Making Modern Metal: Part 3

Mixing & Mastering

- Mastering • Mixing • Recording

By Mark Mynett

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Damnation's Hammer.

Photo: Daley Wilson Photography, Lancashire

Drawing our mini-series to a close, we take you on a detailed tour of the mix and mastering stages of the Damnation's Hammer album Unseen Planets, Deadly Spheres.

Although mixing is a creative endeavour, the technical aspects are important, especially in the metal genre. Attending to the technical tasks before you mix avoids interruptions to the creative process. For this project, drum edits (discussed last month) were done before overdubs were recorded, and considerable time was spent establishing the right guitar and bass tones when tracking, so re-amping was not required. The only additional performance edits involved the occasional isolated guitar riff, which needed to be super-tight, and a number of vocal timing tweaks.

It's often overlooked by inexperienced producers, but accurate vocal timing can significantly enhance the rhythmic impact of a mix. During our half-day of vocal tracking I'd captured great attitude, but the start points of certain lines or syllables occasionally needed a nudge in the right direction. I also removed headphone bleed and other unwanted sounds between lines, while carefully retaining unobtrusive breath inhalations which enhance the emotional content of a performance. I then went through the guitar and bass tracks, editing to remove hiss/noise from the starts and ends, and during breaks in the performance. Partly due to the cymbal bleed in the tom mics, I also edited the toms, leaving little other than the tom hits themselves with appropriate fades (edits are far more accurate than noise gates).

Having consolidated my edits and crossfades, I labelled the tracks and made unwanted tracks inactive and hidden. I then added timeline markers for the song sections, and created some

In this article...

group busses.

Polarity & Phase

The tonal density required in this style means incorrect polarity settings or comb filtering must be corrected early on: if a track requires polarity inversion or phase-alignment when the mix is under way you'll also need to rework any EQ settings, which severely interrupts the creative flow. So I set about finding the combination of polarity settings that provided the fullest low-frequency reinforcement across the drums as a whole (a subject explored, along with phase, in SOS April 2008 (www.soundonsound.com/techniques/mix-rescue-phase-relationships) and April 2010 (www.soundonsound.com/techniques/phase-demystified).

I used Sound Radix's Time Align plug-in to phase-align the kick and snare spot mics with the kick port-hole and snare top, which were used as the reference 'anchor' (and later used to implement sample reinforcements, which were also phase-aligned) as well as the bass DI and amp-modelled track. Due to the care taken when recording, no phase alignment was required for the rhythm guitars.

Sound Radix's Auto Align was used to phase-align the Helix bass amp/cab-modelled track to the bass DI. Instances on both communicate with each other and calculate the time offset required for the optimum phase relationship between the signals.

The last preparatory technical task was to gate the kick and snare. As a rule, the unfocused off-axis spill captured by kick spot mics seldom benefits the overall drum sound, whereas for some projects, the spill captured by snare mics provides a subtle but valuable contribution. In this instance, hat/cymbal spill on the snare tracks (due to the hard-hitting metalwork performance discussed last month) would be accentuated unhelpfully by the high-frequency boosts I'd knew I'd need to apply to help the snare to punch through the wall of guitars. So I hard-gated the snare top and bottom mics, using the UAD SSL 4000 E channel strip's (reduced-chatter) G2 expander/gate.

Damnation's Guitars

To ascertain the central mix challenges and decide my broad processing tactics, I sometimes start a mix using only pan pots and level balancing — no processing. Other times, I'll go from left to right across the mixer, gating, compressing and EQ'ing the kick, then the snare, and building up the rest of the drums before introducing the bass... and so on. For this project, the band desired a warmer, less clicky drum sound than is typical of modern metal, so to inform my decisions about drum processing and samples, I decided to start with the guitars and bass.

The distortion involved means this style of rhythm guitar has a heavily restricted dynamic range, so many producers/mix engineers, myself included, don't compress the rhythm guitars when mixing. By contrast, it's extremely rare for guitar sounds not to need EQ. The tonal sculpting available on guitar amps isn't very precise, and the other elements in the mix have a significant bearing on what's required of the guitars. Whether EQ is best deployed on individual channels, the mix group or both is a question about which opinions vary. My approach typically involves applying the majority of processing to the mix group, with channel EQ then being used sparingly to manage unwanted resonances and content that are specific to the track. I also tend to use analogue tape emulation for subtle warmth, cohesion and character, usually the UAD Studer A800 plug-in, using 456 tape formulation at 30ips tape speed, and the calibration set to +9dB.

