guitar, acoustic guitar, and anything like banjo or mandolin are orange. Piano, keyboards, and strings are bright green. All vocal tracks are yellow. In the edit window all of the inserts, sends, and I/O controls are hidden because all of that is controlled in the mix window. Scrolling is turned off so I can continue editing or automating a section while the music continues to play. Plugins are moved to a second computer screen along with buss faders, the transport, the track coloring dialog box, and the system usage window. This is not a definitive way to work. It is simply a familiar setup that I have found to be efficient for my work style. Different strategies work for others. This only encourages engineers to find a system that works best for them.

Drum Samples (Steven Slate's Trigger)

For the most part, I will add kick, snare, and possibly tom samples at this point unless the drums are closer to a jazz kit or some sort of experimental sound. All of the kicks, snares, and toms in this project, with the exception of the snare and toms played with brushes in "The Fall", were augmented with drum samples. I normally do not entirely replace the original drum sounds
because it inevitably leads to an artificial drum machine sound.

I like to have a fairly consistent kick, snare and tom sound throughout an album so I chose an Ayotte 22" sample from a library of kick samples, Steven Slate’s Snare 11 Z3, Steven Slate's Snare 11 SSDR and Graham's Master of Maple tom samples that were recorded during the production/tracking process. The "Z3" qualification on the Steven Slate Snare refers to the amount of ambience/room sound on the snare sample. Z1 is a sample with virtually no ambience, Z3 has about as much ambience as one would expect from a close microphone recording in a medium sized room, and SSDR is a stereo sample of a snare recorded with room microphones at a distance. To avoid too many inharmonic tones created between the original snare and the snare sample, the predominant pitch of the sample snare is tuned to the predominant pitch of the original snare. The same tuning is used for the SSDR sample.

The plug-in I used to place the drum samples, Steven Slate's Trigger, has several other parameters that can alter the sound of the sample. The input parameter is meant to set Trigger so that the loudest transients match the loudest sample. All other recorded snare hits would trigger a sample with a lower velocity. This parameter can also be treated almost as a limiter. The input can be increased so that some of the quieter transients trigger the loudest sample, which produces a more dynamically even sample track. Typically the highest input setting should be avoided since it would always trigger the loudest sample and create an unnatural drum machine sound. Also, if the input signal is too high, sustain of the drum will create false triggers.

Sensitivity controls how much of the nuance in the original signal should trigger a sample. It is helpful to turn up the sensitivity if ghost notes on a snare track or the quietest snare hits during a drum buildup need to be detected. Excessive sensitivity, just like high input, can read the sustain of the drum as another transient. In most other cases, the default setting does well to trigger at the appropriate time.

Retrigger can be used to lengthen the time that it takes to trigger the next sample. If the retrigger time is set too low, sustain of the drum will cause a false trigger. Adjusting this setting with the input and sensitivity helps to avoid any false triggers.

Detail controls how much of the dynamic range should be triggered. At zero, quiet soft transients will trigger a sample and at the maximum setting, only the loudest transients will trigger a sample. This often needs to be adjusted throughout a track to pickup ghost notes in one section and avoid transients that are produced by microphone bleed.

For the output, I normally add about 3db. Strangely, the output of Trigger tends to be a little quieter than the original signal. The output signal is raised so that the drum sample and original signal have roughly the same level. If the sound of a snare sample needs to be very close to that of the original recorded snare, the sustain and release can be adjusted to shorten the length of the snare ring and the snares, respectively. When working with a client’s mixes I have the mindset of respecting the sounds they provided so I create a snare sound very similar to the original. For this project I chose to keep the full length of sustain and release.

When limiting the original snare signal with the input does not create a snare sample that
is sufficiently dynamically level, the dynamic parameter can be reduced. Trigger, like most other drum sample plugins, tends to over-accentuate the dynamics. Usually quieter snare hits trigger snare samples that have a far more subdued velocity. Setting the dynamics around .55 produces a snare sample track that is more natural sounding and closer to the original performance.

After drum samples have been recorded to new tracks, each sample is aligned with the original drum hits at the millisecond level and the phase is corrected if necessary. Normally I align the kick drum sample with the microphone inside the drum, as opposed to a microphone outside of the drum. In this case I had two microphones at the same distance from the kick drumbeater. The snare sample was aligned to the top snare transient. Once any alignment issues on the close snare sample have been corrected, it is used to trigger the snare room sample (SSDR). I align the SSDR with room transients but do not bother aligning each sample since it will be low in the mix and serves more as a reverb. If the drum tracks have dynamic builds or detailed snare parts the alignment can take twenty to thirty minutes. Fortunately, Trigger is rather accurate so if the tracks are not too intricate only a few samples will need to be moved. To keep files organized and consistent, the drum sample tracks are consolidated to the beginning of the session, renamed, and excess files are deleted just as they were at the beginning of the session preparation.

Next, any extraneous noise is removed. Groups are made for anything that should be edited together like all of the drum tracks, multiple bass microphones, or multiple guitar microphones. The speediest way to do this is to bring up Strip Silence with apple + U. The Region End Pad should be set to the maximum to allow long decaying sounds to end naturally. The rest of the default settings tend to work pretty well. Select each region in the session and hit strip. I finish by looking and listening for any sections that the instruments are not playing but have noise, deleting those sections, and putting fades on both sides of the regions. For guitars, bass, or any other instruments that were played through an amplifier, the region is faded so that amp noise is reduced but the decay of the instrument is still natural.

Finally, the vocal tracks are checked for any pop and/or clicks. Usually when a singer takes a breath there are pops as a result of the mouth opening. The pops are made more noticeable if the vocals were tracked with compression. There are a few ways to solve this problem. The simplest way is to delete the section that has the unwanted noise. However, the section may contain audio that is needed. For instance, deleting the middle of a breath or the breath entirely would be undesirable since a singer that does not breath sounds unnatural. The next option would be cutting the audio at the beginning of the breath and fading it in until the singer's first note. Sometimes the pops are no longer audible with a fade, other times the pop is still noticeable when the track is soloed but will not be detectable with all of the other tracks playing. If the first two options do not resolve the issue or if there are pops in the middle of a singer's sustained note then it might be necessary to zoom into the millisecond level and use the drawing tool. Typically the pop will look like a spike among an average waveform. The best way to fix the pop is to draw over the spike so the waveform is smooth without any anomalies. If the drawing tool does not work, I usually compromise by using a long fade over the breath. Not all pops and clicks can be removed. So it is wise to use all these techniques in combination, listen with other tracks to see what will be noticeable in the mix, and be willing compromise.
The Mixing Process and Equipment

A mix can be somewhat of a back and forth process. At times addressing one issue may make another issue come to light followed by another issued exposed and so on. But in general, there is a method and direction to a mix.

The following mixing details display one type of mixing process.

**Inputs, Outputs, Bussing, Inserts**

First an I/O appropriate to the board and equipment being used is made. The I/O below is tailored to the studio in which the album was mixed.

![Figure 3.1 I/O Input](image)

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### Figure 3.2 I/O Output

![I/O Setup Diagram]

<table>
<thead>
<tr>
<th>Name</th>
<th>Format</th>
</tr>
</thead>
<tbody>
<tr>
<td>A 1-2</td>
<td>Stereo L R</td>
</tr>
<tr>
<td>A 2-4</td>
<td>Stereo L R</td>
</tr>
<tr>
<td>L 3/4</td>
<td>Mono L B R</td>
</tr>
<tr>
<td>L 1-2</td>
<td>Stereo L R</td>
</tr>
<tr>
<td>L 3-4</td>
<td>Stereo L R</td>
</tr>
<tr>
<td>L 5-3</td>
<td>Stereo L B R</td>
</tr>
<tr>
<td>L 6-5</td>
<td>Stereo L B R</td>
</tr>
<tr>
<td>L 7-6</td>
<td>Stereo L B R</td>
</tr>
<tr>
<td>L 8-7</td>
<td>Stereo L B R</td>
</tr>
<tr>
<td>L 9-8</td>
<td>Stereo L B R</td>
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<tr>
<td>L 10-9</td>
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<tr>
<td>L 11-10</td>
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<tr>
<td>L 15-14</td>
<td>Stereo L B R</td>
</tr>
<tr>
<td>L 16-15</td>
<td>Stereo L B R</td>
</tr>
</tbody>
</table>

**New Path...**  **Delete Path**  **Delete All**

**Controller Input Path:**
- Name: NEW PATH
- Description: Description

**Default Output Bus:**
- A 1-2

**Default Monitor Format:**
- Stereo L B R

**I/O Path Order:**
- C A B L R C L R A B C

**Low Latency Monitoring:**
- Default: 20

---

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Figure 3.3 I/O Busses
Routing Details

- Spc: Out Interface A Analog 1 (Channel 1 on 192) -> SpaceStation -> In Interface A Analog 1 (Channel 1 on 192)

- Spr: Out Interface A Analog 5-6 (Channel 5-6 on 192) -> Accuvibe (Spr) -> In Interface A Analog 5-6 (5-6 on 192)

- Sta: Out Interface A Analog 7 (Channel 7 on 192) -> Sta Level -> Ltd -> In Interface A 7 (7 on 192)

- Dr: Out Interface A Digital 1-2 (Channel 9-10 on 192) -> Radar 1-2 -> Trident Channel Return 11-12

- Gtrs: Out Interface A Digital 3-4 (Channel 11-12 on 192) -> Radar 3-4 -> Trident Channel Return 13-14

- Key: Out Interface A Digital 5-6 (Channel 13-14 on 192) -> Radar 5-6 -> Trident Channel
After an I/O has been established, master faders are created for the main outputs. They are K (kick)/ Sn (snare), Dr (drums), Bass, Gtrs (guitars), Keys, Vox (vocals), and Fx. All non-pitched instruments and percussion is sent out the drum output. Both electric and acoustic guitars along with plucked stringed instruments like banjo and mandolin are sent out the guitar
output. Synthesizers/ keyboards, pianos, pads, and strings use the keys output. Effects that are add to a sound via buss and not directly on the channel like delays and reverbs go out the FX output.

**Bussing**

To make a mixing session move quicker, like sounds are bussed to a single auxiliary channel. For instance, an inside kick microphone and an outside would be sent to a single auxiliary channel. For these mixes there was a kick auxiliary, snare auxiliary, drum room auxiliary, bass guitar auxiliary, and multiple guitar auxiliaries. All of the electric guitars were recorded with two microphones, with the exception of a couple guitar tracks, so each two-microphone combination was sent to individual auxiliaries. Those exceptions will be described in the detailed mix explanations when they apply. There will also be unique bussing situations that apply to particular situations that will be described later.

**Levels**

General levels are carved in as soon as possible. At the very start, exact levels are not incredibly important since they will change as compression and equalization is applied. But there are a few things to focus on in the early stages. First, the auxiliary tracks should be receiving as much level as possible, without clipping, from the source (sending) tracks. Once the auxiliary tracks have a strong level the sending tracks can be hidden to minimize the amount of faders that need control.

For the purpose of gain staging, in terms of building the entire mix so that it does not clip the stereo output, the kick drum level should be close to it's final level. There are a few ways to find a good level for the kick. The first is to do a very quick rough mix focusing on the relationship of the lead vocal to the kick drum since the lead vocal is often the loudest in a mix and consequently what would cause the mix to clip. With this approach, it is likely that the mixes overall output will need to be adjusted since the foundation is just an approximation. The other option is to gain familiarity with the mixing console or whatever equipment is being used. When working with an analog console, familiarity is extremely important since they often have a certain range of stereo output that is optimal. If a mix is too quite, the appealing characteristics of the console may not be entirely present. If the mix is too loud then clipping occurs. Ideally, for music that needs plenty of impact, an analog console's output should be pushed right below its breaking point. The best way to know the range where a console operates best is to mix several songs on it. Fortunately, after preparing countless sessions on the Trident 80C, I had an accurate assessment of where to begin a mix.

**Plugins, Hardware Inserts, Bussing to Effects**

At this stage plug-ins, hardware inserts, and sends to effects are introduced. Since the album will have an overall similar sound, session data is imported to all of the mix sessions following the first mix. For tracks like drums, bass, rhythm guitars, and vocals, the volumes are
imported from the first mix. Buss routing for sound sources utilizing multiple microphones, sends to effects, plugins, and hardware inserts are imported to like tracks. Importing session data is a great way to establish a foundation for a mix quickly and efficiently. But, a mix is far from complete at this point. Each mix is unique and will need the same attention that the preceding mix received.

**Universal Mix Settings**

Several aspects of the mix setup used on this album remained static throughout the album. In other words, settings on some hardware and plug-ins were the same for every song.

**Hardware Inserts**

All of the hardware inserts had a compressor or limiter followed by an EQ with the exception of the Vx1 hardware insert that had a LA3A feeding into another LA3A followed by a Rupert Neve Eq. Almost all of the compressors in the hardware insert signal chains were static. The only thing that changed on most of the hardware chains was the input and output levels. The frequencies that were boosted or attenuated on EQ units were the same for every song. Since many of the sounds processed with the hardware signal chains were fairly consistent and further adjustments were made in Pro Tools, the degree of boosting and cutting was the same with many EQs throughout the album.

**Bs Hardware Insert**

The Bs hardware insert was dedicated to bass guitar. Both the compressor and EQ in the Bs signal chain were static. The bass was first routed to a Tube-Tech CL1B that had a threshold around -12dB and a ratio of approximately 4:1. The fixed attack and release setting was chosen so the attack had a continuously variable rate of 0.5 milliseconds to 300 milliseconds and the release had a continuously variable rate of 0.05 seconds to 10 seconds (Tube Tech). Gain reduction was typically about 3dB but occasionally 5dB.

*Figure 3.5 Tube-Tech CL1B Compressor on Bass Guitar.*

After the Tube-Tech, the bass guitar went to a Manley Enhanced Pultec EQ. The low shelf was set to 90Hz and the boost/ cut knob was about 8.5. The amount of boost measured in decibels is continuously variable dependent on the frequency selected so the boost/ cut knob does
not reflect a specific decibel. Based on the Pultec manual, a knob setting of 8.5 is around 7 or 8db. (Manley Laboratories, INC.) The high frequency was set on 4kHz and the boost knob was at 8.5. The bandwidth can be set from 1 to 10, 1 being the narrowest and 10 being the widest. (Manley Laboratories, INC.) At 4kHz, a bandwidth of 1 and a boost setting of 8.5 will translate to approximately to a 17dB boost (Manley Laboratories, INC.).

![Manley Enhanced Pultec EQ on Bass Guitar.](image)

**GME 1-2 Hardware Insert**

The GME 1-2 hardware insert was used on acoustic guitars. First in the signal chain was an Art ProVla stereo compressor with the threshold at 0db, the ratio at 2:1, and fast release selected. On average there was about 2-3db of gain reduction but occasionally hitting 4db of gain reduction. The attack time is program dependent with 20ms the slowest at 20db of compression (Applied Research and Technology). As gain reduction decreases, the attack becomes faster (Applied Research and Technology). The fast release, which has a fixed release time of 300ms, was engaged (Applied Research and Technology).

![Art Pro VLA Dual Channel Compressor on Acoustic Guitars.](image)

From the compressor the acoustic guitars went to a pair of Melcore GME two-band EQs. On the low band 55Hz, 110Hz, 220Hz, or 440Hz can be cut or boost 10db. For the high band, a 10db cut or boost can be made at 880Hz, 1.76kHz, 3.52kHz, or 7kHz. Since the GME EQs were dedicated to the same acoustic guitar sound, with the exception of Cast In Different Melodies, the chosen frequencies and their alterations remained the same for every mix. Even though the acoustic guitar on Cast In Different Melodies sounded somewhat differently, the same settings on the GME hardware insert proved to be sufficient with further adjustments in Pro Tools. The frequencies were set to 440Hz and 7kHz with a 2db cut and a 2db boost, respectively. This reduced excessive middle frequencies and amplified the attack of the guitar.
Api 1-2 Hardware Insert

The Api 1-2 hardware insert was primarily used for rhythm guitars that were recorded with a Shure Sm57 and Royer 121 microphone combination but for a few songs it was used on a different guitar sound that required different EQ settings. However, the first units in the signal chain, a pair of Dbx 160 VU compressors, were static for the entirety of the album. The ratio was 1.5:1, the threshold was at -3db, and gain reduction ranged from 5 to 10db. The attack and release times on the Dbx 160 VU compressors are program dependent (Dbx Profesional Products). The attack reacts in 15 ms when the signal exceeds the threshold 10db (Dbx Profesional Products). When the signal breaks the threshold by 20db it takes 5 ms to react and at 30db it takes 3ms. (Dbx Profesional Products) The release moves 120db over one second (Dbx Profesional Products).

The Api 550A EQs have three bands. On the low frequency band the selections are 50Hz, 100Hz, 200Hz, 300Hz, and 400Hz. The middle frequency band can be set to 400Hz, 800Hz, 1.5kHz, 3kHz, and 5kHz. For the high frequency band, 5kHz, 7kHz, 10kHz, 12.5kHz, or 15kHz can be chosen. The amount of EQ was different on several songs but the frequencies from low to high stayed on 300Hz, 800Hz, and 5kHz.

Gt 1-2 Hardware Insert

Some lead and rhythm guitars recorded with the Shure Sm57 and Neumann U87 microphone combination were processed with the Gt 1-2 hardware insert. This signal chain included a pair Universal Audio 1176 replicas made by Purple Audio called MC77s. The MC77s have an extremely fast attack (20usec – 800usec) so the attack was put on the slowest setting.
(Purple Audio LLC). The release, which ranges from 50msec to 1.1sec, was set to the fastest
(Purple Audio LLC). The ratio was set to 4:1 and the gain reduction was about 5-10db.

![Figure 3.10 Purple Audio MC77 Compressors on Electric Guitars.](image)

Following the MC77s was a pair of Chandler Limited EQs with a low-boost, mid-cut,
high-boost band. A cut or boost, depending on the capability of the band, can be made in
increments of 2db. The low band can be boosted up to 20db at 50Hz, 70Hz, 100Hz, or 200Hz
with a peak band as well as 100Hz and 200Hz with a shelf band. Up to 20db can be cut on the
mid-cut band at 350Hz, 400Hz, 500Hz, 600Hz, and 700Hz. At 1.2kHz, 1.8kHz, 2.2kHz, and
3.9kHz the high boost works as a peak-band that can be boost up to 18db. A shelf is used with
the selection of 5.8Hz, 6.8kHz, 8.1kHz, 12kHz, or 16kHz.

![Figure 3.11 Chandler Limited EQs on Electric Guitars.](image)

In general, the rhythm guitars recorded with the Shure Sm57 and Neumann U87 were
lacking some force and fullness so 100Hz was brought up 2db. Excessive low middle
frequencies were cut at 350Hz by 2db while 5.8Hz was increased by 4db. These adjustments
with Chandler produce some interesting harmonics and had a big contribution to the aggressive
nature of the rhythm guitar sound. The lead guitar sounds that were processed with Gt 1-2
hardware insert varied, but the same settings worked well as a broad-stroke approach.

**Act 1-2 Hardware Insert**

The Act 1-2 hardware insert was used primarily on clean or lead guitars. Purple Audio
Action compressors were first in the signal chain. The Actions are essentially a “500" model of
the MC77. The ratios, attack time, and much of the circuitry is identical to the MC77 (Purple
Audio LLC). Consequently, they were used in the same fashion. The attack was set to the
slowest and the release was set to the fastest. Gain reduction ranged from 5-10db.
Followed by the Action compressors was a two channel JDK R24 EQ. The JDK was modeled after a vintage APSI 562 EQ (JDK Audio). APSI was a company that had a sort of partnership with API that granted them access to components that were used in API units (Pan60). Even though the company was short lived, some units are rare and highly desireable (Pan60). On this replica, each channel has four peak-bands that are continously variable with 12db of boost or cut. The first band ranges from 20Hz-200Hz, the second 100-1kHz, the third 500-5kHz, and the fourth 2kHz-20kHz.

The frequencies selected were 200Hz, 350Hz, 2.5kHz, and 5kHz. Both low frequencies were cut 3db and both high frequencies were boosted 3db. Different guitar sounds were processed with this EQ, but in general guitars often need low frequencies and low middle frequencies reduced along with significant high-end boosting.
Tla 1-2 Hardware Insert

Another hardware insert used for clean and lead guitars was TLA 1-2. Two Dbx 160x compressors were followed by a TL Audio EQ-1 Dual Valve EQ. The Dbx compressors provided about 4-5dB of gain reduction with a 2:1 ratio. Attack time on the Dbx compressors is program dependent and ranges from 3ms to 15ms (Dbx Professional Products). Release times are program-dependent and range from 0-500ms (Dbx Professional Products). The second compressor had the slave button in so that it’s compression mirrored the compression of the first compressor and the Over Easy button was in so that the compressor had a softer knee (Dbx Professional Products).

Figure 3.14 DBX 160x Compressors on Electric Guitars.

The TL Audio unit has two channels of four-band switched frequency EQ that can be cut or boost 12db on each band (Tony Larking Professional Sales Limited). The low-frequency band has a shelf at 60Hz, 120Hz, 250Hz, or 500Hz. Next is a low-middle frequency peak band selection of 250Hz, 500Hz, 1kHz, or 2.2kHz. The high-middle frequency is also a peak band that can be switched to 1.5kHz, 2.2kHz, 3.6kHz, or 5kHz. The low-middle and high-middle frequency bands have a Q of 0.5. Like the low-frequency band, the high-frequency band is a shelf. This band can be switched to 2.2kHz, 5kHz, 8kHz, or 12kHz.

For these mixes, the chosen frequencies were 120Hz, 250Hz, 2.2kHz, and 5kHz. The low and low-middle frequencies were cut 3db and the high-middle and high frequencies were boosted 3db. Both channels were identical. The approach used with the JDK EQ applied to the TL Audio EQ as well.

Figure 3.15 TL Audio Dual Channel EQ on Electric Guitars.

Sta Hardware Insert

The lead vocal used the Sta hardware insert with the same treatment on every song. Compression was done with a Retro Sta-Level. The instruction manual only gives vague descriptions of how the unit works, but the controls are fairly simple. Mode has three attack options (Retro Instruments). Single is a slow attack, double is a program dependent attack and
release, and triple is program dependent like double with a faster attack (Retro Instruments). The recovery time, or release time, has selections from slow to fast (Retro Instruments). The amount of gain reduction is set with the input control. On average, the Sta-Level gave the vocal about 20dB to 25dB of gain reduction. In general, the Sta-Level is very transparent. But with the aggressive gain reduction it was best to set the compressor on the slowest attack at single and the fastest release all the way to the right (Retro Instruments).

![Figure 3.16 Retro Sta-Level Compressor on Lead Vocal.](image1.png)

The Chandler Limited LTD-1 has a high pass filter that can be set to 50Hz, 80Hz, 160Hz, 300Hz. There are four choices for the low frequency shelf band. It can be set to 35Hz, 60Hz, 110Hz, and 220Hz with 10db of cutting or boosting. A bell shape is used for the middle frequency band. The band can be boost or cut up to 18db at 270Hz, 380Hz, 560Hz, 820Hz, 1.2kHz, 1.8kHz, 2.7kHz, 3.9kHz, 5.6kHz, and 8.2kHz. The high frequency shelf band has a selection of 3.3kHz, 4.7kHz, 6.8kHz, 12kHz, and 16kHz that can be boost or cut 18db.

When singing words with plosives, there is air moved that cause distracting low frequency sounds. The Chandler LTD-1 was set to a 80Hz high-pass to resolve that issue. Kevin’s voice has a pleasantly warm sound. However, in a dense mix some of the frequencies that compose that warm tone must be reduced to fit the vocal amongst the instruments. That was accomplished by making a 2db cut at 60Hz. Even though the cut was made below the 80Hz high-pass, that particular band is fixed on a bell curve. Consequently, the adjustment has a subtle effect as the bell tapers off at higher frequencies and approaches zero. The vocals also had a low middle frequency reduction at 270Hz by 2db and 16kHz was boosted 2db to help the vocals standout above all other aspects in the mix.

![Figure 3.17 Chandler LTD-1 EQ on Lead Vocal.](image2.png)

**Vx1 Hardware Insert**

A secondary vocal chain labeled Vx1 included one Urei LA-3A feeding into another followed by a AMEK (Rupert Neve design) 9098 pre-amplifier and EQ. An LA-3A fed into another because the attack time (1.5ms being the slowest) becomes slower with more gain reduction (United Recording Electronics Industries). To keep the compressor reacting quickly,
reduction was set conservatively at 3-5db. The first LA-3A also had the compression switch selected so that there is a low ratio and soft-knee. The mild compression kept the vocals from sounding too transparently processed.

![LA-3A Audio Leveler](image)

**Figure 3.18 LA-3A on a Second Vocal.**

The second LA-3A was switch to the limiter mode so that any excessively loud vocals that made it through the first stage would be kept under control. However, the limiting characteristics only come about with extreme compression (United Recoding Electronics Industries). If gain reduction is low then the unit functions just as it would when switched to compression (United Recoding Electronics Industries). Since the second LA-3A only had a bit more aggressive gain reduction of 5-7db, the attack and release time, the ratio, and the knee did not change much. Therefore, up to 12db of gain reduction was accomplished with out severely altering the natural attack of the vocal. With low gain reduction the vocal sustain also maintains subtle characteristics because the compressor releases according to the signal at a rate of 0.06 seconds (United Recoding Electronics Industries). Only The gain and peak reduction controls changed to maintain a consistent amount of gain reduction for each song.

![LA-3A Audio Leveler](image)

**Figure 3.19 LA-3A Set to Limit on a Second Vocal.**

The 9098 is a six-band EQ. The high pass filter ranges from 20Hz to 300Hz and the low pass ranges from 4.5kHz to 30kHz. The purpose of the extreme low pass range is to “remove distortion…in the upper frequency bands” (Amek). The other four bands can be cut or boost
18dB with the low frequency and high frequency band capable of working as a bell or shelf curve (Amek). On the low frequency band (100Hz-1kHz) the bell mode has a Q of 0.7 and on the high frequency band (2kHz-21kHz) the Q is approximately 0.45 (Amek). The low middle band ranges from 30Hz-1kHz with a Q from 0.65-2 and the high middle band has a range of 500Hz-4.5kHz with the same Q as the low middle band (Amek). Additional controls include a glow button on the low band and a sheen button on the high band that adds “warmth” as described in the manual (Amek). All bands also have a gain +/- button to limit the cutting and boosting to 9db so that smaller adjustments can be made (Amek).

Not much changed on the EQ for each mix. The high pass was set to 60Hz and the low pass was at 30kHz. The low frequency band utilized the shelf mode at 150Hz with the glow button in. For the low middle frequency band, 300Hz was selected and the high middle frequency band was set to 4.5kHz. Lastly, the high frequency band was at the maximum of 21kHz. In general, the low and low middle band was cut while the high middle and high band was boosted. However, the amount of cutting and boosting varied slightly on each song.

![Figure 3.20 Neve 9098 EQ on a Second Vocal.](image)

**Parallel Processing**

Much of the compression applied directly to the signal at hand was conservative. Typically compressors did not exceed 10dB of gain reduction and ratios were 2:1 to 4:1. These types of compressor settings are conducive to a natural sound. However, further compression is necessary to hear low-level details of an instrument or vocal.

**Dbx 119 Parallel Compression**

To circumvent the conflicting advantages and disadvantages of heavy compression, some parallel compression/limiting was employed. The first type of parallel compression was a sum of the K/Sn (kick and snare) output and Dr (drum) output. K/Sn and Dr were patched from group out 15-16 on the console to a Dbx 119. The settings on the Dbx 119 are fairly minimal. One control sets compression from infinity to 1:1 ratio. That control continues to move into expansion up to 2:1. The compressor has a linear mode and an “above threshold” mode. In “above threshold”, the unit only compresses signals above the set threshold (Dbx Professional Products). Unlike most compressors, the threshold on the Dbx 119 is expressed in volts as opposed to decibels. The threshold ranges from 1 volt to 10 millivolts, which converts as 2.2dB to -37.78dB. The threshold control is only labeled at its minimum and maximum so only an estimate of the setting can be made. Judging off the control’s range, the threshold was set to compress above roughly -30db. So most of the signal passing through the unit was being compressed. The ratio was set a little above 3.5:1. The signal of the Dbx 119 was brought back to Channel Return 1-2 a few decibals lower than the unprocessed K/Sn and Dr outputs. This technique preserves the characteristics of uncompressed drums, particularly a present high-end,
and improves the audibility of ghost notes and other quiet details in the drummers playing.

Dbx 160x Parallel compression

The same principle was applied by routing the entire mix to group outs 19-20 into two Dbx 160x compressors/limiters. The units were set to limit at a 10:1 ratio, but the threshold was set to +10db so that only the loudest signals engaged the limiters. On average, there was 5-6db of gain reduction. The limiters were given 5db of make-up gain and routed back to channel return 5-6. Again, having a stereo output independent of a limited stereo output makes low-level audio easy to hear without perceptively degrading effects of aggressive limiting.

GTQ2 Parallel EQ

The entire mix was also routed to group out 21-22, which went to a Phoenix Audio GtQ2 Mark II Eq. The GtQ2 has two channels that have identical three band EQs. The EQs have a fixed 12kHz shelf, a fixed 80Hz shelf, and an adjustable midrange band that can be set to 400Hz, 1.6kHz, and 3.2kHz (Aurora Audio). The 80Hz and 12kHz shelves were given a boost of about 7.5dB and the midrange band was left at zero (Aurora Audio). Having the GtQ2 routed back to channel return 7-8 along with the original stereo output adds fullness and sheen.
Tube-Tech Parallel Compression

A Tube-Tech LCA2B was used to provide parallel compression over the entire mix. It was designed with a Fairchild compressor in mind (Tube Tech). The circuitry is not a clone of the Fairchild but things like the attack and release time constants are identical (Tube Tech). The attack time ranges from 0.3ms to 70ms and the release ranges from 0.07s to 2s (Tube Tech). Attack and release times can be manually adjusted or set by one of the six presets (Tube Tech). Position 1 has an attack of 1.5ms and a release of 0.25s (Tube Tech). Position 2 has an attack of 1.5ms and a release of 0.8s (Tube Tech). Position 3 has an attack of 3ms and a release of 2.2s (Tube Tech). Position 4 has an attack of 6ms and a release of 5s (Tube Tech). Position 5 has an attack of 3ms and a release of 0.5 to 4s (Tube Tech). Position 6 has an attack of 1.5ms and a release of 0.5 to 4 to 20s (Tube Tech). Position 5 and 6 are program dependent (Tube Tech). In position 5, sounds with a short sustain have a release of 0.5; while sounds with long sustain have a release of 4s (Tube Tech). In position 6 the release works in the same way but the additional 20s release time occurs when the level remains high (Tube Tech). The compression ratio can be set from 1.6:1 to 20:1.

On this album, the Tube-Tech applied very gentle compression. The threshold was at +3db so that only the peaks of the mix were compressed. Gain reduction was 1 to 2db at the most and the ratio was 3:1. The attack and release time was set to preset 1 (1.5ms attack and 0.25s release) so that the compressor was engaged for a very short time. This compression was not meant to provide significant dynamic control. Instead, the intention was to keep extreme peaks under control, produce a gentle dynamic bounce, and add pleasing harmonics that tube based hardware produces.
**Additional External Equipment**

**Space Station**

On most of the songs, all of the drum tracks besides the overhead and room tracks had a send feeding at -20dB to an Ursa Major Space Station SST-282. A low frequency cut at 20Hz up to 10dB and a high frequency cut at 7kHz up to 10dB has an effect on the input signal and delays created by the Space Station (Seven Woods Audio, Inc). For the drums, there was a 3dB low frequency cut. There are sixteen delay programs that are divided into four families. Most of the programs have multiple delays called taps that can be turned up or down with the taps controls under the audition delay mixer. As one might expect, the room family uses fairly random delay times to emulate early reflections of a room, room one being the smallest room and room four being the largest room (Seven Woods Audio, Inc). All of the other families increase in delay time according to a higher numerical name as well (Seven Woods Audio, Inc). Comb delay programs are more unconventional special effects. Delay clusters have a set of delays that are very close in time (Seven Woods Audio, Inc). The fatty cluster of delays is very close to the original signal and is indiscernible as a separate signal. The cloud cluster, which is what was used for the drum effect, is still very close in time to the dry but is noticeably behind the signal. The slap and echo clusters are even further in time from the original signal. Two to four repetitions of the original signal can be chosen with the space repeat selections. An additional reverb or echo mode is available and reverb or delay decay, depending on the selected mode, can be adjusted with the feedback control (Seven Woods Audio, Inc). When the reverb mode is selected, a long reverb program takes more time to develop early reflections and has longer delay times while the medium program introduces early reflections sooner and has a more linear decay (Seven Woods Audio, Inc). Reverb mode was selected with a long program, feedback at six, and all of the taps at full volume. Even with the long reverb program and feedback turned up, the effect was still fairly short due to the quick delays created by the cloud cluster program. But, it proved to be helpful in adding depth and interesting special characteristics to the drum sound. The Space Station signal returned to a stereo auxiliary at -10.7dB with a short delay timed to the tempo of the song at hand. (See 4Am mix notes for calculating delays to song tempos.)

![Figure 3.25 Ursa Major Space Station SST-282 for Delay on Drums.](image-url)
**SPX 90**

For a few songs, the lead vocal and sometimes background vocals were sent at -2dB to a digital multi effects processor made by Yamaha called SPX 90 (Yamaha). The unit has all sorts of effects like reverbs, delays, modulators, noise gates, and compressors. A pitch changing effect was used on the lead vocal. First, a mono signal was sent from the lead vocal to the SPX 90, which duplicated the signal so there is a left and right channel. The left channel was tuned down 8 cents and the right channel was tuned up 8 cents. That signal was then returned on a stereo auxiliary at -9dB in Pro Tools and blended quietly with the lead vocal. This gave the vocal a wide stereo feel and the perception that the vocal was louder and bigger.

![Figure 3.26 Yamaha SPX 90 for Vocal Effects.](image)

**Trident 80C Output (Main Stereo Output)**

**Alan Smart C2**

All of the main Pro Tools outputs and parallel processing outputs were assigned to the main output of the Trident console. From there, the summed mix went to an Alan Smart C2. Alan Smart originally worked for Solid State Logic so many people agree that the C2 has a similar sound to a Solid State Logic two-buss compressor (Smart Research Ltd). Even though the unit is primarily a two-bus compressor, which is was it is used for on this project; it can be used for a wide variety of applications (Smart Research Ltd). The second channel can be slaved to the first channel so that compression is identical on both channels. For this application, the channels had the same settings but were left to compress individually so that the left and right channels kept their own unique dynamic characteristics. All of the controls are stepped. The ratio choices are 1.5:1, 2:1, 3:1, 4:1, 10:1, or limit. Attack time is 0, 0.1, 0.3, 1, 3, 10, or 30ms and release time is 0.1, 0.3, 0.6, 1.2, or 2.4s. Similar to the Tube-Tech LCA 2B compression, the C2 compression was very minor. A gain reduction of 1db at the maximum, an attack of 30ms, and a release of 0.1s caused the mix to very subtly move with the same dynamics, which in turn gave the mix more cohesiveness.

![Figure 3.27 Alan Smart C2 Compressor on Main Stereo Output.](image)
**Millenia NSEQ**

Next was a vacuum tube Millenia NSEQ that is intended for stereo program material and commonly used for mastering (Millennia Media, Inc). The low and high frequency bands function as a shelf at 6dB per octave or a peak with a fixed Q of 1 (Millennia Media, Inc). Low band frequency selections are 20Hz, 34Hz, 56Hz, 100Hz, 180Hz, and 270 Hz. The high band has a selection of 4.8kHz, 5.8kHz, 8kHz, 10kHz, 16kHz, and 21kHz. Low mid and high mid bands have two ranges of frequencies. With the X10 button out, the low mid band is adjustable from 20Hz to 220Hz (Millennia Media, Inc). When the X10 is in, the band ranges from 220 to 2.5kHz (Millennia Media, Inc). The high mid band ranges from 250Hz to 2.5kHz with the X10 button out and 2.5kHz to 25kHz with the X10 button in (Millennia Media, Inc).

Normally, each band has a 20dB cut or boost capability. There is also an option to limit the bands to a 10dB gain range for finer adjustments (Millennia Media, Inc). To avoid stepping on the toes of the mastering engineer, this option was engaged. Only the low mid band and high band were used with the X10 button selected on the low mid band. A half dB dip was made around 350Hz to prevent cloudiness and a half dB shelf boost was made at 16kHz to add shine.

![Figure 3.28 Millennia NSEQ on Main Stereo Output.](image)

**Plug-ins**

**Focusrite D3 Compressor/ Limiter**

A Focusrite D3 compressor was inserted on the K/Sn (Kick and Snare) master and the Dr (Drum) master. Both compressors were set with conservative ratios of 2:1. Their attack and release times were set rather fast at 30ms and 100ms, respectively. Both compressors would be keyed by sends on several drum set tracks. With these settings the compressors were inducing a very light bouncing effect, particularly when the kick and snare played since they are usually the loudest components in the drum-set.
A.I.R Stereo Width

An AIR Stereo Width plug-in was inserted on an auxiliary named Wide. The Spc output was assigned directly to the Wide auxiliary to widen the perception of the Spc signal or any other signal sent to the auxiliary.

Figure 3.29 Focusrite d3 compressor on Dr and K/Sn.

Figure 3.30 A.I.R Stereo Width on Wide.

4 A.M. Mix Details
Kick (K)

The kick drum was the first thing to address in this mix just as it was in all of the other mixes that included a drum-set. When it came time for mix, the Sm91 microphone seemed unnecessary. It was made inactive and hidden (the same applies to every mix except Perfect Remedy). The Audix D6 and the Ayotte kick sample were bussed to a single mono auxiliary and that auxiliary was assigned to the K/Sn (Kick and Snare output). The insert chain began with a Waves Solid State Logic (SSL) channel plugin. Because the drums were pretty heavily compressed during tracking, there was not much need for more compression. But there was a generous amount of EQ used. A boost of 3.8dB at 6.8kHz and 4dB at 4.01kHz was made since the kick is generally a low frequency source, which is typically difficult to hear. A significant cut of 7.5dB at 500Hz was made because most sound sources provide plenty of low middle frequencies. If low middle frequencies are allowed to build up in a mix it will likely lack clarity and definition. For the high middle frequencies and the low middle frequencies the Q was set to 0.10. The kick along with the bass guitar will be the main source of impact, frequencies that can be felt, so 104Hz was boosted 0.5dB.

Oddly enough, the sustain of the kick trailed on a bit too long so an SPL Transient Designer was used to shorten the sustain. Even though the same kick sample, kick drum, and microphone was used on all but one song, this was not an issue on the other mixes.
The kick also had two sends. The first send fed to the Space Station at -30.8dB and returned on the stereo auxiliary labeled Spc with a Digidesign short delay inserted.

A delay was calculated so that the Space Station signal would coincide with the tempo of the song. The delay was calculated using a simple equation. Since there are a 1000ms in a second there is 60,000ms in a min. By dividing 60,000 by the BPM, 159BPM in this case, we find the quarter note expressed in ms (Owsinski, The Mixing Engineer's Handbook). Divide that number by two to find the eighth note expressed in ms, again by two to find the sixtieth note.
expressed in ms, and so on. The delay on the Spc auxiliary was set to sixty-fourth notes, which is equal to 23.58ms. For other songs the pre-delay for the drums might be set to 128th notes. It is dependent on the tempo of the because higher BPM songs will produce shorter sixty-fourth notes just as slower BPM songs will produce longer sixty-fourth notes.

The second send on the kick track was set to unity and keyed the Focusrite compressor on the Dr and K/Sn master faders. The Sn, SnComp, Toms, SamToms, OH, and Rms tracks all had a send at unity to key the Focusrite compressor as well.

**Sn (Snare)**

The first auxiliary named Sn (Snare) did not have compression inserted to introduce some parallel compression. By using parallel compression, the track without compression maintains higher frequencies that would be slightly muffled by compression and the compressed track brings up the volume of quiet details that would normally be hard to hear. Having a parallel signal without compression also adds to the volume and keeps important changes in dynamics.

On the Sn auxiliary there were two plug-ins. The first plug-in was a SPL transient designer. The attack was increased. There was also a troublesome ring in the snare so the length of the ring was reduced with the SPL’s sustain control.