Before EQ'ing, it's important to optimise the balance between the various guitar tracks; significant balance changes later can mean starting your EQ moves from scratch. I started by balancing the double-miked parts, before setting the relative levels of the separate rhythm performances in each speaker. Quad-tracked rhythm guitars tend to sound most effective if you treat two performances (one each side) as a main pair, with the remaining pair tucked in behind, lower in level. I felt the rhythm tracks played through the Celestion Vintage 30s were best suited to the main-pair role. I hard-panned these, and opposition-panned the remaining tracks at about 85 percent width. Symmetry of the whole ensemble is vital, so I then made sure the perceived level in the left and right channels was equal.

For heavy rhythm guitar, I prefer focusing on corrective EQ cuts rather than boosts, so I applied a number of surgical cuts to attenuate abrasive qualities in different areas of the individual guitar channels. Turning to rhythm guitar bus EQ, using the UAD SSL E Series plug-in's slightly cleaner-sounding 'black' mode, I applied an high-pass filter at the point where the cabinet thump transitioned into texturally 'harder', more controlled sonic weight (about 85Hz). Similarly, a low-pass filter at roughly 7.5kHz marked the boundary where pleasing aggressive brightness transitioned into raspiness or fizz. Fairly broad mid-range cuts at 260Hz and 1.3kHz were also dialled in, with their depth and width later fine-tuned to suit the context provided by the bass and the rest of the mix.

Corrective cuts tend to be more effective than boosts on heavy rhythm guitar tracks. A number of surgical cuts removed abrasive qualities in the individual guitar channels, while broader cuts were applied to the rhythm guitar bus.

These moves helped reveal the low-end weight and top-end presence of the guitars, but pockets of low-end resonance still compromised their tightness. There was also some unwanted high-end abrasion. To address this, I used Universal Audio's Massenburg Design Works (MDW) EQ: an extremely narrow notch at 131Hz was joined by a broader cut at 224Hz, and another surgical cut at 3.5kHz (with heavy guitars, the 3-4 kHz region typically contains some 'abrasive' content). The 131Hz notch alleviated the most obvious resonance, but there was still problematic 'chug-thump' resonance during palm-muted sections. Much like sibilance, chug-thump is a dynamic, frequency-specific issue; deepening the corrective EQ would leave the low frequencies in the non-palm-muted sections under-represented. I tackled this using the Brainworx bx_dynEQ V2 dynamic EQ plug-in, centred at 131Hz, with near-instant attack.

Universal Audio's Massenburg Design Works EQ.

The guitar solos were surprisingly easy to knock into shape, simply requiring a healthy dose of compression (medium attack, fast release, about 6dB of gain reduction) and a touch of additional brightness. Something that later proved straightforward, yet effective, was sending these solos to the same effects I'd set up for the vocals, with broadly similar send levels.

The Brainworx bx_dynEQ V2 tackled problematic 'chug-thump' on the rhythm guitars during palm-muted sections, but without compromising the guitar tone at other times.


The SSL AWS924 console at the heart of the Damnation's Hammer mixing sessions.

Photo: Alex Beldea (wwwalexbeldeacom)

A Bass Sound To DI For

Due to the prominent low-end content of the guitars, the bass guitar was tracked with greater emphasis in the low-mids than the lows. This reduced the potential for frequency masking and exploited the low-mid 'hole' in the scooped guitars. Also worth highlighting is that Jamie uses his fingers rather than a plectrum, and has a really powerful, percussive playing style. To prevent the percussive note attack/high-frequency 'clank' overshadowing the guitars' note onsets, or appearing disconnected from the kick hits, the bass part's dynamics required careful management.

I decided to employ 'frequency bracketing', a technique whereby high- and low-pass filters confine component signals to specific frequency roles. This often comprises separate tracks bracketed to deliver the lows, mids and highs, respectively, or sometimes just lows on one track and low-mids upwards on another. The amount of compression applied to each track can then be manipulated according to its frequency role, leaving you less reliant on creative EQ to achieve the desired sound.

In addition to a distorted signal, bracketed to focus on the mids, the bass sound consisted of three tracks: one DI, restricted to the sound's low frequencies (a); a Line 6 Helix-modelled track focused on the low-mids (b); and another DI for the highs (c).