![Figure 3.34 SPL Transient Designer on Sn.](image)

The second was an EQ with a high pass at 65Hz to reduce any unnecessary frequencies that could interfere with the kick and bass. A boost of 7.4dB was done at 1.41k to emphasize crack and 5k was brought up 4.4dB to give the snare more detail.
A send on the Sn track fed Spc at -15.8dB.

Compressed Snare (SnComp)

The snare top microphone, bottom microphone, and sample were also bussed to another auxiliary. The second auxiliary was named SnComp (Snare Compression) and had an SSL Channel plug-in inserted with a 2.7:1 compression ratio and 0.36s release. The snare was already compressed during tracking so there was not much need for extreme compression. The SSL Channel applied about 3db of gain reduction so that details in the snare were brought out. At 3.94kHz there was a 4.3dB boost to compensate for the compression dampening of high frequencies and the lack of a sufficient high frequency response in the microphone. A boost of 3.5dB was made at 1.01kHz to bring out to crack and attack in the snare. A cut of 5.9dB was made at 400Hz to reduce some ring in the snare and 280Hz was boosted to add some punch.

Figure 3.35 McDsp Filter Bank E6 EQ on Sn.

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A send on the SnComp track fed the Space Station at -15.8dB.

**Toms**

The high and low frequencies were enhanced with the Waves Renaissance six-band Equalizer. The tom sound was carved by listening for the center frequencies that the stick attack and resonance rest. Boosting at 100Hz provided the resonance and boosting at 4268kHz and 7885kHz brought out the attack of the toms so that they can be easily heard in the thick arrangement. A severe cut was done at 375Hz to give the toms definition.
The toms recorded in the drum sampling session were used to augment the original toms but only the signal from the Sm81 was used. The sample toms had no plug-ins inserted. This could be considered a pseudo form of parallel processing (in this case parallel EQ) since the original toms and sample toms were the same toms with the same recording equipment. Having the original toms with EQ along with the unprocessed sample toms provided a defined but full sound.

The original toms and the sample tom tracks were sent to the Space Station at -21.3dB.

**Overheads (OH)**

The left and right overheads were put on to a single stereo track to minimize faders but the left overhead was a little louder. The simple solution was to insert a trim plugin, unlink the plugin so that the left and right channels worked independently, and reduce the left channel volume.

Because the overheads have such a broad range of dynamics, compression is usually
applied generously. This can result in a duller sound, which calls for quite a bit of high frequency boosting to match commercial mixes. To start, a high pass was engaged at 84Hz to give the kick and bass guitar their own space in the low frequency spectrum. A 2.6dB boost was made at 14.04kHz to add a bit of sizzle to the cymbals. Not many other instruments have significant frequency content in that range so it is a good space to let the cymbals shine. A 4.3dB cut at 310Hz was made to reduce cloudiness.

![Figure 3.39 Waves SSL Channel on OH.](image)

Note: Specific volumes have no meaning in and of themselves. The purpose of specifying volumes is to show the relation that volume changes have throughout the mix.

For most of the song, the overheads were set at -9.6dB. Their level was lowered by 2.5dB in the break right before the chorus because their level was excessively loud in comparison to the rest of the track. Similarly, the high-hats, which were captured with the overheads, were brought down 3.3dB.

*Drum Room Microphones (Rms)*

The room microphones and snare room sample were bussed to a stereo auxiliary. The EQ approach was similar to the overhead, but the frequencies were shifted and the cuts and boosts varied. A high-pass was set at 84Hz, a 5.9dB boost was made at 14.12kHz, 3.29kHz was boosted 5.5dB, and a 2.1dB was cut at 310Hz.
For simplicity and efficiency, the bass guitar was recorded directly into a pre-amplifier with the intention of simulating the sound of a bass amplifier. The BsDI (bass D.I.) track was duplicated and renamed BsA (bass amplifier). Then Decapitator was inserted on the BsA track. Decapitator is a plug-in that emulates “the sound of analog gear being pushed hard to the point of saturation and beyond” (SoundToys, Inc.).

The style buttons emulate different types of analog equipment that have their own distortion character (SoundToys, Inc.). The “T” button, chosen for the bass sound, emulates a Culture Vulture by Thermionic Culture that add even harmonics to a signal (SoundToys, Inc.). Punish was turned on to significantly increase distortion and drive was turned up to produce a little more distortion. As the tone control is turned clockwise the low-end is decreased and the high-end is increased (SoundToys, Inc.). The BsA track was made radically brighter because low-end would be maintained with the BsDI track and plenty more low-end would be added with the Bs hardware insert. The high-cut was turned a little past one o’clock to remove some static-like distortion.

Bass (Bs)
The BsDI and BsA tracks were assigned to an auxiliary labeled Bs and the most substantial processing was done with Bs hardware insert. The equipment settings in the Bs hardware insert signal did not change from song to song and served as a global starting point. (Refer to Universal Mix Settings: Hardware Inserts). The bass guitar sound was fine-tuned with additional plug-ins.

Since bass guitar is so incredibly dynamic when changing from low notes to high notes, an L1 limiter was introduce to maintain a more consistent bass level. The varying levels of the bass guitar were controlled mostly with the compression from the Bs hardware insert so the L1 limiter was set to only control the loudest points of the bass guitar.
Usually bass guitar can be felt but not noticeably audible because it is predominantly composed of low frequencies that are difficult to hear. A brighter sound was achieved with a 1.5dB increase at 1.89kHz. This brought out attack and harmonics that gave the bass performance more rhythmic definition. A boost like this is very important when translating a mix to a sound system that lacks adequate low frequency. Even though much of the instruments character will be lost, there will still be some aspects present.

MaxxBass provided additional low frequency content by introducing harmonics built on the bass guitars fundamental frequencies (Waves Audio Ltd.). The low frequency harmonics that
were generated by MaxxBass were limited by setting the high pass to a 24dB/ oct filter (Waves Audio Ltd.). Too much extremely low frequency content can overload a sound system and degrade the clarity of a mix. The decay of the generated harmonics was shortened to avoid excessive frequency content (Waves Audio Ltd.).

![Image](image.png)

**Figure 3.44 Waves MaxxBass on Bs.**

A touch of volume automation was used on the bass guitar. The bass entered on the first pre-chorus at -10.1dB. In the break preceding the first chorus, the note ringing out had a fade down to -18.9dB so that there was more of a volume difference between the break and the chorus. When the chorus hit, the bass jumped back up in volume at 0.3dB above the pre-chorus. The performance in the second verse also needed some attention. Because the bass was played so aggressively in the verse, it was louder than the rest of the track. Multiple levels of compression helped the issue, but it still needed to move down to -11.4db.

**Rhythm Guitars (Pwr)**

Dr1 was panned hard left with Dr2 panned hard right. These two guitar tracks served as the main rhythm guitars. They were bussed to a single stereo auxiliary labeled Pwr (short for Power chords).

Depending on the situation, the hardware inserts may come first or second in the signal chain. Typically, the hardware inserts serve as the major global changes to a sound. However, a Filter Bank E6 EQ was inserted first to low pass unnecessary low frequencies that were activating the compressor in the hardware chain undesirably. The high-pass was set to 61Hz and a low-pass was set at 13.1kHz give drum overheads their own space in the frequency spectrum. After following inserts were introduced, this plug-in continued to be used for smaller adjustments at 873Hz for more aggressive middle frequencies, 1.50kHz for a touch more guitar pick sound, and 5kHz for a little more presence.
On most of this album, API 550As were used for the main left and right rhythm guitars. But after some experimentation, the Gt 1-2 hardware chain complimented the rhythm guitars that were recorded with the Sm57 and Neumann U87 very well.

When the mix was revisited at a later time with fresh ears, the rhythm guitars were still a little weak. Further boosting with the Chandler was more than needed since it was designed with stepped potentiometers (they only move in fixed positions). To make smaller adjustment the Pultec EQ plug-in, which also creates harmonics that further the aggressive nature of distorted guitars, was added to the signal chain. At 100Hz there was a 3db boost and at 4kHz there was a hint of a boost to improve the audibility among other instruments.
The rhythm guitars followed the same dynamic theme as the bass guitar. The chorus level started at -12.6db and then moved up 1dB for the choruses. When the song entered the second verse, the guitars were brought down to -13.1dB and back to -12.6dB so that there was a continual progression to a dynamic climax in the choruses.

**Verse Guitar (VsGt)**

Dr4 was panned hard left and Dr5 was panned hard right. These tracked played an identical simple lead line for a time and then Dr5 plays harmonies of Dr4. The rhythms remain identical so the two guitar parts were assigned to a stereo auxiliary to be processed as one sound.

In the second verse, the lead played at the same time as the main rhythm guitars. Because the rhythm and lead were recorded with the same guitar, amp and microphones, their sonic qualities were extremely similar and difficult to discern in the mix. Just as with the rhythm guitars, a Filter Bank E6 EQ was originally used as a high (61Hz) and a low pass filter (10.4kHz) while the Act 1-2 hardware insert shaped the overall guitar sound.

But even with broad changes to the Act 1-2 chain, the lead line still needed pretty significant changes to avoid a frequency collision with the rhythm guitars. The Filter Bank E6 EQ was revisited and used to further brighten the lead line and create contrast with the rhythm guitars. Low middle frequencies at 325Hz were reduced by 2dB to give the guitars more definition, 1.33k was boosted 3.5dB, and 5kHz was boosted 4.2dB.
In the beginning of the song there is only an acoustic guitar, the lead vocal, and the left and right electric guitar lead line. For that section VsGt was at -10.2dB. For the second verse the bass guitar, rhythm guitars, and drums were present so VsGt needed to be brought up to -6.7db.

**Reverse Guitar (RevGt)**

DrRev1 was panned hard left while DrRev2 was panned hard right. All of the tracks were assigned to the RevGt auxiliary and no processing was used in Pro Tools.

**Acoustic Guitars (Ag)**

Each acoustic guitar track was recorded with two microphones, a microphone in front of the acoustic guitar body and a microphone pointing at the third fret. When it came time for the mix, the neck microphone pointing at the fret was superfluous. The neck microphone tracks were made inactive and AgBod1 and AgBod3 were panned hard right and left, respectively. During the first verse, AgBod 3 was muted so that only Ag1Bod plays back with AgBod2, which
contains a percussion part that was played by muting the guitar strings and striking the strings on beat three. Both Ag1Bod and Ag2Bod were panned 11% to the left so that they seemed to be one instrument and did not conflict with the lead vocal that was panned to the center.

The first step to carving the acoustic guitar sound was the GME 1-2 hardware insert. To continue down that path of the GME signal chain, a Waves VEQ4 was inserted. A low-pass was set at 82Hz with a cut at 390Hz. To set the acoustic guitars apart from the electric guitars, different frequencies were boosted. At 2.2kHz and 10kHz there was a 3dB boost.

The acoustics were given more sustain with an SPL Transient Designer.

Only a single acoustic guitar played at the beginning of the song so the auxiliary was set louder, -1.5dB, at the beginning than the rest of the song. In the pre-chorus, another acoustic guitar was introduced so the level was brought down to -6.9dB. When the bridge came about, the only thing playing was acoustic guitars and the vocal so the auxiliary was brought to -1.4dB to suit the dynamic of the section.
Lead Vocal (LdVx)

The vocal processing began with the Sta hardware insert. At this point, the lead vocal had generous compression during tracking, which was followed by further compression with the Sta signal chain. But in Pop music, it is imperative to make the vocals/lyrics as clear as possible. It is difficult to achieve this with one compressor because the gain reductions become more transparent. For more subtle compression, it is best to use multiple levels of compression. The Waves Renaissance Vox was set to apply approximately 6dB of gain reduction on top of the tracking compression done with the Distressor and the compression done with the Sta-Level.

Figure 3.50 Massey De-Esser on LdVx.

One of the side effects of heavy compression is amplified consonant letters like “S” and “T”. They become harsh and make it difficult to turn the vocal level up without becoming bothersome. De-Essers automatically lower the level of the harsh consonants. The De-Esser was set to reduce consonants around 6kHz by approximately 9dB.

Some plosives were still noticeable so a Filter Bank EQ was inserted with an 85Hz high-pass.
The vocal was sent to several effects. The first send was very slight at -20dB to an external Yamaha SPX90 that was set to a stereo pitch deviation. (Refer to Universal Mix Setting: Additional Equipment Used for Effects).

In pop music, the lead vocal should be the center of attention so it is often the loudest signal. This can be problematic because if the vocal stands out too much there can be a sense of detachment from the rest of the mix. Delays, or reverbs, allow a vocal to be at the forefront of the mix but still sound as if it is part of the mix and the singer is performing with the band.

For this mix, two delays were used for the lead vocal. One send fed at unity to an auxiliary labeled VxDl (Vocal Delay) with an EchoBoy TelRay emulation. Saturation was turned up to produce some distortion similar to what would occur with an electric guitar amplifier. The MIDI was selected so that the tempo of the delay matched the tempo of the song and a dual echo was chosen. Both the left and right delay was a 16th note but there was still a slight variation in their time so that a stereo image was created. The feedback was set somewhat conservatively so that delay repeats did not trail too far and clutter the overall space of the song. The low cut and high cut was used to give the delayed vocal sound different frequency content than the original vocal.

VxDl remained throughout the mix so the VxDl auxiliary was automated to be appropriate for the section. The delay level was set at -14.7dB through most of the song. In the beginning of the song where the instrumentation was very sparse, the delay was too transparent so it was lowered by 12.6dB to -27.3dB. Likewise, in the bridge the delay was turned down 6.7dB to -21.4dB.
Another send named ChDl (chorus delay) was automated to turn on during choruses to help define the different sections of the song. The send was set at nominal gain and went to a corresponding auxiliary labeled ChDl. On this auxiliary, an EchoBoy delay was set to a Memory Man emulation with a dual echo eighth note delay. Again, the MIDI button was selected to match the song’s tempo and the high-cut and low-cut shaped a different timbre from that of the dry vocal signal. On this delay, the saturation was turned to its maximum to create plenty of distortion on the delayed vocal signal.

In addition to bypass automation on the ChDl send was volume automation on the ChDl auxiliary. As delay levels change, the landscape and depth of the vocal change. First, the ChDl auxiliary was set at -10.2dB. Then as the chorus played back in real-time, I moved the fader up and down to match the inflections and different intensities of the vocal. Those types of small details give a mix more life and fluidity.
A third send set at -20dB went to an auxiliary named Vrb (Reverb), which went to an auxiliary with an AltiVerb plug-in inserted. A vocal plate preset was chosen. Then the pre-delay and reverb decay time was altered to fit the song. A pre-delay of 23.58ms was derived from the same formula that was used to calculate the pre-delay for the drums. The reverb decay time was found by listening to the pulse of the song. A decay time of 1.2 seconds, roughly one half note, seemed to fit the vocal phrasing. It was not a precise calculation, but sometimes precision just is not that important. The reverb created the perception that the singer is a little farther away.

The Vrb send was automated to turn off after the introduction. The intention was to move from a somewhat live and distance environment in the introduction to an aggressive and hyper-realistic environment for the rest of the song.
Throughout the song the lead vocal level changed to maintain dominance over the rest of the mix. For the introduction and bridge the instrumentation was sparse, and as a result the dynamics were much lower than the rest of the song. The vocal was settled into the sections at -7.6dB for the introduction and then -5.7dB to fit the increased dynamics and additional instruments. The vocal then moved up another 1.3dB for the chorus. For the second verse, most of the instrumentation from the chorus remained, which called for a louder level, -5.7dB, than the first verse. The pre-chorus and remaining choruses had the same levels as the first. The only change in automation was a dip down to -7.7dB for the sparse bridge.

Lead Vocal 2 (LdVx2)

During the bridge, the vocal phrases overlap. To prevent the vocal line from being rushed for the sake of fitting the lyrics in a limited time, the phrases were staggered and recorded on two tracks. A hardware insert can only be used on one track so the Vx1 signal chain was matched as closely as possible to the Sta signal chain. The plug-in chain was almost identical to the LdVx1 plug-in chain. Just a slight bit more compression was used on LdVx2.
The Massey de-esser plug-in settings and sends were identical to LdVx1 with the exception of the ChDl send that was not used.

**Lead Vocal Double (Dbl)**

A lead vocal double was recorded for the chorus to add more fullness. On this track the Vx2 hardware insert was used. The Renaissance Vox compressor and Massey de-esser settings were the same as the LdVx1 plug-ins. The SPX, VxDl, and ChDl sends were the same as well. There were just a couple differences from the LdVx. The level was lower since the double is just meant to be a support to the lead vocal and the panning was 24% to the left so that it was not entirely masked by the lead vocal.

**Background Vocal 1 (Bg1)**

The background vocal was in unison with the last phrase of the choruses with the exception of the final chorus. The processing was very simple. A Filter Bank E6 EQ was used to fit the background vocal in with the other vocals. There was several vocals happening at the same time so generous EQ was needed. The lead vocal already had enough low-end so much of the background vocal low-end was removed. A high-pass was set at 151Hz and 440Hz was cut by 3dB. To keep the background vocal at a lower volume but still audible, 4kHz was boosted by 5dB and 10kHz was boosted by 3dB.
The background vocal was then panned hard right and sent to an auxiliary labeled BgDl (Background vocal delay) at nominal gain. The auxiliary had an Echo Boy delay inserted. On the delay the studio tape emulation was chosen, feedback was fairly short, and the delay time was a thirty-second note. The auxiliary was panned hard right so that there seemed to be two vocals singing the same part. Using the low-cut and high-cut further increased the differences in the original and delayed signal, which in turn enhanced the stereo feel.

At the end of the song, the background was more an additional vocal melody than support to the lead vocal so the delay send was muted and the panning was changed to 13% right.

Figure 3.56 McDsp Filter Bank E6 EQ on Bg1.

![Figure 3.56 McDsp Filter Bank E6 EQ on Bg1.](image)

Figure 3.57 Sound Toys Echo Boy on BgDl.

![Figure 3.57 Sound Toys Echo Boy on BgDl.](image)
Remaining Mixes

To avoid excessive repetitive language the remaining mix explanations will focus on aspects unique to the mix. The majority of the mixing techniques and the purpose behind those techniques remained the same throughout the album so plug-in settings, hardware settings, automation, and effects will be similar and in some cases exactly the same.

While there is value in comparing the similarities and differences of one mix to another to obtain a clearer idea of the purpose of a type of audio manipulation, no mix can be fixed as cornerstone. What works well for one mix may fail for another mix. As a result, specific plug-in settings, equipment settings, levels, and automation moves will be briefly stated with the assumption that the approach is the same as that expressed in (not fixed by) the description of the 4AM mix. When there is a significant deviation from approaches explained in 4AM and/or creative avenues unique to the mix at hand those differences will be discussed in detail.

Breaking Point Mix Details

Kick (K)

The inside kick microphone and sample kick were assigned to a mono auxiliary labeled “K”. A SSL Channel plug-in on the auxiliary had a 4.6dB boost at 6.8kHz and 4.01kHz. There was also an 8.4dB cut at 300Hz and a 0.5dB boost at 103Hz.
The auxiliary also had sends feeding to the DrComp key and the Space Station. The DrComp send was at unity and the Space Station send was set at -20db. The sends and their levels were copied to the rest of the drum-set tracks with the exception of the Space Station send, which was not used on the overhead or room microphone tracks. The delay on the Space Station auxiliary was changed to 21.29ms to fit the tempo of the song. (Refer to explanation of calculating pre-delays.)
For the uncompressed snare signal there was a high-pass at 65Hz, a 1dB boost at 256.2Hz, a cut of 3.4dB at 460Hz, a 7.9dB boost at 1.41kHz, and a boost of 4dB at 5kHz.

The SPL Transient Designer shortened the ring of the snare.
**Compressed Snare (SnComp)**

The compressed snare had a ratio of 2.75:1, a release at 0.36s, and gain reduction around 3dB. A 4.4dB boost was made at 8.57kHz and 4.40kHz while 550Hz was cut 3dB and 230Hz was boosted 1.6dB.

![SPL Transient Designer on Sn.](image)
For the toms, 3dB was added at 100Hz, 10.2dB was cut at 375Hz, 4268Hz was boosted by 4dB, and a boost of 4dB was made at 7885Hz.
Sample Toms (SamToms)

The processing for SamToms was exactly the same as Toms.

Overheads (OH)

An SSL Channel was used to boost 14.04kHz by 3.7dB, 3.92kHz by 3.5dB, and cut 310hz by 2dB. There was also a high-pass at 84Hz.
Rooms (Rms)

The SSL Channel was used for the rooms as well. Just like the overhead EQ, a high pass was set at 84Hz. There was a 4.9dB boost at 9.73kHz, a 2.8dB boost at 5.8kHz, and a 2.1dB cut at 310Hz.
**Tambourine (Tambo)**

A tambourine was played during the instrumental section following the acoustic guitar introduction, the choruses, and the instrumental sections following the choruses. The drum-set cymbals masked the tambourine, so it needed a boost where the cymbals were not dominating. A Filter Bank E6 EQ was inserted and 4.1dB was added at 3.83kHz to resolve the issue.

![Figure 3.65 Waves SSL Channel on Rms.](image-url)
**Bass (Bs)**

Only a D.I. signal was recorded for bass so the BsDI track was duplicated. The second track was named BsA and both tracks were assigned to an auxiliary named Bs. There was 20dB of distortion added with the punish button and a touch more was added with the drive knob. Tone was set around 2 o’clock and high-cut was set just past 1 o’clock. Style “T”, the culture vulture emulation, was selected.

![Figure 3.66 McDsp Filter Bank E6 on Tambo.](image)

![Figure 3.67 Sound Toys Decapitator on BsA.](image)
Next the Bs hardware insert was applied to the Bs auxiliary. The settings for the Bs hardware insert were static throughout the album. (Refer to Universal Mix Settings: Hardware inserts). A Waves L1 limiter was inserted with the threshold at -4.4dB, the out ceiling at -7.1, and the release at 1 second.

![Waves L1 Limiter on Bs.](image)

Lastly, a Filter Bank P6 EQ was used to cut excessive low frequencies at 100Hz by 1dB and low middle frequencies at 249.6Hz by 2.8dB. A 3dB boost was made at 1.38kHz as well as a 4.1dB boost at 7.22kHz.
The rhythm guitars were first processed through the API hardware chain. The API EQs had a 4dB boost at 5kHz, a 2dB boost at 800Hz, and a 2dB cut at 300Hz.

After the global characteristics of the rhythm guitars were established with the API hardware signal chain, a Filter Bank E6 EQ was used for fine-tuning. The EQ had a high-pass at 80.7Hz, a 1.5dB boost at 357Hz, a 1dB boost at 2.39kHz, and a 1.5dB at 9.14kHz.
Hook Lead (HkLd)

HkLd was a melodic guitar part that played during the instrumental following the acoustic intro and following the first and second chorus. Most of the processing was done with the Act 1 hardware insert. But, the melodic guitar line still needed some subtle changes to stand out among the dense rhythm guitars. A high-pass was set at 90.4Hz and 9.15kHz was boosted by 1.4dB.
The guitar line was also sent to an auxiliary labeled GtDL (guitar delay) with an Echo Boy delay inserted. The delay created a wide stereo effect that momentarily switched the focus of the mix to the guitar part.

*Pre-chorus Melody 1 (PreMldy1)*

PreMldy1 was a simple melody heading into the chorus. The track also contained a
repeating melody that built in dynamics during the bridge. The TLA hardware insert was used to make the most substantial changes. PreMldy1 also had a 3dB boost at 9.15kHz.

![Image](image.png)

*Figure 3.74 McDsp Filter Bank E6 EQ on PreMldy1.*

The melody also had a volume fade during the bridge to build dynamics.

**Bridge Melody (BrMldy)**

A second repeating melody that built dynamics during the bridge ran through the second TLA 1-2 hardware insert.

**Chorus Lead 1 (ChLd1)**

The chorus lead guitar melody was first processed with the Act 2 hardware signal chain. The chorus lead guitar line was further manipulated with a Filter Bank E6 EQ. A high-pass was set at 87.9Hz, a boost of 4dB was made at 2.04kHz, and another boost of 3.1dB was made at 6.97kHz.
Chorus Lead 2 (ChLd2)

A second lead guitar line introduced after the build into the repeated chorus at the end of the song was shaped with a Filter Bank E6 EQ. The EQ was given a high-pass at 98.5Hz, a boost of 3.8dB at 1.43kHz, and a boost of 5.7dB at 6.67kHz.

Figure 3.75 McDsp Filter Bank E6 EQ on ChLd1.
During the one bar dynamic build leading into the repeated chorus at the end of the song there were two guitars playing ascending octaves. One take, Gt16, was panned hard left. The other take playing that same guitar line, Gt17, was panned hard right. Both takes were then assigned to a stereo auxiliary panned hard right and left and labeled “Build”. On the Build auxiliary, the Gt 1-2 hardware insert was used for the majority of the sonic shaping. The Gt 1-2 hardware insert was then followed by a Filter Bank E6 EQ that had a high-pass at 90.4Hz, a cut of 2dB at 293Hz, a boost of 3.8dB at 2.12kHz, and a boost of 4.6dB at 6.74kHz.

Figure 3.76 McDsp Filter Bank E6 EQ on ChLd2.

Build

During the one bar dynamic build leading into the repeated chorus at the end of the song there were two guitars playing ascending octaves. One take, Gt16, was panned hard left. The other take playing that same guitar line, Gt17, was panned hard right. Both takes were then assigned to a stereo auxiliary panned hard right and left and labeled “Build”. On the Build auxiliary, the Gt 1-2 hardware insert was used for the majority of the sonic shaping. The Gt 1-2 hardware insert was then followed by a Filter Bank E6 EQ that had a high-pass at 90.4Hz, a cut of 2dB at 293Hz, a boost of 3.8dB at 2.12kHz, and a boost of 4.6dB at 6.74kHz.
An additional lead guitar played the upper register of what the “Build” guitars played. Gt15 was assigned to a mono auxiliary and that auxiliary was labeled BuildLd. The auxiliary was then panned 23% to the right to complete a left, center, right stereo field with the “Build” guitars. It was set a little off center so that it would not compete with the lead vocal. Next, a Filter Bank E6 EQ had a 98.5Hz high-pass engaged, a 3.8dB boost at 1.43kHz, and a 5.7dB boost at 6.67kHz.

Build Lead Guitar (BuildLd)
Acoustic Guitars (Ags)

The acoustic guitars played throughout the song. AgBod1 and AgBod2 were panned hard left and right. Then both tracks were assigned to a stereo auxiliary labeled Ags that was also panned hard left and right. On the Ags auxiliary the processing began with the GME 1-2 hardware insert.

More compression was added with a Renaissance Axx plug-in. The threshold was set at -7.9dB and there was about 3 - 6dB of gain reduction.
The last insert in the signal chain was a VEQ4. A high-pass was set at 82Hz, a 3dB cut was made at 100Hz, a 2.3dB cut was done at 330Hz, 2.7kHz was boosted 3.6dB and 6.8kHz was boosted 2.3dB.

Upright Piano (UpPno)

Only a SSL Channel plug-in was used on the piano. Very slight compression was used on the piano. The ratio was 2.9:1 and the release was 0.40s. There was occasionally 3dB of gain reduction. Piano often needs severe EQ to be heard in a dense mix. Rock pianos in particular are very bright. A 7.6dB boost was done at 8.80kHz for presence and 2.69kHz was boosted to bring out the attack. A 2.6dB cut reduced muddiness and a 2.1dB boost gave the piano fullness.
**Lead Vocal (Vx)**

The lead vocal first ran through the Sta signal chain. Additional compression was added with a Waves Renaissance Vox plug-in. Gain reduction hovered around 3dB and at it’s maximum hit 6dB. The automatic make-up gain produced by the compressor was peaking the channel so the output gain was reduced to -2.2dB.

![Figure 3.81 Waves SSL Channel on UpPno.](image)
A Massey De-Esser lessened consonant harshness in the vocal by lowering around 6kHz about 6dB.

The lead vocal had sends feeding to an auxiliary labeled VxDl and another auxiliary labeled ChDl. On both auxiliaries an Echoboy delay was inserted but each delay had it’s own

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**Figure 3.82 Waves Renaissance Vox on Vx.**

**Figure 3.83 Massey De-Esser on Vx.**
settings. For the VxDI Echo Boy, the MIDI button was selected to synchronize the delay with the song’s tempo. An Echo Plex emulator and the Dual Echo delay mode were selected. Both echoes were set at a sixteenth note and the low cut and high cut were set around fifty percent. The feedback was set fairly low to keep the delay from interfering with the dry vocal phrases that followed.

VxDI was used for the entire song but the auxiliary return level was 2.5dB lower during the verses than it was during the choruses so that it was not readily noticeable as an individual effect.

The ChDI Echo Boy was made to contrast with the VxDI. A Memory Man emulation was selected and the Midi button was selected to match the delay timing with the song’s tempo. Saturation was turned to the maximum to enhance the qualities of the Memory Man. A ping-pong delay mode was selected and both the ping and the pong was a sixteenth note. The groove and feel were increase slightly to allow the delay to fall slightly behind the beat. The low-cut was at approximately 3 o’clock and the high-cut was around 1 o’clock to create an even more severe low-quality radio sound.

The ChDI send was automated to turn on only during the Choruses. The only volume automation on the ChDI auxiliary was at the end of the second chorus. The auxiliary was turned up so the delay stood out for just a moment and a repeat of the vocal could be heard slowly fading out.
Volume automation ensured audibility and clarity of the vocal throughout the song as well as dynamic compatibility with the instruments. For each main section of the song; verse, pre-chorus, chorus, the vocal moved up 1dB. When the song came to the bridge, the vocal was at it’s quietest to achieve a little more intimacy.

**Lead Vocal Double (Dbl)**

Another vocal that sang in unison with the lead vocal during the chorus was processed through the Vx1 hardware insert. The vocal first went through a LA3A set to compress a maximum of 5dB. The first LA3A was routed into another LA3A set to limit a maximum of 7db. The last piece of the signal chain was a Rupert Neve 9098 EQ. The Neve Eq attenuated 150Hz by 3db and 300Hz by 3db. At 4.5kHz and 21Khz there was a boost of 3dB. The 300Hz cut and 4.5kHz boost both used a Q of .65.

**Burning All The Pages**

**Acoustic Guitar (Ag)**

AgBod1 was panned hard right and AgBod2 was panned hard left then both were assigned to a stereo auxiliary labeled Ags. The first processing applied was the GME 1-2 hardware insert.

Renaissance Axx, a compressor, was next in the signal chain. Threshold was set at -10.4 so that there were approximately 3 to 6dB of gain reduction. The plug-in has automatic make-up gain so the output was reduced to -6.9dB to prevent the channel from peaking. The attack was set at 5.00 (specific time constants are not stated in the manual so what 5.00 refers to is unknown) (Waves Audio Ltd.).

![Figure 3.85 Sound Toys Echo Boy on ChDl.](image)
The acoustic guitars were still too dynamic but increased compression was becoming perceptible so another Renaissance Axx was inserted. The threshold was at -7.9dB and attack was again at 5.00. The second compressor applied an additional 3dB of gain reduction, which gave the acoustic guitars the last bit of dynamic consistency they needed to fit in the mix.

Figure 3.86 Waves Renaissance Axx on Ags.
A VEQ4 had a high pass at 82Hz, a 2.5dB cut at 100Hz, a 3.4dB cut at 330Hz, a boost of 3.9dB at 2.7kHz, and a 3.5dB boost at 6.8kHz.

A little more attention was given to the acoustic guitars by writing volume automation in real time. When transitioning from the verse to the chorus, the volume was moved up about 2dB. During the bridge, the volume was gradually increased until the final chorus, which was then held at the same average volume.
Piano (Pno)

The piano had a 5dB boost at 8.8kHz, a 5.1dB boost with a Q of .10 at 2.69kHz, a 6.4dB cut with a Q of 2.50 at 300Hz, and a 2.1dB cut at 127Hz.

A send fed to the Space Station at -20.5dB and returned on the Spc auxiliary with the short delay set at 13ms.

Figure 3.89 Waves SSL Channel on UpPno.
There was also a fair amount of volume automation on the piano written in real-time to fit it appropriately in each section. In the first verse, certain quiet chords were brought up so they were not lost in the mix. In the bridge, it was brought down lower than it was in the rest of the song to settle with the rest of the instruments that had low dynamics in that section.

**Lead Vocal 1 (Vx1)**

The majority of the EQ and compression was done through the Sta hardware signal chain. Additional compression was done with a Renaissance Vox compressor that had a threshold at -5.3dB to give the vocal another 6dB of gain reduction. The output gain was reduced by 0.4dB to prevent peaking.
A Massey De-Esser lowered harsh consonants about 6dB around 6kHz.

The lead vocal had a send feeding at -21.3dB to an auxiliary labeled VxDI with an EchoBoy delay inserted. A Memory Man delay emulator was chosen and the MIDI button was
selected to match the delay times with the songs tempo. A single echo was delayed a dotted sixteenth note with very little feedback so the delay was not incredibly noticeable but developed the depth of the vocal. A low and high cut was used to give the delay a mid-range focus.

The VxDl had volume automation written throughout the song to mimic inflections of the vocals. Most notably, the delay was boosted about 8dB in the interlude following the first chorus. Kevin had stepped away from the microphone to sing the section, so the delay enhanced the perception of distance.

Another send was automated to turn on during the choruses and feed at -25dB to an auxiliary named ChDl. On the auxiliary, there was an Echo Boy delay with a Memory Man style and the saturation turned to the maximum. A ping-pong delay was selected. The Ping had a sixteenth note delay and the Pong had an eighth note delay. Feedback was set around 10 o’clock so that it had a moderately long decay. The low cut was set around 3 o’clock and the high cut was around 1 o’clock.
A Filter Bank E4 EQ had a high pass at 254.6Hz, a 9dB cut at 242.4Hz, a 3dB boost at 1kHz, and a 3dB boost at 5kHz. The EQ altered the delay so that it was focused in a high-end range that is easily audible.

The ChDl did have volume automation other than turning on during the choruses, but it was fairly minimal. It had a significant boost in the interlude following the first chorus to further develop the automation that had been done on VxDl in the same section.
The final send fed at -20dB to an auxiliary named Vrb that was set at -9.5dB. On the auxiliary was an Altiverb plug-in inserted with a decay of 2.4s. The reverb softened the vocal and gave it a deeper and wider sound.

![Figure 3.96 Altiverb 7 on Vrb.](image)

Real-time volume automation was written to the vocal. The choruses were about 1dB louder than the verses. Most notably, the interlude between the first chorus and second verse was brought down about 5dB. Vocal breaths also had major dips of about 10dB to avoid distraction.

**Lead Vocal 2-Bridge Lead Vocal (Vx2)**

The lead vocal during the second chorus overlapped the lead vocal during the bridge so they were split onto different tracks. The vocal went through the Vx1 hardware chain. After the hardware insert was a Renaissance Vox compressor. The compressor had a -9.1dB threshold and gave the vocal about 6dB more of gain reduction.
Next was a Massey De-Esser with the same settings as the De-Esser on Vx1.

Vx2 also sent to VxDl at -21.3dB and had volume automation to lower a few loud syllables.

**Background Vocal 1 (Bg1)**

Bg1 was a female background vocal. It was panned 15% to the left so it would not collide with the lead vocal. The first processing was done with a Filter Bank E6 EQ. There was a high pass at 66.1Hz, a 1.5dB cut at 271.4Hz, a boost of 4.2dB at 3.54kHz, and a 2.5dB boost at 11kHz.
The Renaissance Vox compressor set with a -5.3dB threshold gave the vocal about 3dB - 5dB of compression. The output gain was brought down to -0.4dB to compensate for excessive automatic make-up gain.

Figure 3.98 McDsp Filter Bank E6 EQ on Bg1.

Figure 3.99 Waves Renaissance Vox on Bg1.
Following the compressor was a Massey De-Esser with the same settings as the De-Esser on Vx1.

Bg1 had a send feeding to VxD1 at -21.3dB and another send set at -19.5dB feeding to ChD1 to give it the same space as the lead vocal. Automation on this track was minimal since it only happened during the choruses. But, it did have a few words brought up or lowered to ensure dynamic consistency and prevent it from overpowering the lead vocal at any time.

**Background Vocals (Bgs)**

Bg6 was panned hard left; Bg7, which was a double of Bg6 was panned hard right, and Bg8 was panned left 16%. All three vocals were assigned to an auxiliary named Bgs so that they could be processed as one sound. First, a Filter Bank E6 EQ had a high pass at 123.7Hz, a 2dB cut at 260.8Hz, a 5.6dB boost at 3.15kHz, and a 1.9dB boost at 9.42kHz.

Next, a Renaissance Vox compressor had a threshold of -5.3dB and compressed the background vocal around 3dB - 5dB.

Figure 3.100 McDsp Filter Bank E6 EQ on Bgs.
The background vocals fed to the VxDl auxiliary at -21.3dB and the ChDl auxiliary at -19.5dB.

Cast In Different Melodies

Kick (K)

A SSL Channel was used to boost the kick at 6.80Hz by 4dB, and 4.01kHz by 3.2dB with a Q of 0.10. A 7dB cut with a Q of 0.10 was made at 370Hz and a 0.5dB boost was done at 103Hz.
The kick was sent to the Space Station at -25dB with a 25ms delay on the return auxiliary.

Another send keyed the Focusrite compression at unity, which was copied to the Sn,
SnComp, Toms, and SamToms tracks as well.

**Snare** (*Sn*)

An Spl transient designer was used to give the snare more crack and less ring.

![SPL Transient Designer on Sn](image)

**Figure 3.104** SPL Transient Designer on Sn.

Next in the chain was a Filter Bank E6 EQ with a high pass at 65Hz, a 2.7dB boost at 256.2Hz, a 4.1dB cut at 392.7, an 8dB boost at 1.3kHz, and a 4.6dB boost at 5kHz.
The Snare track was then sent to the Space Station at -15.8dB, which was the same level that the compressed snare fed to the Space Station.

*Compressed Snare (SnComp)*

A SSL Channel plug-in gave the snare about 3db of compression/gain reduction with the release at 0.36s and a 2.7:1 ratio. The snare was also boosted at 8.57kHz by 4.5dB and 4.40kHz by 4.7dB with a Q of 0.10. A 3dB cut was done at 550Hz with a Q of 0.10 and 280Hz was boosted 4.6dB.
Toms

A Renaissance EQ boosted the toms 3dB at 100Hz, cut 10.2dB at 375Hz, boosted 6dB at 4268Hz, and boosted 4dB at 7885Hz.
Both tom tracks fed to the Space Station at -21.3dB

*SamToms*

The sample toms were treated exactly as the original toms.

*OH*

An SSL Channel was used to make a high pass at 84Hz, a 3.71dB boost at 14.04kHz, a 3.5dB boost at 3.92kHz, and a 5.4dB cut at 310Hz.
An SSL Channel for the room microphones was set similarly to the overheads. There was a high pass at 84Hz, a 6.9dB boost at 2.71kHz with a Q of 0.10, a 1.6dB cut at 310Hz with a Q of 0.10, and a 2.1dB boost at 103Hz.

**Rms**

An SSL Channel for the room microphones was set similarly to the overheads. There was a high pass at 84Hz, a 6.9dB boost at 2.71kHz with a Q of 0.10, a 1.6dB cut at 310Hz with a Q of 0.10, and a 2.1dB boost at 103Hz.
Drum 2 (Dr2)

A drum overdub that played before the first and second chorus was recorded with overhead and room microphones. The overhead and room microphones were then assigned to an auxiliary named Dr2. On the auxiliary was a SansAmp plug-in that distorted the drums and gave them a low fidelity sound.
An Echo Boy delay was automated to turn on at the last drum hit so that the drums had a decaying echo. An EchoPlex style was chosen, the MIDI button was selected so that the delay was in time with the song’s tempo, and the echo time was set at a half note.

**Tambourine (Tambo)**

The tambourine was treated with a Filter Freak 1 that was set to a band-pass filter. This focused the track in a narrow frequency range so that it could be heard among the other instruments but not take up too much of the frequency spectrum.
Bass (Bs)

The bass was recorded directly into a pre-amplifier. That track was duplicated and a Decapitator plug-in was used to create bass amplifier characteristics. Style “T”, the Culture Vulture emulator, was chosen. The punish button was turned on to add 20dB of distortion and the drive control introduced a little more distortion. The tone control was set around 2 o’clock to drastically brighten the bass and the high cut was set in the 15kHz to 17kHz range to reduce static-like distortion created by the plug-in.