I had a bass DI and (modelled) bass amp/cab signal to play with, but chose to duplicate the DI track twice (once for a dedicated distortion channel), to give me four bass channels to sculpt and blend. You might assume that the amp/cab modelled signal would be the preferred source for the lows, with the DI processed to fine-tune the bass's upper frequencies, but the reverse is often more effective. I applied high- and low-pass filters at 96 and 285Hz respectively to the tight, controlled DI track, with the comparatively 'lumpy' amp/cab signal delivering the mids between 305 and 900 Hz, the combination delivering a tighter, more focused sound. The first duplicate DI was then filtered at 950Hz and 4.5kHz and blended in, to deliver a clear, wiry note definition. I focused the distortion channel, processed with UA's Ampeg SVT-3 Pro plug-in, on the upper mids, cutting everything but 600Hz, 900Hz and 2kHz on the plug-in's graphic EQ.

The key to successful frequency bracketing, especially with bass, is compression that's empathic to the role of each track. This often involves experimenting with post- and pre-EQ compressor placement. As the DI track assigned to the lows no longer contained any mid-to-high-frequency attack, I could compress this radically: I opted for pre-EQ compression, using a UAD 1176 Rev E, with a fast attack (which, with a broadband bass part would normally flatten/dull things excessively), and this provided a really stable low end. The post-EQ compression applied to the mid-focused amp/cab track was set to a medium attack with less gain reduction, retaining the transient detail.

UA's Ampeg SVT-3 Pro plug-in was employed on the bass DI.

A side-benefit of the frequency bracketing was that I had easy and immediate control over the level and dynamics of the top-end attack and finger-clank. A post-EQ, high-ratio, high-threshold compressor with a fairly fast (3ms) attack and a medium release targeted only the signal peaks, and was sufficient to smooth out this signal.

To provide a kick sound with solid, automatable low-frequency content, the Subkick track was treated to fast-attack, fast-release, aggressive gain reduction, via the UAD 1176 Rev E plug-in.

Frequency bracketing can, however, lead to a somewhat synthetic sound, and close attentionis required to avoid this. With the bass playing alongside the guitars and work-in-progress drums, I spent plenty of time adjusting the crossover points and slopes, compression settings and level of each track. My aim was to ensure the guitars and bass didn't fight each other for space; I wanted each to appear loud, clear and present. The only treatment I applied at this stage to the bass group was the UAD Studer A800 tape sim, followed by 4-5 dB gain reduction using the UAD LA-2A Silver compressor plug-in. Later, I added a low-mid (385Hz) cut using the SSL desk EQ.

Drumming Up Business

Having coaxed the desired impact from the guitars and bass, I could focus on the drums. But first, I activated the SSL desk's master-bus compressor. Mix-bus compression is a valuable element in my mix workflow, and typically I'll engage the compressor relatively early in the process, leaving it active when printing the final mixes. Largely due to its VCA design and very wide knee when set to a 2:1 ratio, the SSL bus compressor is renowned for its ability to 'glue' a mix together in a smooth, transparent manner. I tend to use a 10ms attack and the programme-dependent release, with the threshold adjusted to provide up to 3-4 dB gain reduction during peaks. As a mix progresses, I revisit the threshold regularly to ensure this nominal level of gain reduction is retained.

Kick-ass Kicks

Having already gated and phase-aligned the kick and snare spot mics, I began applying compression and EQ to these tracks. Starting with the Subkick, I used quite heavy-handed, high-ratio compression and, as there was no top end to compromise, a fast attack. As the mix evolved, this channel was treated to level automation too. During song parts that featured straighter or slower kick patterns, with more space for lower, slower frequencies to decay, I could place greater emphasis on this signal, to add more low-end weight.

As is often the case with port-hole mics, the Sennheiser 602 track required significant corrective EQ. A broad, deep cut at 265Hz addressed some boxy low-mids and, as the Subkick was taking care of the lows, I applied a 120Hz high-pass filter to deal with an undesirable low-frequency resonance. Given the corrective nature of these EQ moves, I opted to place the compressor after the EQ, with a high threshold, high ratio and medium attack time delivering roughly 4-6 dB of gain reduction. A similar compression approach, also post-EQ, was applied to the Sennheiser e901 'attack mic': the EQ featured a broad, deep cut at 541Hz and a wide, chunky boost at 7.5kHz to emphasize beater attack.

After optimising the balance of the three kick channels (in the context of the bass and guitars), I applied 4-5 dB of medium-attack, fast-release compression to the kick group, followed by with a broad 4dB 400Hz cut using the desk EQ.