![Figure 3.113 Sound Toys Decapitator on BsA.](image)

The BsDI and BsA tracks were assigned to a mono auxiliary. The Bs hardware insert was first in the chain followed by a L1 Limiter. The Limiter threshold was at -4.4dB and gave the loudest sections of the bass about 3 to 5dB of reduction.
The bass sound was shaped further with a Filter Bank P6 EQ. At 104.8kHz there was a 1dB cut along with a 3.5dB dip at 259.6kHz. There was also 1.4dB boost at 1.9kHz, a 3.7dB boost at 1.27kHz, and a 3.1dB boost at 9.52kHz.

The sub-sonic frequencies in the bass were overpowering the mix so a Filter Bank F2 was used to make a high pass at 30.9Hz. The bass also had some high frequency noise so a low pass
was set at 18kHz.

![Figure 3.116 McDsp Filter Bank F2 on Bs.](image)

**Clean guitar 1(Cl1)**

Cl1 was a clean guitar melody played during the verses. Both microphones used to record the part were bussed to a mono auxiliary, which was panned hard left and sent through the Tla signal chain. After the hardware chain, another 6dB of compression/gain reduction was done with a Renaissance Axx compressor. The attack was set at 5.00 and the threshold was at -9.5dB. The gain was brought down to -6.3dB to avoid peaking.
Frequency carving continued with a Filter Bank E6 EQ. A high pass was engaged at 66.1Hz with a 4.4dB boost at 1.22kHz and a 3dB boost at 9.15kHz.

Figure 3.117 Waves Renaissance Axx on C1.

Figure 3.118 McDsp Filter Bank E6 on C1.
Dynamics in the second verse were slightly higher than the first verse, so the second verse was automated to be about 2dB louder than the first.

**Clean Guitar 2 (Cl2)**

Cl2 was a single guitar note sustain from the beginning of the song until the first chorus. A 3dB boost at 9.15khz was done with a Filter Bank E6 EQ.

![McDSP Filter Bank E6 on Cl2](image)

Figure 3.119 McDsp Filter Bank E6 on Cl2.

A PanMan plug-in continuously panned the guitar drone back and forth. The offset was directly in the center so the track panned to the left and right channel equally. Width was a hair past half way and Smoothing was at the softest setting so that the change in panning direction had a smooth transition.
Clean Guitar 3 (Cl3)

Cl3 was another guitar melody played during the verses. The two microphones / tracks used to record the guitar part were assigned to a mono auxiliary and panned hard right. The Tla2 hardware signal chain was the first treatment of the guitar part. A Renaissance Axx compressor was next. The threshold was at -9.5dB and the attack was at 5.00. The compressor introduced another 6dB of gain reduction. The output gain was turned down to -6.3dB since the automatic make-up caused the track to peak.
A Filter Bank E6 EQ was used to make a high pass at 66.1Hz, a 4.4dB boost at 1.22kHz, and a 4.9dB boost at 6.16kHz.

![Figure 3.122 McDsp Filter Bank E6 on Cl3.](image)

**Clean Guitar 4 (Cl4)**

A clean guitar melody panned hard left played through the first two choruses. This track utilized the Act1 hardware insert. After the insert, a Filter Bank E6 EQ was used. It had a high pass at 90.4Hz, a 3.5dB boost at 1.74kHz, and a 2.3dB boost at 5.97kHz.
Clean Guitar 5 (Cln5)

A second clean guitar melody was played during the choruses. The Cln5 auxiliary was panned hard right. The Act 2 hardware insert shaped the overall guitar sound. A Filter Bank E6 EQ was used to make further adjustments. A high pass was set at 90.4Hz, a 3.5dB boost was made at 1.74kHz, and a 2.3dB boost was done at 5.97kHz.

![Figure 3.123 McDsp Filter Bank E6 EQ on Cl4.](image-url)
Main Distorted Electric Guitars (Big)

Dr6 was panned hard left and Dr7 was panned hard right. Those tracks were then assigned to a stereo auxiliary named Big, which was run through the API hardware signal chain. The API EQs had a 2dB boost at 5kHz, a 2dB boost at 800Hz, and a 2dB cut at 300Hz.

Then a Filter Bank P6 EQ was inserted to make a 1.4dB dip at 357.8Hz along with a 3dB
boost at 2.39kHz and a 2.2dB boost at 9.15kHz.

Figure 3.126 McDsp Filter Bank P6 on Big.

**Acoustic Guitar (Ag)**

The acoustic guitar was panned 61% to the right and treated with the GME 1 hardware chain. More EQ was done with the VEQ4. A high pass was at 82Hz with a 2.5dB cut at 100Hz, a 3.4dB cut at 330Hz, a 3.9dB boost at 2.7kHz, and a 3.5dB boost at 6.8kHz.

Figure 3.127 Waves VEQ4 on Ags.
A Renaissance Axx compressor gave the acoustic guitar about 3dB - 5db more of gain reduction. The threshold was at -7.8dB and the attack was at 5.00. The gain was left at 0dB.

The acoustic guitar was the only guitar that played throughout the song so it had more volume automation than any of the other guitar tracks.

**UpPno**

The piano was brightened with a SSL Channel plug-in. There was a 7.6dB boost at 8.8khz, a 6.9dB boost at 2.71kHz with a Q of 0.10, 1.6dB cut at 310Hz with a Q of 0.10, and a 2.1dB boost at 103Hz. Compression was a 2.9:1 ratio with a 0.40 release, and a -1.5dB threshold. There was a rare gain reduction of 3dB.

![Figure 3.128 Waves SSL Channel on UpPno.](image)

Quite a bit of automation was written to the piano track. Most of the automation was written in real-time by riding the fader. In the introduction, the piano starts around -3dB. When the drums, guitars, and bass entered for the verse, the piano was brought up about 1.5dB to keep it at the forefront of the mix. It was then momentarily brought down roughly 1dB for a breakdown, and then returned to around -1.5dB for the chorus. The pattern was repeated in the second verse and chorus with slight changes to make individual chords louder or quieter. The performance in the bridge was played quietly and was rather close to the intended dynamic of the section. It was brought down about 0.5dB from the chorus just to subtly accentuate the
performance. For the end of the song, the guitars and drums are louder than the rest of the song. The piano was brought up to -1dB to still be one of the main focuses of the mix.

**Lead Vocal 1 (Vx1)**

Most of the lead vocal sound was carved with the Sta hardware signal chain. Another 6dB - 8dB of gain reduction was done with a Renaissance Vox compressor. The output gain was reduced to -1.6dB to prevent the channel from peaking.

![Figure 3.129 Waves Renaissance Vox on Vx1.](image)

A Massey De-Esser reduced sibilance by 6dB around 6kHz.
The lead vocal was sent to an auxiliary named VxDl at -22.2dB. On the auxiliary, an Echo Boy delay was inserted. The Memory Man style was chosen and the MIDI button was selected so that the delay times fit with the song tempo. A single echo with a short feedback was delayed a dotted sixteenth note.

Through most of the song, the VxDl auxiliary was at -14.5dB. The introduction, breakdown, and bridge were instrumentally sparse and the vocal was more naked. The delay was brought down 5dB in those sections so that the effect was not obvious.
The ChDl was high passed at 254.6Hz with a Filter Bank E4. Another 9dB was cut at 242.4Hz while 1kHz was boosted 3dB and 5kHz was boosted 3dB.

The vocal also had a send at -20dB that turned on during the choruses and fed to an auxiliary labeled ChDl. On the auxiliary, an Echo Boy delay had a Memory Man ping-pong delay with the MIDI button selected. Both the ping and the pong were a quarter note delay with the feedback a little longer than the VxDl Echo Boy around 9 o’clock. The low cut was just under 3 o’clock and the high cut was around 1 o’clock.
The ChDl Echo Boy length on the voice fit the song, but long repeats were distracting at the end of the first and second chorus. A fade out on the auxiliary solved the issue.

The vocal had a send feeding to auxiliary named Vrb at -20dB. On the auxiliary was an Altiverb reverb with 2.4s decay. The auxiliary was brought down to -8dB.

Finally, the vocal had volume automation to match the dynamic peaks and dips of the song. For the introduction when there was only a piano, vocal, and guitar drone, the vocal was at -7.7dB. When more instruments were introduced in the verse, the vocal level moved up 1dB. When the song transitioned into the chorus, the drums were played louder and there was an
overall higher energy. The dynamic changes called for another 1.5dB volume increase of the vocal. When the song returned to the verse, dynamics were slightly higher than the first verse. Setting the second verse 0.5dB louder than the first verse was appropriate for the higher intensity. The second chorus was identical to the first in volume, and the bridge vocal was brought down to the introduction volume of -7.7dB, since instrumentation was again only piano. The end of the song jumped in dynamics so the vocal was set 1dB louder than the chorus at -4.3dB.

**Background Vocal 1 (Bg1)**

Two identical female chorus backgrounds were put on a stereo track and panned hard left and right. The vocals were compressed with a Renaissance Vox compressor. The threshold was at -10.9dB so that there were about 6 to 8dB of gain reduction. The output gain was turned down to -5.3dB to avoid the channel from clipping.

![Figure 3.135 Waves Renaissance Vox on Bg1.](image)

The vocal had a very wide dynamic range and needed further compression. But, too much compression with one unit often becomes transparent. A second Renaissance Vox compressor had a threshold at -15.4dB with around 10dB of gain reduction. The output gain was brought down to -3dB to reduce some of the automatic make-up gain.
After the compressors, a Filter Bank E6 EQ was used to shape the frequency content of the background vocals around the lead vocal. First on the EQ was a high pass at 107.3dB, which was followed by a 2.7dB at 260.8Hz. There was also a 5.2dB boost at 2.8kHz and 4.5dB boost at 9.42kHz.

Figure 3.136 Waves Renaissance Vox on Bg1.
Bg1 fed to VxDl at -22.2dB and ChDl at -23dB. The volume averaged around -9.7dB and had a +/- deviation of 0.5dB on phrases that were too quiet or too loud.

**Background Vocals (Bgs)**

Bg2 through Bg13 created a pseudo female choir toward the end of the song. All of the backgrounds were stereo tracks that were assigned to a stereo auxiliary labeled Bgs. All of these background vocals were processed as one sound by way of the Bgs auxiliary. First in the chain was a Renaissance Vox compressor that had a threshold of -8.9dB and an output gain of -4.8dB. The compressor provided about 6dB of gain reduction.

![McDsp Filter bank E6 EQ on Bg1.](image)
The second insert in the chain was a Filter Bank E6 EQ with a high pass at 142.6Hz and 3.8dB dip at 260.8Hz. There was also a 7.2dB boost at 4.15kHz and a 4.5dB at 9.42kHz.

Figure 3.138 Waves Renaissance Vox on Bgs.

The second insert in the chain was a Filter Bank E6 EQ with a high pass at 142.6Hz and 3.8dB dip at 260.8Hz. There was also a 7.2dB boost at 4.15kHz and a 4.5dB at 9.42kHz.
The Bgs auxiliary fed to ChDl at -20dB. Since there were so many vocals creating a choir sound, the breaths became very prominent. To fix the problem, there was a 10dB dip in volume where the breaths happened.

**Additional Background Vocals (Cure)**

Four additional female background vocals sang in unison with the lead vocal for a short section toward the end of the song. The four background vocals were put on two stereo tracks and assigned to a stereo auxiliary named Cure (referring to the lyric sung).

Two Renaissance Vox compressors were used in series. The first had a threshold of -8.9dB, an output gain of -4.8dB and compressed the vocal up to 8dB.
The second Renaissance Vox compressor gave the vocal around 10dB of gain reduction with the threshold set at -15.4dB. The output gain was brought down to -3dB.  

Figure 3.140 Waves Renaissance Vox on Cure.  

Figure 3.141 Waves Renaissance Vox on Cure.
A Filter Bank E6 EQ high passed the background vocals at 107.3Hz. A 2.7dB cut was made at 260.8Hz while a 5.2dB boost was done at 2.8kHz and a 4.5dB boost was made at 9.42kHz.

![Filter Bank E6 EQ](image)

Figure 3.142 McDsp Filter Bank E6 EQ on Cure.

The Cure auxiliary sent to ChD1 at -2dB.

*Love Or Lust*

*Kick (K)*

The sample kick and the kick recorded with an Audix D6 were assigned to a mono auxiliary named K. The kick was then processed with a SSL Channel. There was a 4dB boost at 6.8kHz, a 3.2dB boost with a Q of 0.10 at 4.01kHz, a 7dB cut with a Q of 0.10 at 370Hz, and a 0.5dB boost at 102Hz.
The kick had a send feeding to a Space Station at -20.5dB. The Space Station signal returned on an auxiliary named Spc with a short delay set to 23.44ms.

Figure 3.143 Waves SSL Channel on K.

Figure 3.144 Short Delay II on Spc.
Another send fed the Focusrite key compression at unity. The compression key send was copied to the tracks named Sn, SnComp, Toms, SamToms, OH, and Rms. Lastly, the kick drum was brought down 1dB in the first pre-chorus to suit the calmer mood of that portion of the song.

_Snare (Sn)_

The snare top microphone, bottom microphone and sample snare was assigned to a mono auxiliary labeled Sn and another mono auxiliary labeled SnComp to provide was parallel processing. A SPL Transient Designer increased the attack of the snare and shortened the ring.

![Figure 3.145 SPL Transient Designer on Sn.](image)

The snare was then brightened with a Filter Bank E6 EQ. The EQ had a high pass at 65Hz, a 0.7dB boost at 256.2Hz, a 3.1dB cut at 460Hz, a 7.6dB boost at 1.25kHz, and a 5dB boost at 5kHz.
The Sn track had a send feeding to the Space Station at -21.3dB. The send was copied to SnComp, Toms, SamToms, OH, and Rms.

**Compressed Snare (SnComp)**

The parallel snare signal had a compression ratio of 2.7:1, with a -13dB threshold, a 0.36s release, and about 3dB of gain reduction. The SSL EQ was used as well. A 4.5dB boost was made at 8.57kHz along with a 4.7dB boost at 4.40kHz with a Q of 0.10. At 550Hz there was a 3dB cut with a Q of 0.10 along with a 4.6dB boost at 280Hz.

![Figure 3.146 McDsp Filter Bank E6 EQ on Sn.](image)
The toms were treated with a Renaissance EQ. There was a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.

Figure 3.147 Waves SSL Channel on SnComp.

**Toms**

The toms were treated with a Renaissance EQ. There was a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.
Sample Toms (SamToms)

The sample toms had the same treatment as the original toms.

OH

A SSL Channel was inserted on the drum overhead track. The overheads were given an 84Hz high pass, a 3.7dB boost at 14.04kHz, a 3.5dB boost at 3.92kHz, and a 5.4dB cut at 310Hz.
Drum Rooms (Rms)

The drum room microphones were treated with a SSL Channel as well. A high pass was made at 84Hz along with a 4.1dB boost at 10.57khz and a 5.3dB boost at 4.5kHz with a Q of 0.10. Lastly, a 4.4dB cut was made at 310Hz with a Q of 0.10.

Figure 3.149 Waves SSL Channel on OH.
Drum Rooms 2 (Rms2)

The room microphones had a send feeding at unity to an auxiliary named RmDly with an Echo Boy delay inserted. The Echo Boy style was set to Space Echo and the MIDI button was left off so the delays were more random. The delay rhythm was set to an eighth note, the feedback was at zero and the low cut was around 10 o’clock. The shape was set to decay so that the delays decreased in volume linearly and the repeats were set to 10. This produced a sort of slap delay. However, it was more of an unconventional effect as opposed to a space developing effect.
Drum Room Reverse (RmRev) and Reverse Cymbal (RevCym)

Drum room microphones were processed with the reverse audio-suite plug-in to be incorporated in the song introduction. A cymbal was processed with the reverse audio-suite plug-in and placed before the first two choruses. Neither track had further treatment in Pro Tools.

Bass (Bs)

The bass D.I. track was duplicated and named BsA. The BsA track had a Decapitor plug-in inserted. Style “T” was chosen, the punish button introduced 20dB of distortion, and the drive control gave the track further distortion. The tone knob was set around 2 o’clock to significantly brighten the bass and the high cut knob reduced some high frequency distortion.
The BsDI and BsA tracks were assigned to a mono auxiliary named Bs, which had the Bs hardware chain as the first insert. The bass was played in a register that did not produce many sub-harmonic frequencies that could be heard. Furthermore, the different registers were played during the verses and choruses, which called for different treatments. During the choruses, the bass was played in a high register that caused the dynamics to drop off when they need to be at its pick. Maxx Bass solved this problem by adding sub-harmonics with a center frequency of 80Hz. A high pass was set in the harmonic section so that there was not excessive low harmonics created. Very low frequencies can also be extremely dynamic so the new sub-harmonics were compressed at a 1.5:1 ratio. The Maxx Bass signal was turned down to -17.7dB to avoid low frequencies from drowning out the rest of the mix. The plug-in was automated to turn on only during the choruses and bridge.
The Maxx Bass used during the verses had essentially the same settings. However, the guitar sounds during the verses had far less low frequency content so bringing the Maxx Bass signal to -10dB suited the sections well. This Maxx Bass plug-in was automated to turn on only during the verses.

Even with the Maxx Bass high pass engaged, there was still excessive extremely low frequency content. A Filter Bank E6 EQ had a 30.7Hz high pass to fix the issue. There was also
a 1dB cut at 90.8Hz, a 4.5dB cut at 2.36.7Hz, a boost of 2.5dB at 3kHz, and a 3.5dB boost at 5kHz.

**Figure 3.155 McDsp Filter Bank E6 EQ on Bs.**

**Clean Guitar 1 (Cl1)**

Clean guitar 1 played through the first verse up until the first chorus. The Cl1 auxiliary was panned hard left. The Tla 1 hardware insert was used to make the major changes to the track. After that was a Filter Bank E6 EQ, which had a 72Hz high pass, a 1.6dB dip at 133.9Hz, a 0.8dB dip at 156Hz, a 3db boost at 9.15kHz, and a 1.1dB boost at 16kHz.
When the guitar changed back and forth between muted picking and open picking, the audibility fluctuated. For the muted picking section, the volume was set at -8dB. When the part shifted to open picking, the volume was brought down 3dB. The same automation move was made every time the style of picking changed.

**Clean Guitar 2 (Cl2)**

Clean guitar 2 was a second guitar that played during the first verse and stopped when the first chorus entered. The Cl2 auxiliary was panned hard right and the T1a 2 hardware insert was applied. Further EQ was done with a Filter Bank E6. The EQ had a high pass at 72Hz, a 1.6dB cut at 133.9Hz, a 0.8dB cut at 156Hz, a 3dB at 9.15kHz, and a 1.1dB boost at 16kHz.

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*Figure 3.156 McDsp Filter Bank E6 EQ on Cl1.*
Most of Cl2 was played exactly as Cl1 was played so for the most part the same automation was used. It was not until the pre-chorus that the guitars played different parts. Cl2 played a high melody, which was brought up 2dB to play a more important part over the simple chords that Cl1 played.

*Tapping Guitar/ Clean Guitar 3 (Tap/ Cl3)*

Tap was an intermittent guitar melody played during the verses that was panned hard right. Treatment started with the Act 1 hardware insert and was continued with a Filter Bank E6 EQ. The EQ had an 87.9Hz high pass, a 2.3dB boost at 2.58kHz, and a 2dB boost at 8.61kHz.
All of the guitars in the first verse were a clean tone so the tapping guitar melody needed to be a little quieter at -14.2dB than it was in the second verse at -13dB with all of the heavy gain guitars.

**Chorus Melody Guitar/ Dirty Guitar 5 (ChMldy/ Dr5)**

A guitar melody was played during the choruses and panned 72% to the right. The guitar part was treated with the Act 2 signal chain and more EQ was done with a Filter Bank E6. On the EQ a high pass was set at 87.9Hz. There was also a 5.5dB boost at 1.17kHz and a 3.4dB boost at 5.28kHz.
For the second verse, there was two identical distorted rhythm guitar parts introduced. These tracks, named Dr6 and Dr7, were panned hard left and right, respectively, and assigned to a stereo auxiliary. On the auxiliary was the Gt1-2 hardware insert.

**Verse 2 Lead (Vs2Ld)**

Dr10 tracks were assigned to a mono auxiliary named Vs2Ld, which was panned hard left. This was a picking guitar part played in the second verse. The auxiliary had two Filter Bank EQs. The first was a Filter Bank P6, which had a 1.9dB cut at 42.4Hz, a 0.7dB cut at 231.7Hz, a 2dB boost at 1.74kHz, and a 1.9dB boost at 5.92khz.
It was later determined that a high pass at 87.9Hz was needed. Even though this essentially cancelled out the low frequency adjustment on the first EQ, the other adjustments were still needed. Consequently, the first EQ was left as it was. The second EQ also had a 3.3dB boost at 2.04kHz.