Designer Snare Sound

The level of metalwork (especially hat) bleed in the snare mics restricted the ways I could develop the sound. Gating let through the hat or cymbals if they were struck at the same time as the snare, and while de-essing at 4.5kHz helped contain the hi-hat spill, any significant compression or high-frequency boost brought it back. Nevertheless, subtly emphasising the attack of the snare using an SPL Transient Designer plug-in helped, and a UAD Studer A800 on the snare group enhanced the snare's body and character. The metalwork spill on the snare bottom mic was less intrusive, so I could apply 5-6 dB of medium-attack compression, and, using the musical-sounding UAD Pultec EQP-1a plug-in, add a couple of decibels of 8kHz high-shelf boost without too many unwanted consequences.

Parallel distortion, courtesy of the Vertigo VSM-3 plug-in, brought the acoustic snare channels closer to the target sound in a way that EQ couldn't.

Despite this, when played alongside the layered walls of guitar and bass, the snare lacked presence, punch and crack. Though I knew I'd use sample reinforcements later, I prefer to rely on the samples as little as possible; the wide dynamics and timbral variations of a snare performance make it the kit piece that's most likely to reveal sample use to the listener, and obvious sample replacement usually indicates amateur production standards. Intent on taking the acoustic snare sound as far towards the target sound as possible, I turned to one of my favourite snare production tactics: parallel distortion. I routed the top and bottom snare mics (mainly the top) to an aux track with an instance of Brainworx Vertigo VSM-3, an analogue-modelled saturation plug-in. Its unique '2nd Harmonic FET Crusher' and '3rd Harmonic Zener Blender' harmonic-generation circuits provide powerful control over the distortion that's produced. Using the solo function to monitor only the generated content, I adjusted the second- and third-harmonic input filters so the mainly upper-mids and highs were added. After time spent finessing the drive, shape and THD Mix parameters, and with the parallel signal introduced at an appropriate level, the snare sounded fuller, punchier and more able to punch through that wall of bass and guitars!

Tom Character

I'd edited the toms to remove spill during my mix preparation stage. Now I wanted to work towards ensuring they had tonal qualities similar to the punchy yet warm and full kick, to create a cohesive drum sound. The tom tracks mainly comprised solid fills so the dynamics of each were relatively consistent, and given the attention paid to tuning before tracking, their attack or decay didn't need modifying. The toms slightly lacked vibe and character, but this was quickly remedied through moderate application of 6:1 medium-attack, medium-release compression using a UAD Tube-Tech CL1B plug-in, with another Studer A800 over the toms group bus.

For low-tuned larger toms, a similar EQ approach to the kick can be effective. The most important spectral modifications to the third and fourth toms were extensive cuts to the low-mids at 402Hz and 534Hz, respectively. As well as tackling boxiness, these cuts revealed the low-end weight and top-end attack, and these regions were further emphasised with wide-Q boosts on tom four (4.5dB at 85Hz and 3.9dB at 3.33kHz) and tom three (5.2dB at 115Hz and 6.1dB at 3.85kHz).

The UAD Studer A800 tape plug-in was my primary mix tool for enhancing body, warmth and presence on several sources.

Rack tom two similarly benefited from a deep mid-range cut (670Hz), with broad emphasis of the lows and highs at 133Hz and 3.96kHz, and the smallest tom from a deep cut at 790Hz and broad boosts at 183Hz and 4.52kHz. With the toms panned from audience perspective low to high, and with a number of fills auditioned with the whole mix playing, I made a few level tweaks and was then ready to hit the metalwork.

Applied to the toms group, the UAD Tube-Tech CL1B not only delivered improved signal stability, but also enhanced vibe and character.

Metalwork Class

Because cymbals typically deliver the drum-pattern subdivisions, they contribute significantly to a metal song's drive and energy, so they're important. Even so, unless the sounds themselves, their engineering, or the acoustics of the recording space were defective, the hats, ride and cymbal tracks are unlikely to require the same level of detailed EQ as the shells; high-pass filtering is usually sufficient corrective EQ, and any additive EQ tends to be relatively subtle.

Nevertheless, I needed to make a creative decision about the role the metalwork would play in these mixes. Should these tracks mainly deliver the cymbals, or be asked to assist with the snare sound too? Well, the amount of snare captured by the metalwork mics was slightly compromising my (by now, carefully constructed) snare sound, so I decided to treat the metalwork channels as cymbal mics and applied a 550Hz high-pass filter to each. I tried a similar approach with the hats, but realised there was sufficient hat sound coming from the other metalwork tracks already, so I simply made the hi-hat channel inactive and hid it from view.