Figure 3.160 McDsp Filter Bank P6 EQ on Vs2Ld.

Figure 3.161 McDsp Filter Bank E6 EQ on Vs2Ld.
**Chorus Rhythm Guitars (ChRhy)**

The chorus rhythm and bridge rhythm was composed of Dr17 panned hard left, Dr18 panned hard right, Dr15 panned hard left, and Dr16 panned hard right. Dr15 and Dr16 only happened during the bridge. All of the guitar tracks were assigned to a stereo auxiliary and treated with the Api hardware chain. On both API EQs, 5kHz was boosted 2dB, 800Hz was boosted 2dB, and 300Hz was cut 2dB.

![Figure 3.162 API 550A on ChRhy.](image)

More EQ was done with a Filter Bank E6. The EQ had a high pass at 66.1dB, a 1.9dB cut at 260.8Hz, a 2.3dB boost at 2.39kHz, and a 2.3dB boost at 9.34kHz.
Feedback

Several takes of feedback were recorded to lead into the choruses. Dr19 was panned hard left, Dr20 was panned hard right, Dr21 was panned hard right, and Dr22 was panned hard left. All of the tracks were assigned to a stereo auxiliary named Feedback. No processing was done in Pro Tools.

Acoustic Guitars (Ags)

AgBod1 was panned hard left while AgBod2 and AgBod3 were panned hard right. The acoustic guitars utilized the GME 1-2 hardware insert. Then 3dB - 6dB of gain reduction was added with a Renaissance Axx compressor. The threshold was at -7.9dB and the attack was at 5.00. Output gain was left at unity.
A VEQ4 was used for further adjustments. A high pass was set at 82Hz. At 100Hz, there was a 2.5dB cut along with 3.4dB cut at 330Hz. A 3.9dB boost was made at 2.7kHz and a 3.5dB was made at 6.8kHz.

Acoustic guitar automation was rather simple. For the first verse, it was set at -7dB. The drums came up in dynamics for the pre-chorus so the acoustics were brought up 1dB. In the second verse the acoustic guitars were 0.5dB louder than the first verse to compete with the distorted electric guitars that were not present in the first verse.
**Piano (Pno)**

A piano part was recorded for the song introduction, which was effected with an EchoBoy Delay. A Memory Man style was chosen and the MIDI button was selected to time the delay with the song tempo. However, the primary function of the Echo Boy was not necessarily to create a noticeable delay. The dual echo setting was chosen but both echoes were set at a sixty-fourth note and the feedback was very low. The main objective was to give the piano an old muffled sound so that the following verse seemed to have a very open and present sound. This was accomplished by turning the saturation to roughly 1 o’clock so that the characteristics of the Memory Man style were more noticeable. Then, a low and high cut was set half way up to give the piano a weak and muffled sound.

![Sound Toys Echo Boy on Pno](image)

**Reverse Piano (PnoRev)**

An addition reverse piano track led into the notes of the regular piano. This track had no processing in Pro Tools.

**Lead Vocal (Vx)**

The lead vocal ran through the Sta hardware chain. After the hardware insert, Renaissance Vox was used to give the vocals another 3 to 5dB of gain reduction. The threshold was set at -6.4dB and the output gain was reduced to -2.5dB.
Next, a Massey De-Esser lowered harsh consonant syllables by 6dB around 6khz.

A Filter Freak1 was automated to turn on during the first verse and pre-chorus. It then
turned off for the rest of the song. A band pass filter was chosen and the input was to 3 o’clock to give the vocal a low fidelity distorted sound.

![Sound Toys Filter Freak 1 on Vx](image1)

**Figure 3.169 Sound Toys Filter Freak 1 on Vx.**

The vocal had a send feeding at -22.2dB to an auxiliary labeled VxDl with an Echo Boy delay inserted. A Memory Man style with a single echo and a dotted sixteenth note was chosen. The MIDI was button was selected to match delay times to the song’s tempo and feedback was set rather low. The low cut was set around 10 o’clock and the high cut was set just past half way.

![Sound Toys Echo on VxDl](image2)

**Figure 3.170 Sound Toys Echo on VxDl.**

Another was automated to turn on for the choruses and feed to an auxiliary named ChDl at -20dB. On the auxiliary an Echo Boy was set to a Memory Man style, a ping-pong delay, and a quarter note on both echoes. The MIDI button was selected to match the song’s tempo. The low cut was set just below 3 o’clock and the high cut was around 1 o’clock.
The delay was high passed at 254.6Hz, attenuated 9dB at 242.4Hz, boosted 3dB at 1kHz, and boosted 3dB at 5kHz.

A send was automated to turn on during the verses and feed to an auxiliary named Vrb at -16.1dB. The auxiliary had an AltiVerb reverb that had a 2.4s decay. The auxiliary was brought down to -9.8dB.
The lead vocal also had considerable attention to volume automation. On average, the vocals in the verses hovered around -3.6dB with a 1dB boost on syllables that were sung softly and a little more difficult to hear in the mix. The pre-chorus moved down to -5.6dB to match the low dynamics of the section. Then, the choruses moved up to -4.4dB. The choruses were sung significantly louder than the verses and did not need as much volume.

**Lead Vocal Double (Dbl)**

A lead vocal chorus double was panned 17% to the right and ran through the Vx1 hardware chain.

**Palm Trees In The Mountains**

**Kick (K)**

The kick sample and Audix D6 track was bussed to a mono auxiliary named K. A SSL EQ boosted the kick 5.6dB at 6.80kHz and 5.9dB at 4.01kHz with a Q of 0.10. There was also a 7.9dB cut at 250Hz with a Q of 0.10 and a 0.5dB boost at 104Hz.
The kick had a send to a Space Station set at -24.7dB, which returned on an auxiliary named Spc with a 25.85ms delay.

Figure 3.174 Waves SSL Channel on K.

Figure 3.175 Short Delay II on Spc.
There was also a send to key the Focusrite compression on the Dr and K/Sn masters, which was copied to Sn, SnComp, SamToms, OH, and Rms.

**Snare (Sn)**

The snare top and bottom track along with the snare sample were bussed to a mono auxiliary. On the auxiliary, a SPL Transient Designer was inserted to add attack to the snare and shorten the snare ring.

![Figure 3.176 SPL Transient Designer on Sn.](image-url)

Next in the chain was a Filter Bank E6 EQ with a 65Hz high pass, a 2.7dB boost at 256.2Hz, a 1.9dB cut at 460Hz, an 8dB boost at 1.94kHz, and a 4.9dB boost at 5kHz.
Sn fed to the Space Station at -20dB. The send was copied to SnComp, SamToms, OH, and Rms.

**Compressed Snare (SnComp)**

The parallel snare signal was processed with a SSL Channel. There was a compression ratio of 2.7:1, a threshold of -13dB, and a release of 0.36s. Those settings produced about 3dB of gain reduction.
Sample Toms (SamToms)

The original toms were out of tune and the heads were dead from overuse so only the sample toms were used. The sample toms were used in conjunction with the original toms throughout the album, but a large reason for recording sample toms was to provide a solution for problems like this.

The tom sound was treated with a Renaissance EQ. There was a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.
Drum Overheads (OH)

EQ for the overheads was done with a SLL Channel. A high pass was set at 84Hz, a 3.3dB boost was done at 14.04kHz, and a 4.1dB boost was made at 4.41kHz with a Q of 0.10. At 310Hz there was a 6.8dB cut with a Q of 0.10.
**Drum Rooms (Rms)**

Similar settings on the SSL Channel were used to alter the drum room tracks. There was a high pass set at 84Hz, a 3.4dB boost at 14.04kHz, a 2dB boost with a Q of 0.10 at 3.92kHz, and a 2dB cut with a Q of 0.10 at 310Hz.

Figure 3.180 Waves SSL Channel on OH.
Tambourine (Tambo)

A Filter Bank E6 EQ was inserted on the tambourine track giving it a 2dB boost at 5.26kHz and a high pass at 107.3Hz. The Tambourine was panned to the left 24%.

Figure 3.181 Waves SSL Channel on Rms.
Claps

Clap1 was panned right 67%, clap2 was left 67%, clap2 was left 70%, clap 4 was right 70%, clap5 was left 81%, clap6 was right 84%, clap7 was hard left, and clap8 was hard. All of the claps were assigned to a single stereo auxiliary named Claps. An SSL G Channel was inserted on the auxiliary to give the claps a high pass at 80Hz, a 5.7dB boost at 9.79kHz, and a 5.8dB boost with a Q of 0.10 at 236Hz.
The claps were set at -9.4dB for the choruses. When the song transitioned into pre-choruses the electric guitars stopped playing, which lowered the overall dynamic. The claps were reduced to -10.1dB to maintain an appropriate blend.

**Bass (Bs)**

The bass D.I. track was duplicated and renamed BsA. A Decapitator was inserted on the BsA track. Style “T” was selected and the drive was set a two and a half. The tone was around one and a half to brighten the bass and the high cut reduced some bothersome high frequency distortion.
The BsDI and BsA was then assigned to a mono auxiliary with the Bs hardware insert. The bass was then run through a L1 limiter with the threshold at -4.4dB and the release at 1s. There was a maximum of 6dB gain reduction. The bass was too loud due to auto make-up can so the out ceiling was turned down to -7.1dB.

More EQ was done with a Filter Bank P6 EQ. There was a 1dB cut at 104.8Hz, a 3.5dB cut at 239.9Hz, a 1.4dB boost at 1.9kHz, a 3dB boost at 1.38kHz, and a 3.5dB boost at 6.16khz.
A 32.1Hz high pass and a 18kHz low pass was applied with a Filter Bank F2 EQ.
**Clean Guitar 1 (Cl1)**

Cl1 was a harmonic guitar melody played during the verses, which was panned hard left. The guitar was treated through the Gt 1 hardware chain and then a bit more EQ was done with a Filter Bank E6. There was a 1.1dB boost at 1.49kHz and a 1.1dB boost at 8.79kHz.

![Figure 3.188 McDsp Filter Bank E6 EQ on Cl1.](image)

**Clean Guitar 2 (Cl2)**

Cl2 played during the pre-choruses. The auxiliary was panned hard left and treated with the TLA 1 hardware insert. A high pass was made at 83Hz with a Filter Bank E6. The EQ also had a 1.4dB cut at 503.6Hz, a 3.5dB boost at 2.29kHz, and a 3dB boost at 9.15kHz.
Clean Guitar 3 (Cl3)

A second clean guitar (Cl3) panned hard right was played during the pre-choruses. The part was first processed with the TLA 2 hardware insert and a Filter Bank E6 EQ was inserted next. The EQ had an 80.7Hz high pass, a 1.5dB cut at 512.5Hz, a 3dB boost at 1.88kHz, and a 3dB boost at 9.15kHz.

Figure 3.189 McDsp Filter Bank E6 EQ on Cl2.
Distorted Guitar 3 (Dr3)

Two identical rhythm guitars were put on a stereo track and processed together. The signal ran too hot in the API hardware chain so a trim plug-in reduced the output by 3dB.

The guitars had a 4dB boost at 5kHz, a 2dB boost at 800Hz, and a 2dB cut at 300Hz.
After the guitars ran through the API hardware chain, additional EQ was done with a Filter Bank P6. The EQ had a 1.4dB cut at 357.8Hz, a 2.5dB boost at 2.39kHz, and a 1dB boost at 9.15kHz.
**Chorus Melody (ChMldy)**

A guitar melody (Cl4) that played during the choruses was assigned to an auxiliary named ChMldy and panned left 69%. That guitar part was treated with the Act2 hardware chain and then a Filter Bank E6 EQ. On the EQ, there was a high pass at 87.9Hz and a 3.3dB boost at 2.04khz.

![Figure 3.194 McDsp Filter Bank E6 EQ on ChMldy.](image)

**End Lead Guitar (EndLd)**

Cl8 and Cl9 played identical lead parts at the end of the song. Cl8 was panned hard left and Cl9 was panned hard right. The two guitars were then assigned to an auxiliary named EndLd and treated with a Filter Bank E6 EQ. There was a high pass at 57.4Hz, a 1.4dB cut at 224.2Hz, a 5.9dB boost at 2.12khz, a 6dB boost at 8.45kHz, and a 1.5dB boost at 9.34kHz.
Acoustic Guitars (Ags)

Two identical acoustic guitar tracks were panned hard left and right and assigned to an auxiliary labeled Ags. Processing started with the GME 1-2 hardware insert and was continue with a Renaissance Axx compressor. The threshold was at -3.2dB and the attack was at 5.00. There was a 3dB gain reduction at most.

Figure 3.195 McDsp Filter Bank E6 EQ on EndLd.
A VEQ4 high passed the acoustic guitars at 82Hz. There was a 2.55dB cut at 100Hz, a 3.4dB cut at 330Hz, a 3.9dB boost at 2.7kHz, and a 3.5dB boost at 6.8kHz.

Throughout most of the song the acoustic guitars were set at -1dB. However, only one acoustic guitar was used for the verses. To compensate for the reduction of volume due to only one acoustic guitar playing, the Ags auxiliary was turned up 1.7dB.

**Lead Vocal (Vx)**

The lead vocal was processed with the Sta hardware insert. Next a Renaissance Vox compressor gave the vocal another 10dB of gain reduction by setting the threshold at -12.2dB.
The track was peaking so the output gain was set at -1.9dB.

![Image of audio processing interface]

Figure 3.198 Waves Renaissance Vox on Vx.

Harsh consonants were reduced 6db around 6kHz.

![Image of audio processing interface]

Figure 3.199 Massey De-Esser on Vx.
The lead vocal had a send feeding to an auxiliary labeled VxDl at -22.2dB. An Echo Boy delay with a Memory Man style and a single echo was on the auxiliary. The echo time was a dotted sixteenth note and the MIDI button was selected to synchronize the delay with the tempo of the song. Low cut was set around 10 o’clock and high cut was at 1 o’clock. The delay auxiliary was set at -7.4dB for the whole song except the introduction. Only acoustic guitar and the lead vocal were happening in the introduction so the delay auxiliary was brought down 10dB to give the vocal a fairly unprocessed and natural sound.

The lead vocal sent to another auxiliary with an Echo Boy set on a Memory Man style, a ping-pong echo, and a quarter note for each echo. The saturation was turned all the way up to enhance the memory man characteristics. The feedback was set fairly long around 10 o’clock and the low cut was rather severe at 3 o’clock. The high cut also had quite an effect at 1 o’clock.
The ChDl send was automated to turn on only during the choruses. There was also some volume automation to the vocal track. The song followed a fairly typical dynamic development. The verses were mellow, which were followed by a pre-chorus that had a slight increase in dynamics and choruses that reached the peak of the song’s dynamics. The lead vocal followed this dynamic map by starting at -5dB in the verses, moving up 0.6dB for the pre-choruses, and peaking at -3.7dB for the choruses. The only difference was the calm bridge set at -5.2dB.

**Lead Vocal 2 (Vx2)**

There was a second vocal that was at the introduction and chorus outs. The Vx 1 hardware insert was used for the vocal. There was also additional gain reduction of 6dB to 10dB applied with a Renaissance Vox compressor. The output gain was brought down to -1.9dB since the automation make-up gain was causing the track to peak.
The second lead vocal also sent to VxD1 -20.5dB.

**Perfect Remedy**

*Kick (K)*

The kick was originally recorded with a Shure Sm91 and a Shure Beta 52. The Beta 52 was broken and somewhat distorted so it was made inactive. The Sm91 and a sample kick were assigned to a mono auxiliary named K. A SSL Channel was inserted on the auxiliary to EQ the kick. The kick was boosted at 6.8kHz by 4.7dB and at 4.0kHz by 5dB with a Q of 0.10. An 8.4dB cut with a Q of 0.10 was made at 310Hz, and a 2.6dB boost was made at 104Hz.
The kick sent to a Space Station at -21.3dB. The Space Station signal returned on an auxiliary named space with a 26.21ms delay.
Another send keyed the Focusrite compression on the Dr and K/Sn masters at unity. The drum compression key send was copied to Sn, SnComp, Toms, SamToms, OH, and Rms.

**Snare (Sn)**

The snare top, snare bottom, and snare sample tracks were assigned to a mono auxiliary labeled Sn. The snare attack was increased with an SPL Transient Designer. The snare ring was also reduced by lowering the sustain.

![Figure 3.205 SPL Transient Designer on Sn.](image)

A Filter Bank E6 EQ on the Sn auxiliary had a 65Hz high pass, a 1.2dB boost at 190.9Hz, a 3.8dB cut at 460Hz, a 7.8dB boost at 1.59kHz, and a 4dB boost at 5kHz.
Sn fed to the Space Station at \(-15.8\)dB. The send was copied to SnComp.

**Compressed Snare (SnComp)**

The parallel snare track used a SSL Channel to make a high pass at 109Hz, a 4.4dB boost at 8.57kHz, a 5.3dB boost at 4.35kHz with a Q of 0.10, and a 3dB boost at 264Hz. There was also about 3dB of gain reduction. The compressor had a 2.7:1 ratio, a threshold at \(-13\)dB, and a release of 0.36s.

![Figure 3.206 McDsp Filter Bank E6 on Sn.](image)
EQ adjustments to the Toms were made with a Renaissance EQ. There was a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.
A send fed the Space Station at -25dB. The send was copied to SamToms.

**Sample Toms (SamToms)**

The sample toms had the same processing as the original toms.

**Drum Overheads (OH)**

An SSL Channel was inserted on the drum overheads. There was a high pass at 84Hz, a 3.9dB boost at 14.04kHz, a 4.5dB boost at 3.92kHz with a Q of 0.10, and a 2dB cut at 310Hz with a Q of 0.10. A bit of compression was needed on the overheads, so the ratio was set at 2.2:1, the threshold was at -9.7dB, and the release was at 0.23s. This gave the overheads about 3dB of gain reduction.
The balance of the left and right overhead tracks was uneven so a trim plug-in was used to lower the left overhead by 5.5dB.

![Figure 3.209 Waves SSL Channel on OH.](image)

**Drum Rooms (Rms)**

A SSL Channel was used to EQ the drum room tracks. The rooms were high passed at 84Hz and a 4.9dB boost was done at 10.21kHz. There was also a 4.1dB boost at 3.34kHz with a

![Figure 3.210 Trim on OH.](image)
Q of 0.10 and a 4.2dB cut at 310Hz with a Q of 0.10.

![Image of SSL Channel on Rms.](image)

**Tambourine (Tambo)**

The tambourine track had no processing in Pro Tools.

**BigTom**

Some tom tracks were programmed for a dynamic build in the bridge. BigTom1 and BigTom3 were panned hard left while BigTom2 and BigTom4 were panned hard right. All of the programmed toms were assigned to a stereo auxiliary named BigTom and treated with a Renaissance EQ. They received a 6.5dB cut at 45Hz, a 1dB boost at 151Hz, an 11.4dB cut at 300Hz, a 5.1dB boost at 4268Hz, and a 4dB boost at 7885Hz.
The toms had too much sustain and masked some of the other instruments in the section. A SPL Transient Designer was used to reduce the sustain of the toms, which in turn retained definition in the song section.

The programmed toms had a send feeding the Space Station at -22.2dB.
The bass was recorded with a D.I. and a Sennheiser MD 421. The D.I track was named BsDI and the MD 421 was named BsA. On the BsA track, there was a Decapitator plug-in set to style “T”. The punish button gave the track 20dB of distortion and the drive was set at 2.5 for a bit more distortion. Setting the tone control to 2 o’clock brightened the track and the high cut reduced unwanted high-end fuzz created by the plug-in.

The BsDI and BsA tracks were assigned to a mono auxiliary named Bs with the Bs hardware insert. After the hardware insert was a L1 limiter that had a threshold of -4.4dB and a release of 1s. The automatic make-up gain function of the limiter gave the bass more volume than was desired, so the out ceiling was lowered to -7.1dB. The limiter produced about 3dB - 6dB of gain reduction.
The bass also had a Filter Bank E6 EQ that made a 2.7dB cut at 230.6Hz, a 1.4dB boost at 1.90kHz, a 3dB boost at 1.38kHz, and a 3.5dB boost at 6.16kHz. Some notes played on the bass during the chorus were higher and did not produce an adequate low-end so Maxx Bass was automated to turn on during those sections. Sub-harmonics were created with an 80Hz center frequency and a high pass was engaged to prevent the creation
of excessive sub-harmonics. The Maxx Bass signal was turned down to -10.7dB so that the bass did not overpower the mix.

Some string slides were accentuated by compression and EQ. Very quick ducks ranging from 3dB to 15dB were automated to provide volume consistency. The bass guitar was also turned up from -5.8dB to -5.3dB for the choruses.

**Clean Guitar 1 (Cl1)**

A clean guitar, Cl1, melody that played during the verses was assigned to a mono auxiliary and panned hard left. The TLA 1 hardware insert was used on Cl1 and a 3dB boost at 9.15kHz was made with a Filter Bank E6 EQ.
Clean Guitar 2 (Cl2)

A second clean verse guitar, Cl2, was assigned to a mono auxiliary and panned hard right. The TLA 2 hardware insert was put on the auxiliary and another 3dB at 9.15kHz was boosted with a Filter Bank E6 EQ.

Figure 3.218 McDsp Filter Bank E6 EQ on Cl1.
Clean Guitar 4 (Cl4)

A clean guitar melody, Cl4, that played during the bridge was panned hard right. The Gt1 hardware insert was used for this guitar track. Further treatment was done with a Filter Bank E6 EQ. The EQ had an 87.9Hz high pass, a 1.9dB boost at 1.61kHz, and a 5dB boost at 5.06kHz. The track had quite a bit of high frequency amplifier noise so there was a 12dB cut at 16kHz and a high pass at 5.08kHz.
Clean Guitar 5 (Cl5)

Another clean guitar melody, Cl5, that played during the bridge was panned hard left. A Filter Bank P6 EQ was used to brighten the guitar sound. A 1.9dB cut was made at 42.4Hz as well as a 0.7dB cut at 231.7Hz. A 2dB boost was made at 1.74kHz and a 1.9dB boost was made at 5.92kHz.

Figure 3.220 McDsp Filter Bank E6 EQ on Cl4.
A Renaissance Axx compressed the guitar track 3dB - 6dB. The threshold was at -11.4dB and the attack was 3.5. The compression resulted in excessive signal so the output gain was brought down to -4.7dB.

Figure 3.221 McDsp Filter Bank P6 on CI5.
Clean Guitar 6 (Cl6)

A third clean guitar, Cl6, played at the beginning of the bridge and was panned hard right. The guitar was processed with the Gt2 hardware. From there, a Renaissance Axx gave the guitar about 3dB more of compression. The threshold was at -2.8dB and the attack was at 3.50. The output gain was turned down to -5.6dB.
Two identical clean guitar parts, Dr1 and Dr2, were panned hard left and right, respectively, and assigned to a stereo auxiliary. These guitars were processed with the Act 1-2 hardware chain. More EQ was done with a Filter Bank E6 EQ. There was a high pass at 93.1Hz, a 1.1dB cut at 137.2Hz, a 3dB boost at 1.49khz, and a 3.1dB boost at 4.49kHz. There was troublesome amplifier noise so a low pass was set at 9.95kHz.

**Pre-Chorus Guitars (Pre)**

Two identical clean guitar parts, Dr1 and Dr2, were panned hard left and right, respectively, and assigned to a stereo auxiliary. These guitars were processed with the Act 1-2 hardware chain. More EQ was done with a Filter Bank E6 EQ. There was a high pass at 93.1Hz, a 1.1dB cut at 137.2Hz, a 3dB boost at 1.49khz, and a 3.1dB boost at 4.49kHz. There was troublesome amplifier noise so a low pass was set at 9.95kHz.
More compression was needed so a Renaissance Axx compressor was set with a -5dB threshold and a 5.00 attack. This gave the guitars another 3dB - 5dB of gain reduction. The output gain was lowered to -1.1dB.
Distorted Guitar 3 (Dr3)

A distorted chorus guitar melody, Dr3, was panned hard left. The API hardware insert was used to process this guitar part. This guitar ran through the EQ on the left, which had a 2dB boost at 5kHz, a 2dB at 800Hz, and a 2dB cut at 300Hz.
Some relatively small adjustments were made after the hardware insert with a Filter Bank E6 EQ. A high pass was set at 60.7Hz, a 2.6dB boost was done at 1.43khz, and a 2dB boost was made at 4.86khz. Some high frequency amplifier noise was reduced with a 16.6kHz low pass.

![Figure 3.227 McDsp Filter Bank E6 EQ on Dr3.](image)

When Dr3 entered right before the chorus it was too loud and needed a 0.7dB drop from -5.6dB to -6.4dB.

**Distorted Guitar 4(Dr4)**

The second distorted chorus guitar part, Dr4, was panned hard right. The second API hardware chain was inserted. The second API EQ was identical to the first API EQ.

A Filter Bank E6 EQ came after the API hardware chain. On this EQ there was a 47Hz high pass, a 1.8dB cut at 140.7Hz, a 3.7dB boost at 1.13khz, and a 5.3dB boost at 5.47khz.
A 0.7dB volume reduction was also automated for Dr4 immediately before the chorus.

**Acoustic Guitars (Ags)**

An acoustic guitar track was panned hard left (Ag1) and its double (Ag2) was panned hard right. Most of the processing was done with the GME 1-2 hardware insert. Another 3dB of gain reduction was applied with a Renaissance Axx compressor. The threshold was set to -3.2dB and the attack was at 5.00.
More detailed EQ was done with a VEQ4. A high pass was made at 82Hz, a 1dB cut was done at 100Hz, and a 2.9dB cut was made at 330Hz. There was a 3.2dB boost at 2.7kHz and a 1.9dB boost at 6.8kHz.

**Piano (Pno1)**

A SSL Channel was used to compress the piano and make EQ adjustments. At 8.8kHz there was a 7.6dB boost along with a 7.9dB boost at 1.62kHz with a Q of 0.10. There was a 3.4dB cut at 2.9kHz with a Q of 0.10 and a 2dB boost at 127Hz.
**Lead Vocal 1 (Vx1)**

The lead vocal ran through the Sta hardware chain, which was followed by another 6dB - 10dB of compression with a Renaissance Vox compressor. The threshold was at -11.1dB and the gain was at -1.9dB.
A Massey De-Esser lowered sibilance about 6dB around 6khz.

The vocal had a send feeding to an auxiliary labeled VxDl -20dB. On the auxiliary was an Echo Boy delay with a Memory Man style, a single echo, and a dotted sixteenth. The MIDI
was selected to match the tempo of the song and the feedback was rather low. The low cut was around 10 o’clock and the high cut was just under 1 o’clock.

![Image 1](source_image1.png)

**Figure 3.234 Sound Toys Echo Boy on VxDI.**

In general, the VxDI auxiliary was set at -5.5dB. When the song was at it’s lowest dynamic points in the introduction and bridge, the VxDI auxiliary was lowered to -11.5dB.

A second vocal send fed to an auxiliary named ChDl at -18.2dB during the choruses. An Echo Boy delay was inserted on the auxiliary. The delay had a Memory Man style with the saturation at its maximum. A ping-pong delay was chosen along with a quarter note delay. The low cut was set at 3 o’clock and the high cut was around 1 o’clock.

![Image 2](source_image2.png)

**Figure 3.235 Sound Toys Echo Boy on ChDl.**

For the introduction, the vocal entered at its lowest volume of -6.1dB. When more
instrumentation entered in the verse, the vocal moved up to -4.8dB. The drums became louder in the pre-choruses and called for another vocal increase of 0.5dB, which preceded a 1dB increase in the choruses. In the transition from the second chorus to the bridge, the vocal volume moved from -3.3dB to -4.5dB and moved up to -4.2dB as the bridge increased in dynamics. The final chorus returned to -3.3dB and the outro/ partial verse returned to -4.8dB.

**Lead Vocal Double (Dbl)**

A chorus double was run through the Vx1 hardware chain. After the hardware chain was a Renaissance Vox compressor with a -11.1dB threshold. This produced 6 to 10dB of gain reduction. The gain output was at -1.9dB.

![Figure 3.236 Waves Renaissance Vox on Dbl.](image)

A Massey De-Esser with identical settings to the lead vocal de-esser was used on the chorus double.

The lead vocal double fed to VxDI at -20dB and ChDI -25dB. Both sends were copied to Bgs.

**Background Vocals (Bgs)**

A background vocal and its double were put on a stereo track and compressed with a Renaissance Vox compressor. The threshold was set to -11.1dB for a result of 6dB - 10dB gain reduction. The output gain was set at -1.9dB.
A Filter Bank E6 EQ was used to carve the background vocals into the mix. There was a 78.4Hz high pass, a 1.8dB cut at 305.5Hz, a 4.1dB boost at 3.69kHz, and a 5.9dB boost at 9.34kHz.

Figure 3.237 Waves Renaissance Vox on Bgs.
The second and third phrases of the background vocals were louder than the first phrase and need to be reduced from -11.1dB to -11.6dB.

*Take Your Pride and Dance*

**Kick (K)**

The Audix D6 kick track and sample kick were assigned to a mono auxiliary named K with a SSL Channel inserted. The kick was boosted 5.5dB at 6.80khz and 4.5dB at 4.01khz with a Q of 0.10. There was a 9.9dB cut at 250Hz with a Q of 0.10 and a 1.7dB boost at 104Hz. There was also about 3dB of compression. The threshold was at -7.4dB and the release was 0.40s. The ratio was 2:1.
The kick had two sends. One send fed to a Space Station at -20dB and the other functioned as the key for the Focusrite compression on the Dr and K/Sn masters. The Space Station returned on an auxiliary named Spc with a 22.87ms delay.
The key send was set at unity and was copied to Sn, SnComp, Toms, SamToms, OH, and Rms.

The kick, as well as the rest of the drum-set tracks had a 1dB boost on the fill that transitioned from the bridge to the final chorus.

**Compressed Snare (SnComp)**

The snare top, snare bottom, and snare sample were all assigned to the mono SnComp auxiliary. The snare was compressed with a SSL Channel. The compressor had a 2.7:1 ratio, a release of 0.36s, and a threshold of -13dB. The channel was also used to high pass the snare at 109Hz, boost 8.57kHz by 4.5dB, and 1.92kHz by 5.1dB with a Q of 0.10. There was a cut of 3.1dB at 550Hz with a Q of 0.10 and a 2dB boost at 280Hz.

![Figure 3.241 Waves SSL Channel on SnComp.](image)

The Sn auxiliary had a send set at -15dB sending to the Space Station. The send was copied to SnComp, Toms, and SamToms.

**Snare (Sn)**

The same snare tracks were assigned to the Sn auxiliary, which was not compressed. A
SPL Transient Designer was used to boost the snare attack and shorten its ring.

The snare received further EQ with a FilterBank E6. A high pass was set at 65Hz, a cut of 4.4dB was made at 335.3Hz, an 8dB boost was done at 2.01kHz, and a 2.7dB boost was made at 5kHz.

![Figure 3.242 SPL Transient Designer on Sn.](image)

![Figure 3.243 McDsp Filter Bank E6 EQ on Sn.](image)
**Toms**

A Renaissance EQ was inserted on the tom track. The toms had a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.

*Sample Toms (SamToms)*

The sample toms had the same EQ and sends as the original toms.

*Drum overheads (OH)*

The drum overheads had some SSL Channel EQ. The high pass was at 84Hz. There was a 4.1dB boost at 14.04kHz, a 4.3dB boost at 3.92kHz with a Q of 0.10, and a 6.1dB cut at 310Hz with a Q of 0.10.
The overheads fed to the Space Station at -21.3dB. The send was copied to Rms.

**Drum Rooms (Rms)**

For this song a slightly different drum sound was achieved by not using the room microphones for most of the song. However a snare room sample was still used so it was assigned to the Rms auxiliary with the drum room track. Some EQ was done with a SSL Channel. The high pass was at 84Hz, a 3.4dB boost was made at 14.04kHz, a 4.3dB boost with a Q of 0.10 was made at 3.92kHz, and there was a 6.1dB cut with a Q of 0.10 at 310Hz.
The room tracks were only used during the bridge. Furthermore, all of the other drum tracks were muted and a SansAmp plug-in was used to distort the drum room track. When the drums moved from this low fidelity distorted sound to the full drum-set in the chorus, the dynamics drastically jumped up.

Figure 3.246 Waves SSL Channel on Rms.

Figure 3.247 SansAmp on Rms.
**One Beat (1Bt)**

During the bridge there was a beat created with several percussive sounds. An auxiliary labeled 1Bt had percussive tracks that primarily fell on the one beat assigned to it. Among those was Bt1A was panned left 22%, Bt1B panned to the right 27%, BtD112 panned to the right 30%, Chair1 panned to the left 54%, and Chair2 panned to the right 58%.

1Bt had a send feeding the Space Station at -15.1dB. The same applied to 3Bt.

**Three Beat (3Bt)**

Another set of percussive sounds that mostly fell on the three beat were assigned to an auxiliary named 3Bt. Those tracks were Bt1.87 panned to the right 45%, Bt2.87 panned to the left 41%, Wood1 panned to the right 51%, and Wood2 panned to the left 49%. A Filter Bank E6 EQ was used to make a high pass at 41.9Hz, and a 3.4dB boost at 1.74kHz.

![Figure 3.248 McDsp Filter Bank E6 EQ on 3Bt.](image-url)
**Bass (Bs)**

BsDI (bass D.I.) and BsA (bass amp) were assigned to an auxiliary named Bs. There was a Decapitator plug-in on the BsA track. The drive control was set around 2.5, the tone was turned toward the bright side just past 1 o’clock and the high cut was just past 2 o’clock to reduce high frequency distortion.

![Image of Decapitator plug-in](image.png)

*Figure 3.249 Sound Toys Decapitator on BsA.*

First, the bass was run through the Bs hardware chain. Then it was compressed further with a L1 limiter, which had a threshold of -5.3dB, a release of 1s, and an out ceiling of -7.1dB. The limiter gave the bass approximately 6dB more of gain reduction.
Next in the chain was a Filter Bank E6 EQ with a 1dB cut at 100Hz, a 3.4dB cut at 259.6Hz, a 1.4dB boost at 1.90kHz, a 3.3dB boost at 1.38kHz, and a 4.1dB boost at 6.16kHz. The bass also had a Filter Bank F2 EQ with a 30.9Hz high pass and an 18kHz low pass.

The bass remained at -7.3dB for the entire song besides the bridge. The dynamics developing from the bridge to the final chorus needed to be significant. One of the things that
helped to accomplish this was bringing the bass down to -11dB.

*Clean Guitar 1 (Cl1)*

Cl1 played in the verses and pre-choruses. It was panned hard left and processed through the TLA 1 hardware chain. About 3dB - 6dB of gain reduction was added with a Renaissance Axx compressor. It had a threshold of -7.4dB, an attack of 5.00, and a -4.6dB output gain.

Figure 3.252 Waves Renaissance Axx on Cl1.

One more boost of 3dB was made at 9.15khz with a Filter Bank E6 EQ.
Volume automation for Cl1 was written by riding the fader in real-time. In the verses, the level was around -3.6dB. As the song moved into the pre-chorus the level was pushed up about 1dB and right before the chorus it peaked out around -1.7dB.

*Distorted Guitar 1 (Dr1)*

Dr1 was a slightly distorted volume swell guitar that played during the verses and pre-choruses. The part was processed with the TLA 2 hardware insert as well as a Filter bank E6 EQ, which had a 3dB boost at 9.15kHz.
A Pan Man plug-in automatically panned the guitar back and forth. The width was set half way up so it never panned completely to the left or right and the smoothing control was set all the way to soft so that the panning transitions were smooth.

The performance of Dr1 was naturally dynamic due to the swell effect. Volume automation developed this even further. The fader movements changed with the dynamic of the swells and faded up and down from -19.5dB to -9.9dB.

**Distorted Guitar 2 (Dr2)**

Dr2 was a rhythm guitar part for the first two verses, pre-choruses, and choruses panned
hard right. The Act 1 hardware insert was used, which was followed by a Renaissance Axx that gave the guitar about 3dB of gain reduction. The threshold was set at -5dB, the gain was at -5.2dB, and the attack was at 3.45.

![Figure 3.256 Waves Renaissance Axx on Dr2.](image)

The guitar was also processed with a Filter Bank E6 EQ. The high pass was set at 51.2Hz with a 2.7dB boost at 1.49khz and a 3.3dB boost at 5.26khz.
Dr2 entered around -4.8dB and came up about 0.5dB for the pre-chorus. The performance was louder in the chorus and needed to be brought down to -5.4dB. The same approximate automation was repeated on the second verse, pre-chorus, and chorus.

**Distorted Guitar 3 (Dr3)**

Dr3 was the main melodic guitar line during the first two choruses as well as a melody in the bridge. The guitar was panned hard left and ran through the Act 2 hardware insert.

For the choruses, Dr3 was at -6.8dB. When the song came down to the bridge, the track was brought down to -17dB and slowly faded up to -13.1dB at the end of the bridge. The slow increase in volume built suspense into the song and foreshadowed the climax of the song to come.

**Reverse Guitar (RevGt)**

Dr4 was panned hard left and Dr5 was panned hard right. The guitar tracks were two melodies in reverse at the beginning of the verse. The guitars were assigned to a stereo auxiliary named RevGt with a Filter Bank E6 EQ inserted. The EQ had a 1.8dB boost at 1.81kHz.
Distorted Guitar 6 (Dr6)

Dr6 was a small guitar part leading out of the first choruses and a building melody during the bridge. The guitar part was panned hard right and treated with a Filter Bank E6 EQ. It had a high pass at 52.7Hz, a 2.9dB cut at 331.9Hz, a 3.8dB boost at 1.96kHz, a 4.4dB boost at 5.47kHz, and a 3.1dB boost at 9.73kHz.

Figure 3.258 McDsp Filter Bank E6 EQ on RevGt.
Rhythm Guitar 1 (Rhy1)

The main rhythm guitar at the end of the song was Dr7 panned hard left and Dr8 panned hard right. The guitars were assigned to a stereo auxiliary labeled Rhy1 and processed with the Gt 1-2 hardware chain. A Filter Bank E6 EQ was also used to make a 48.3Hz high pass, a 1.4dB cut at 317.8Hz, a 1.5dB boost at 800Hz, and a 0.5db boost at 5kHz.

![McDsp Filter Bank E6 EQ on Dr6.](image-url)
Distorted Guitar 9 (Dr9)

A slide guitar, Dr9, was introduced at the end of the bridge. The part was panned left 38% and treated with a Filter Bank E6 EQ that had a 2.5dB at 1.22kHz, and a 2.3dB boost at 4.49kHz.
Distorted Guitar 10 (Dr10)

Another melodic guitar, Dr10, was played during the last chorus. The guitar melody was panned left 64% and processed with the API 1 hardware insert. The API EQ was set to a 4dB boost at 5kHz and a 2dB boost at 800Hz.

A 60.7Hz high pass, a 1.6dB boost at 1.81kHz, and a 3.4dB boost at 5.92kHz were made with a Filter Bank E6 EQ.
Dr11 was a re-introduction of the main chorus guitar melody, Dr9, played in the two choruses. However, Dr11 had more amplifier distortion. The melody first ran through the API 2 hardware chain. The second API EQ had the same settings as the first API EQ. Then, a Filter Bank E6 EQ was used to make a high pass at 74.1Hz and a 3.5dB boost at 6.16kHz.

**Distorted Guitar 11 (Dr11)**

Dr11 was a re-introduction of the main chorus guitar melody, Dr9, played in the two choruses. However, Dr11 had more amplifier distortion. The melody first ran through the API 2 hardware chain. The second API EQ had the same settings as the first API EQ. Then, a Filter Bank E6 EQ was used to make a high pass at 74.1Hz and a 3.5dB boost at 6.16kHz.
Acoustic Guitars (Ags)

AgBod1 and AgBod 3 were panned hard left and AgBod2 was panned hard right. All three guitars were assigned to a stereo auxiliary named Ags and processed with the GME 1-2 hardware insert. Renaissance Axx gave the guitars another 3dB - 6dB of gain reduction. The compressor had a threshold of -7.9dB and the attack was at 5.00.