Bringing In Reinforcements

Now I was ready to treat the Coles 4038 room mics (discussed last month) and apply some parallel compression, but to provide context, I first attended to sample reinforcement of the kick and snare. Modern, heavy music production always requires drum samples. For the drummer's intentions to remain clear, and for the kick, snare and toms to contribute effectively to the production's heaviness, they need to punch through a dense wall of bass and distorted rhythm guitars. When balanced at a suitable level without sample use, even a credibly engineered, high-standard, hard-hitting drum performance on a top-quality well-tuned kit might still fail to do this, regardless of what processing is applied. Even very subtle reinforcement with samples can make enough difference. Regardless of the extent of their contribution though, samples must be used discreetly!

As discussed last time, the drum tracking session included a multitrack recording of clean hits on each kit piece, from which I could create samples that shared the same acoustic space, sound sources and mics as the performance. The idea was to facilitate a far more natural and unique result than you could achieve when relying only on library samples. In a similar process to mixing a song, I bounced the tracks down to create single samples. For the kick, I only ever use very hard, single-velocity hits. But to avoid audible 'machine-gunning', I exported multiple kick hits of the same force, which would be triggered randomly. I took largely the same approach with the snare, but created samples from hard rimshots and exported multi-velocity samples. For every hit, I started with the spot mic signals, and introduced the overheads for a further sample, then the Blumlein-pair room mics for a third. This involved experimenting with balance and polarity, fade-outs to control each hit's decay, and moderate EQ, just as if I were mixing.

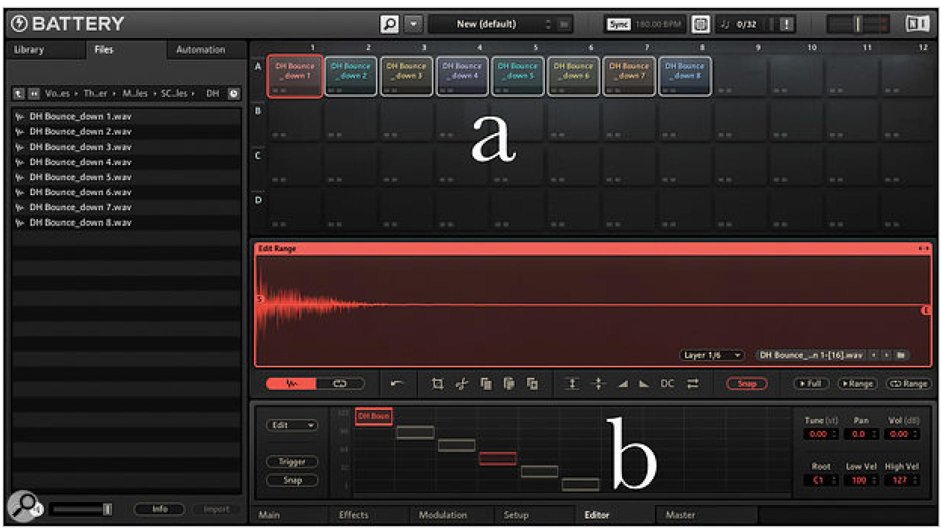
While this painstaking process might strike you as being time-consuming, in my experience it tends to be time well spent, because on auditioning the resulting files, I'm often surprised to discover which sample is most effective in the mix. Reinforcing a kit component with a very tight dry hit, created from just the relevant spot mic, might be all that's required to help an element cut through the mix. Conversely, a more ambient sample might introduce a far better sense of three-dimensional size than is enabled by reverb processing.

Trigger Happy

Massey's DRT software converted the drum audio to MIDI, which was used for sample triggering.

I've used the majority of sample-trigger approaches over the years and I still find audio-to-MIDI conversion — with the resulting MIDI information used to trigger samples — the most reliable (credit where it's due: this was recommended to me by Andrew Scheps!). Not only does this avoid spill concerns in the trigger source, but potentially allows greater dynamic control over the samples than triggering from audio. For example, during sections featuring quieter snare dynamics, the velocity of the MIDI notes can be easily adjusted to prevent sample use sounding obvious.

Using the Massey DRT software for audio-to-MIDI conversion, I selected the D112 kick track (to which the other kick tracks had been phase-aligned) opting to level out the trigger velocities. For the snare-top-triggered MIDI track, I largely retained the natural performance dynamics but raised the velocity of a few faster parts where the performance hits were lacking.



With multiple hits of the same velocity triggered randomly (a), and multi-velocity hits triggered according to MIDI velocity (b), NI Battery 4 allowed the album's snare reinforcements to retain a human feel.