Figure 3.264 McDsp Filter Bank E6 EQ on Dr11.
More EQ treatment was done with VEQ4. The high pass was at 82Hz, a 3dB cut was done at 100Hz, and a 2.3dB cut was made at 330Hz. At 2.7kHz there was a 3.6dB boost and at 6.8kHz there was a 2.3dB.

**Lead Vocal 1 (Vx1)**

The lead vocal was run through the Sta hardware insert chain and further compressed with a Renaissance Vox compressor. The compressor had a threshold of -12.1dB, which resulted in approximately 6dB - 10dB of gain reduction. The output gain was turned down to -2.3dB.
A Massey De-Esser turned down sibilance about 6dB around 6kHz.

The lead vocal had a send feeding to a stereo auxiliary named VxDI at -22.2dB. An Echo
Boy delay was inserted on the auxiliary and set to a Memory Man style with a single echo and a
dotted sixteenth note. The feedback was very short. The low cut was turned to approximately
10 o’clock and the high cut was just past half way.

![Figure 3.269 Sound Toys Echo Boy on VxDl.](image)

The VxDl auxiliary was set low, -9.6dB, at the beginning of the song so it was very
subtle. When drums entered, the delay was turned up to -5.7dB so it could still be heard. The
delay came up again in the pre-chorus to -2.5dB to help the transition into the chorus, which had
a noticeably different space with the delay at unity. The same approach was used for the second
verse, pre-chorus, and chorus. When the song entered the bridge, the delay was brought down to
-4.3dB to give it a more up-front and intimate sound. It then returned to unity for the ending.

A second send fed to a stereo auxiliary named ChDl at -20dB with an Echo Boy delay
inserted. This delay was also set to Memory Man, but the saturation was turned to its maximum
and a ping-pong delay with a sixteenth note was chosen. The low cut was just below 3 o’clock
and the high cut was around 10 o’clock.
The ChDl was automated to turn on only during the chorus so the level remained the same for the first two choruses. It moved from -17.5dB to -16dB for the climax of the song to give the impression that the vocal was in a slightly bigger space.

The vocal volume changes were rather small. The verses were at -5.2dB, which only moved up 0.6dB for the pre-choruses and another 0.3dB in the choruses. Typically volume changes that small are not discernible, but what is noticeable is the changing relation to the other instruments. The bridge came down to -7.2dB and built up 1dB over the length of the bridge. Finally, the vocal hit its loudest at -3.5dB for the grand finale ending.

*Lead Vocal 2 (Vx2)*

There was several sections throughout the song were the lead vocal would overlap so it was recorded on two tracks. The second lead vocal was processed with the Vx 1 hardware insert. A Renaissance Vox compressor gave the vocal another 6dB - 10dB of gain reduction.
A Massey De-Esser with the same settings as the Vx1 De-Esser was inserted. Last in the chain was a Filter Bank E6 EQ. Vx2 was a little darker than Vx1 so a 3.4dB boost was made at 2.12kHz along with a 2.2dB boost at 10.1kHz.
Since Vx was meant too sound as close to Vx1 as possible, the sends from Vx1 were copied to Vx2. Some volume automation was needed to make the lead vocals transition smoothly. The quiet introduction of the song was at -5.4dB. The lead vocal re-entered 1dB louder at the beginning of the denser second verse and did not appear again until the loud ending of the sound at -3.5dB.

**Lead Vocal (Dbl)**

A lead vocal double for the second chorus was panned to the right 19%. The double also had a send feeding to VxDl and ChDl at -20dB.

**The Fall**

**Kick 1 (K1)**

The Audix D6 track and sample kick were assigned to a mono auxiliary labeled K1. EQ treatment to the kick was done with a SSL Channel. There was a 4dB boost at 6.80kHz, a 2.7dB boost at 4.01kHz with a Q of 0.10, an 8.6dB cut at 420Hz with a Q of 0.10, and a 1.9dB boost at 104Hz.

Figure 3.272 McDsp Filter Bank E6 EQ on Vx2.
The kick sent to the Space Station at -20dB and returned to an auxiliary named Spc with a 26.23ms delay.

There was also a send set at unity that keyed the Focusrite compressor on the Dr and

Figure 3.274 Short Delay II on Spc.

There was also a send set at unity that keyed the Focusrite compressor on the Dr and
K/Sn masters. The send that keyed the drum compression was copied to Sn, SnComp, Toms, SamToms, OH, Rms, K2, Sn2, SnComp2, Toms2, SamToms2, OH2, and Rms2.

The kick also had one volume change. Up until the bridge the kick was at -0.5dB. For the bridge it was brought down to -2.2dB to create a larger volume gap moving into the big ending.

*Snare (Sn1)*

The snare top and snare bottom were assigned to a mono auxiliary named Sn. A SPL Transient Designer boosted the snare attack and shorted the snare ring.

![Figure 3.275 SPL Transient Designer on Sn1.](image)

This snare track was played with brushes so the EQ was rather different than snare EQs used on the album. In particular, a severe 12dB cut was made at 377.5Hz to reduce excessive mudiness. There was also a 65Hz high pass, a 2.7dB boost at 256.2Hz, an 8dB boost at 1.86kHz, and a 3.4dB boost at 5kHz.
Sn fed the Space Station at -15dB and that send was copied to SnComp.

**Compressed Snare (SnComp1)**

A parallel snare signal was compressed with a SSL Channel. The compressor had a -13dB threshold, a 2.7:1 ratio, and a release of 0.36s. This gave the snare about 3dB of gain reduction. The EQ was also used to make a 5dB boost at 8.95kHz, a 5.9dB boost at 4.4kHz with a Q of 0.10, a 15dB cut at 380Hz with a Q of 3.5, and a 2.6dB boost at 280Hz.

Figure 3.276 McDsp Filter Bank E6 EQ on Sn1.
The toms were treated with a Renaissance EQ. There was a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.

Figure 3.277 Waves SSL Channel on SnComp1.

**Toms (Toms1)**

The toms were treated with a Renaissance EQ. There was a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.
The toms fed to the Space Station at -20dB.

**Drum Overheads (OH1)**

EQ adjustments were made with a SSL Channel. There was a high pass at 84Hz, a 3.7dB boost at 14.04kHz, a 3.5dB boost at 3.92kHz with a Q of 0.10, and a 2dB cut at 310Hz.
The overheads were set at -3.4dB. For the first two choruses, the high-hats were too loud so the overheads were brought down to -4.2dB. The overheads were then brought down even further to -5.9dB to build on the dynamic development into the climax of the song.

**Drum Rooms (Rms1)**

The SSL Channel was used on the drum room tracks as well. The high pass was set at 84Hz. A 3.7dB boost was made at 14.04kHz along with a 6.7dB boost at 4.52kHz with a Q of 0.10. A 9.2dB cut with a Q of 0.10 was made at 280Hz and 101Hz was cut 6.6dB.
There was a second drum-set played with regular stick for the climax of the song. The Audix D6 and sample kick were assigned to the K2 mono auxiliary. A SSL Channel was inserted on the auxiliary for EQ and compression. The compressor applied about 3dB of reduction with a 2:1 ratio, a -7.4dB threshold, and a release of 0.40. The EQ had a 5.5dB boost at 6.80kHz, a 5dB boost at 4.01kHz with a Q of 0.10, a 7.8dB cut at 390Hz with a Q of 0.10, and a 0.5dB boost at 104Hz.

**Kick 2 (K2)**

Figure 3.280 Waves SSL Channel on Rms1.
The second kick fed the Space Station at -25.5 dB.

**Snare 2 (Sn2)**

The snare top, snare bottom, and snare sample tracks were assigned to the mono Sn2 auxiliary with a SPL Transient Designer inserted. The snares attack was boosted and the snare ring was lengthened.
The snare was brightened rather significantly with a Filter Bank E6 EQ. The EQ had a high pass at 65Hz, a 0.8dB boost at 256.2Hz, a 4.4dB cut at 335.3Hz, an 8dB boost at 2.01kHz, and a 2.7dB boost at 5kHz.

Sn2 had a send feeding a Space Station at -15.4dB, which was copied to SnComp2, Toms2, and SamToms. There was also a 0.5dB volume boost on the drum fill entering the Figure 3.282 SPL Transient Designer on Sn2.

Figure 3.282 SPL Transient Designer on Sn2.

Figure 3.283 McDsp Filter Bank E6 EQ on Sn2.

Sn2 had a send feeding a Space Station at -15.4dB, which was copied to SnComp2, Toms2, and SamToms. There was also a 0.5dB volume boost on the drum fill entering the Figure 3.283 McDsp Filter Bank E6 EQ on Sn2.
climax of the song.

**Compressed Snare 2 (SnComp2)**

The snare top, snare bottom, and snare sample were assigned to the mono SnComp2 auxiliary track for some parallel compression and addition EQ. The compressor had a threshold of -13dB, a 2.7:1 ratio, a release of 0.36s, and about 3dB of gain reduction. The EQ boosted 8.57kHz by 4.5dB and 1.92kHz by 5.1dB with a Q of 0.10. A 3.1dB cut with a Q of 0.10 was made at 550Hz and a 2.6dB boost was made at 280Hz.

![Waves SSL Channel on SnComp2](image)

**Toms2**

A Renaissance EQ was used on the toms. They had a 3dB boost at 100Hz, a 10.2dB cut at 375Hz, a 6dB boost at 4268Hz, and a 4dB boost at 7885Hz.
SamToms

The sample toms had the same processing and sends as the original toms.

Drum Overheads (OH2)

The overheads had a SSL Channel inserted with an 84Hz high pass, a 4.1dB boost at 14.04kHz, a 4.3dB boost with a Q of 0.10 at 3.92kHz, and a 6.1dB cut with a Q of 0.10 at 310Hz.

Figure 3.285 Waves Renaissance EQ on Toms2.
Drum rooms (Rms2)

A SSL Channel high passed the drum rooms at 84Hz, boosted 14.04kHz by 3.4dB, boosted 3.92kHz by 2dB with a Q of 0.10, and cut 310Hz by 2dB.
Bass (Bs)

The bass D.I. track was duplicated. The duplicate was named BsA and a Decapitator plug-in was inserted. The drive knob was around 2.5 and the style was set to “T”. The tone control was set just past 1 o’clock to brighten the bass and the high cut was just before 3 o’clock to attenuate some unwanted high frequency distortion.
The BsDI and BsA tracks were assigned to the mono Bs auxiliary and processed through the Bs hardware insert. Next, L1 limiter gave the bass 3dB - 6dB of gain reduction with a -4.4dB threshold, a release of 1, and an out ceiling at -7.1dB.

A Filter Bank E6 EQ was used to cut the bass 1dB at 104.8Hz and 2.5dB at 239.9Hz. There was a 1.4dB boost at 1.9khz, a 3dB boost at 1.38kHz, and a 3.5dB boost at 6.16khz.

Figure 3.288 Sound Toys Decapitator on BsA.

Figure 3.289 Waves L1 Limiter on Bs.
The song was rather mellow up until the entry of the big chorus at the end of the song. A Maxx Bass was automated to turn on at the end of the song to give the bass a little more impact. The Maxx Bass low-end signal centered at 80Hz and was set at -9.5dB.

The extremely low sub-harmonics created by Maxx Bass needed slight attenuation so a...
high pass was set at 30.9dB with a Filter Bank F2 EQ. There was also a 18kHz low pass to reduce high frequency noise.

The bass guitar was set at -7.7dB up to the bridge where it was brought down to -9.3dB. After the bridge it moved up to -5.5dB to give the ending plenty of impact.

**Clean Guitar 1 (Cl1)**

A guitar swell melody, Cl1, that played during the verses was panned hard right. Two Filter Bank E6 EQs were inserted on the auxiliary. The first EQ had a 62.5Hz high pass. A 2dB cut was made at 384.5Hz, and a 2dB boost was made at 2.29kHz and 5kHz.
The second EQ had a 3dB boost at 9.15khz.

Cl1 started out at -10.3dB in the first verse. Since Cl1 was the main melody, it needed to be raised 2dB in the second verse to maintain dominance over guitars that were not present in the first verse.

Clean guitar 2 (Cl2)

Cl2 was a clean finger picking guitar part in the verses that was panned to the left 71% and run through the Act 1 hardware chain. An additional 3dB of gain reduction was added with Renaissance Axx. The compressor had an -8.8dB threshold and the attack at 5.00. The output gain was lowered to -2.6dB.

Figure 3.293 McDsp Filter Bank E6 EQ on Cl1.
A Filter Bank E6 EQ was used to make a 3dB boost at 9.15khz.

Figure 3.294 Waves Renaissance Axx on Cl2.

Figure 3.295 McDsp Filter Bank E6 EQ on Cl2.
Clean guitar 3 (Cl3)

Cl3 was a melody played during the chorus and panned left 71%. The Act2 hardware was used to process the guitar part. A few more small adjustments were made with a Filter Bank E6 EQ. There was a 62.5Hz high pass and a 1dB cut at 300.9Hz.

![McDsp Filter Bank E6 EQ on Cl3.](image)

Clean guitar 4 (Cl4)

Cl4 was one of two electric guitar parts that played during the bridge. This guitar part was panned hard left and processed with the TLA 1 hardware insert. Cl4 also have a fader ride during the bridge. The guitar was faded from -15dB to -11.5dB to build into a dynamic peak of the song.

Clean Guitar 5 (Cl5)

Cl5 was the second electric guitar part that played during the bridge. This part was panned hard right and treated with the TLA 2 hardware insert. Cl5 had the same approach to volume automation as Cl4 but it faded up from --14.5dB to -10.5dB.
**Lead Guitar 1 (Ld1)**

The chorus out had two lead guitars. Dr1 was assigned to a mono auxiliary labeled Ld1 and panned hard left. The lead guitar was run through the Gt1 hardware chain.

**Lead Guitar 2 (Ld2)**

Dr2 was the second chorus out lead guitar panned hard right. This guitar was assigned to a mono auxiliary named Ld2 and run through the Gt2 hardware chain.

**Power Rhythm Guitars (Pwr)**

Four heavily distorted guitars (Dr3, Dr4, Dr5, and Dr6) enter the song toward the end of the song following the bridge. Dr3 and Dr5 were panned hard left while Dr4 and Dr6 were panned hard right. All four guitars were assigned to a stereo auxiliary named Pwr and run through the API 1-2 hardware insert. The API EQs had a 4dB boost at 5kHz, a 2dB boost at 800Hz, and a 2dB cut at 300Hz.

Some subtle adjustments were made with a Filter Bank E6 EQ. A high pass was set at 81.1Hz and a 1dB cut was made at 357Hz. There was also a 0.8dB boost at 859Hz, a 1dB boost at 2.40kHz, and a 1.5dB boost at 9.14kHz.

![API 550As on Pwr.](image)
Acoustic Guitars (Ags)

AgBod1 was panned hard left and AgBod2 was panned hard right. Both guitars were assigned to a stereo auxiliary named Ags. The GME 1-2 was inserted on the auxiliary. The acoustic guitars also had approximately 3dB - 6dB of gain reduction from the Renaissance Axx. The threshold was at -12.8dB and the attack was 5.00.

Figure 3.298 McDsp Filter Bank E6 EQ on Pwr.
EQ for the acoustic guitars was done with a VEQ4. The high pass was at 82Hz, a 2.5dB cut was made at 100Hz, and a 2.6dB cut was made at 390Hz. There was a 3.6dB boost at 2.7kHz and a 2.3dB boost at 6.8kHz.

The acoustic guitar performances were fairly dynamic. In order to keep the same balance, each section required a different volume. The first verse was set at -5.3dB, which was brought up to -3.5 for the chorus. The acoustics then came down a 1dB in the chorus out and down 1dB again for the second verse. The performance in the second chorus was played much louder so there was no need for a change in volume until the chorus out, which returned to -4.6dB. The bridge then dropped down to -6.8dB to continue building on the dynamic development into the climax. The acoustics were then brought down to -8.1dB to serve as
support to the electric guitars that drove the ending of the song.

**Upright Piano (UpPno)**

EQ treatment for the piano was done with a SSL Channel. There was a 5dB boost at 8.8kHz, a 5.1dB boost with a Q of 0.10 at 2.69kHz, a 6.4dB cut with a Q of 2.5 at 300Hz, and a 2.1dB cut at 127Hz.

![Figure 3.301 Waves SSL Channel on UpPno.](image)

**Lead Vocal (LdVx)**

The Sta hardware insert was used for the lead vocal. Next, a Renaissance Vox plug-in was set to a -10dB threshold. About 10dB of gain reduction was applied and the output gain was reduced to 3.8dB.
A Massey De-Esser provided about 6dB of gain reduction for harsh consonants around 6kHz.

Figure 3.302 Waves Renaissance Vox on Vx.

Figure 3.303 Massey De-Esser on Vx.
The lead vocal had a send feeding to a stereo auxiliary named VxDl with an Echo Boy delay inserted. The Echo Boy was set to a Memory Man style, a single echo, and a dotted sixteenth note. The MIDI button selected, the low cut was at 10 o’clock, and the high cut was around 1 o’clock. Feedback was set very short.

![Figure 3.304 Sound Toys Echo Boy on VxDl.](image)

The lead vocal also sent to an auxiliary named ChDl for the choruses. On the auxiliary was an Echo Boy delay. A Memory Man style was selected with the saturation turn all the way up. The delay was switched to ping-pong and both echoes were given a quarter note delay. The low cut was right around 3 o’clock and the high cut was around 1 o’clock.

![Figure 3.305 Sound Toys Echo Boy on ChDl.](image)
The chorus delay was high passed at 254.6Hz, cut 9dB at 242.4Hz, boosted 3dB at 1kHz, and boosted 3dB at 5kHz.

General volumes were set for each song section. The verses were at -6.2dB and the choruses were at -5dB. The bridge then dropped down to -8.6dB and slowly faded up to -2.9dB at the end of the song. What was more significant was volume automation on loud breaths and consonants. Depending on the vocal track, compression can bring breaths up to a distracting level. Also, letters like “S” and “T” become very harsh. When the consonants are too loud, the vocal can be painful to listen to even when the majority of the vocal track is too quiet. To remedy the situation, breaths were lowered 7-10dB and consonant letter were lowered 3-4dB.

**Lead Vocal 2 (Vx2)**

Vx2 was a short lead vocal overdub in the bridge. It was panned to the right 28% and processed with the Vx 1 hardware insert. Additional compression was applied with a Renaissance Vox compressor. The threshold was at -4dB to give the vocal another 3dB - 6dB of gain reduction.

A Massey De-Esser with settings identical to the Vx1 De-Esser followed the Renaissance Vox compressor.

A Filter Freak 1 band passed the lead vocal overdub to give it a radio effect.
The second lead vocal sent to VxDl and ChDl at unity.

**Background Vocals (Bgs)**

Two identical background vocals were recorded to continue through the song. The backgrounds, Bg1 and Bg2, were panned 60% to the left and 60% to the right, respectively. A Filter Bank E6 EQ helped to enhance and de-emphasize the background vocals around the lead vocal. The EQ had a high pass at 138.6Hz, a cut of 3.1dB at 444.2Hz, a boost of 5dB at 4kHz, and a boost of 3.3dB at 10kHz.

![Figure 3.307 Sound Toys Filter Freak 1 on Vx2.](image-url)
Then a Renaissance Vox compressor was set with a threshold of -4dB to give the vocal about 3dB - 6dB of gain reduction.

Figure 3.308 McDsp Filter Bank E6 EQ on Bgs.
Lastly, a Massey De-Esser with the same settings as the lead vocal was inserted.

The background vocals had a send feeding VxDI at unity.

**Choir (Chr)**

The remaining background vocals, Bg3 through Bg8, were meant to create a choir sound. They were panned to fill up the stereo field and assigned to an auxiliary named Chr (Choir). Bg3 and Bg4 were panned to the left and right 30%, Bg5 and Bg6 were panned hard left and right, and Bg7 and Bg8 were panned left and right 70%. The choir was then processed with two EQs. The first Filter Bank E6 EQ had a high pass at 138.6Hz, a 1.8dB cut at 392.9Hz, a 3.7dB boost at 4kHz, and a 2.3dB boost at 10kHz.
The second Filter Bank E6 EQ had a 1.4dB dip at 357.8Hz, a 3dB boost at 2.39kHz, and a 2.2dB boost at 9.15kHz.

Figure 3.310 McDsp Filter Bank E6 EQ on Chr.
The choir had a send feeding ChDl at unity.

Figure 3.311 McDsp Filter Bank P6 EQ on Chr.
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