Having created the kick and snare MIDI tracks, I inserted Native Instruments' Battery 4 drum sampler and started auditioning the drum samples I'd created earlier. For the kick, it was clear that the 'driest' versions, created from just the three (attack, port-hole, Subkick) spot-mic tracks offered the best impact. To avoid machine-gunning, I set Battery to trigger the same dry kick samples randomly, then blended the channel with the acoustic signals. After attenuating the sample channels' low mids around 400Hz and moderately accentuating the attack at 3.8kHz, I was close to my target sound. Later in the process, when the drums were auditioned with the bass and guitars, I felt the kick lacked a little density and 'point', and remedied this by blending in a Steven Slate sample ('Tommy Kick') at a moderate level behind the first sample.

With the snare reinforcements, the most ambient sample (created from the spot, overhead and room mics) made the sound punchier, yet somehow more three-dimensional. Here, I was triggering multiple-velocity samples, but I again tucked a library sample (the EZ Drummer 'John Tempesta' snare) 'behind' my own, to provide a little more 'point'. A touch of the parallel snare distortion I'd set up earlier benefitted both samples — and, finally, I was happy with the snare sound!

Back In The Room

Despite the control the spill removal and samples had given me, when auditioning the results in the context of the mix, the different kit pieces seemed a little isolated from each other. My quest for a more natural, believable drum sound brought my attention to the room mics and parallel compression. While these are usually the best way to provide 'glue', it's important to note that excessive use of these cohesive signals can undesirably soften a clear, punchy drum sound, especially when aggressive compression is involved. To avoid this, my approach tends to involve very fast-attack compression and radical gain reduction for the parallel drum tracks, with more moderate gain reduction on the room mics.

As mentioned last month, the Blumlein-pair room mics were the only drum signals I'd printed with compression and analogue processing. Listening back to them, the printed compression delivered exactly what was required, so I didn't apply further compression, but the slight harsh/abrasive cymbal qualities in the room mics led me to filter the highs aggressively, with a 2kHz, 6dB-per-octave low-pass filter. This improved matters, but the punch and tightness of the kick in the composite drum sound was slightly compromised, so I quickly corrected this with a 185Hz high-pass filter on the room signal. Hard-panning the processed Blumlein tracks and mixing them in at a suitable level gave the drums a much improved sense of cohesion and dimension, without sacrificing any of the punch I'd coaxed from the spot mics.

When it came to parallel compression, I needed to avoid reintroducing any abrasive cymbal qualities, so while I took a parallel compression feed from the kick group, I was careful to take feeds only from the (spill-free) snare samples. I also used the edited tom tracks, but took no feed from the metalwork.

My preferred plug-in for parallel compression is UA's Neve 33609, a model of Neve's diode-bridge compressor. With its compression and limiting stages engaged, and the dual-channel controls linked, I selected the compressor's fastest 100ms release setting and a 6:1 ratio (the attack time is fixed). On the post-compressor limiter section, I chose fast 2ms attack and 50ms recovery times. The two sections combined to treat the parallel signal to roughly 18dB of gain reduction.

Given the low levels of spill in the signals sent to the parallel compressor, generous post-compression EQ could be used to emphasise weight and brightness in the drums, without introducing any abrasive cymbal content.

Introducing this compressed parallel signal to the mix gave the drums a lovely density and vibe without robbing the kick of impact. Quite the opposite, in fact! So, given the less clicky kick the band sought, I tried amplifying the lows of the parallel signal, settling on a post-compression low-shelf boost of several decibels at 190 Hz. Similarly, several decibels of 6.9kHz high-shelf boost emphasised the brightness and punch of the shells nicely.

Drum Reverb

Knowing the room mic and parallel compression would have an impact on the ambience of the drums, I'd left the reverb until last. In addition to my usual short, dense UAD EMT Plate 140 patch, I experimented with a Relab LX480 medium-sized room. The plate had a 10ms pre-delay and 850ms decay, but I used automation to increase these times when there was space in the beat to warrant doing so. The room reverb had a 15ms predelay and 1.4s decay, and to enhance the 'air' in this reverb I applied a 7.5dB, pre-reverb, high-shelf boost at 11kHz.

I spent plenty of time experimenting with the choice of sources and send levels for the two reverbs. The snare samples provided a stable, spill-free feed, but lacked some spit and attitude, which I found in a snare-bottom feed. Sends were also routed from the toms (less from the floor toms, given their longer decay), and just a touch from the kick samples, metalwork and parallel compressor.

Vexing Vocals

The main focus of the vocal recording had been capturing a performance with the necessary attitude and energy, and we'd managed this without double-tracking. But when it came to the mix, the vocal just seemed a bit narrow and lost in that wide, dense wall of electric guitars.

Partly due to the compression I'd printed during tracking, the sibilance (at around 9.3kHz, fairly high for a male voice) was quite pronounced, but UA's Precision De-esser quickly took care of that, delivering 7-8 dB gain reduction on sibilant peaks. I was pleased with the consistency and presence of the vocal when appropriately balanced in the mix, but felt a little more compression would improve intelligibility throughout. An instance of UA's Empirical Labs

Distressor plug-in, with fairly fast attack and release times, set to deliver up to around 5-6 dB of gain reduction did the job.

The vocal had already been printed with nearly 20dB of carefully constructed series compression, so mix-stage compression was conservative: 5-6 dB of gain reduction was sufficient. The Distressor's 'band emphasis function' helped suppress the vocal's mid-range content, and 'Dist 2' introduced a touch of second-harmonic warmth.

Powerful vocal performances often contain subtle but valuable energy below the fundamental, so I set my high-pass-filter down at 80Hz. Nothing below there is likely to benefit a vocal, but I could be sure there were no LF nasties creeping through. To deal with some annoying resonances, I applied two several-decibel surgical cuts: one at 608Hz, the other at 3.77kHz. After sweeping an EQ boost to locate the body and warmth of the vocal (usually in the 125-300 Hz range), I applied a broad, subtle (1.2dB) 140Hz boost.

My next EQ consideration, brightness, is critical to placing a vocal in a heavy mix. As a guiding principle, you need to broadly match the brightness of the vocal to that of the guitars, while ensuring these instruments don't battle for dominance. Needless to say, these EQ moves should be judged in context. Sweeping in the 3-7.5 kHz range told me that the most constructive area of non-harsh brightness began around 6.8 kHz, so I applied a smallish (2.3dB) shelving boost there. A further high shelf boost (3.1dB) at 10kHz emphasised 'breath and air'.

Despite this compression and EQ, the vocal still needed greater density, which I achieved via parallel distortion using SoundToys' Decapitator plug-in. Only the subtlest introduction of this parallel channel was required to add the necessary richness and impact.

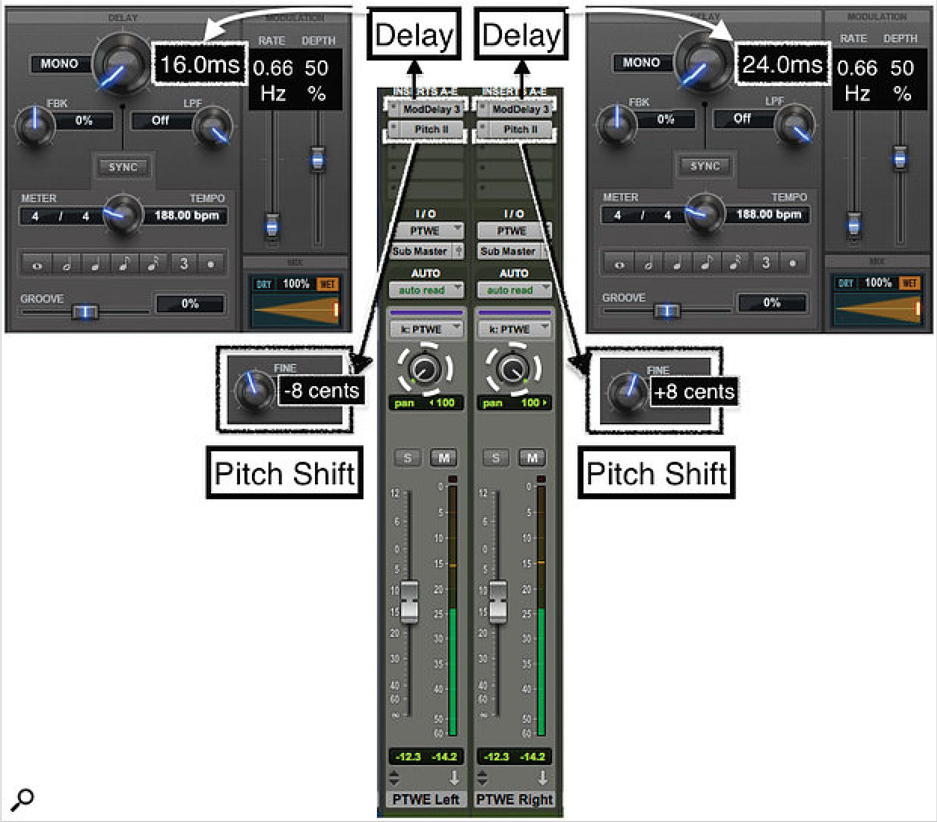
Vocal Effects

Now I could apply the all-important vocal effects. As with the drums, I set up two reverb channels. The first was a UAD RMX16 plug-in's Plate patch, with a 1.2s decay and no pre-delay. The second was markedly different: a kind of 'slapback reverb', using a 1.6s room from the Relab LX480 plug-in, with an eighth-note pre-delay programmed in milliseconds (for the 'Temple Of The Descending Gods' track, which averages around 189bpm, this was 159ms). The first reverb was effective for pushing the vocal 'backwards', and my slapback reverb added depth and dimension while keeping the vocal close to the listener.

I then set up a mono slapback delay before turning to multi-tap delays. The first started life as a quarter-note delay on each side, but I increased one side by 5ms and lowered the other by the same amount. With moderate feedback this was a musically appropriate repeat, but the left/right differences subtly widened the vocal — a useful effect in the context of that wall of guitars. Low and high cuts were applied in the delay plug-in itself (SoundToys Echoboy) and these, combined with (post-delay) tape emulation, prevented the delays sounding too obvious. The other multi-tap delay used a half note on the right and a dotted half-note on the left, both with quite strong feedback. Again, I low-pass-filtered this, but to 'gel' this delay more assertively into the mix I followed it with a healthy dose of SoundToys Decapitator's Punish setting.

The last effect was some classic pitch-thickening width enhancement [shown below], courtesy of two opposition-panned mono aux tracks, each with a mono delay plug-in. These zero-feedback delays, at 16 and 24 ms, put the average delay at 20ms — similar to the traditional ADT echo. A post-delay pitch-shifter on each return increased the delay pitch of one by eight cents and decreased the other by the same, thickening things while leaving the average pitch unchanged.

Having introduced seven different vocal effects (parallel distortion, two reverbs, slapback echo, two multi-tap delays, and pitch-thickening width enhancement) I spent quite some time experimenting with the contribution from each to ensure that, while each did its job, the vocal as a whole didn't sound overly processed.



A classic ADT-style pitch-thickening width-enhancement tactic thickened and widened Tim's vocal.

Final Touches & Mastering

With the mix now working nicely, I spent quite a while tweaking the processing and levels, at this stage using the studio's SSL AWS 924 desk EQ rather than software, only turning to automation when I was entirely happy with the static levels. After completing each mix, and with the desk's master bus compressor active, I printed each song back to Pro Tools, ready for mastering. Given mastering's reputation as something of a 'dark art', you might be anticipating an account featuring a whole range of complex multiband, parallel and Mid-Sides processing techniques. But mastering proved to be the simplest and least challenging stage of the whole production.

The initial part of the process involved experiments with mastering processing, to highlight where the mix itself could be improved. For example, after applying mastering compression to the first printed mix, I tried out various mastering high-pass filter positions. Through this, I realised that although the guitars and bass presented no low-frequency problems, the kick benefited from a slightly higher cutoff. Similarly, mastering compression lifted the rhythm guitars, drum reverb tails and vocal effects just a touch more than ideal. Rather than try to adjust the mastering processing to compensate for such issues, I returned to each mix to tackle them at source.

A few cycles of this process brought me to a point where the only mastering EQ (other than nominal 39Hz high-pass and 20kHz low-pass filters) was a 1.4dB low-shelf dip at 68Hz, just to slightly tighten up the low end, and a broad 0.7dB peak boost at 3.3kHz. In each instance, I felt the UAD Manley Passive Mastering EQ delivered a preferable result to doing the equivalent moves in the mix.

From here, I moved onto compression, employing just the VCA (not the optical) side of the UAD Shadow Hills Mastering Compressor. Combining the compressor's lowest 1.2:1 ratio with a very low threshold made it responsive to the signal's body, rather than its peaks. To retain signal transients, I used the longest available attack time (30ms), along with a relatively fast release, providing approximately 4-5dB of gain reduction during peaks. A side-chain high-pass filter prevented energy-rich frequencies below 90Hz from 'pumping' the compressor.

The album's mastering signal chain.

I played with the modelled transformer options to see if they added anything worthwhile. Liking the coloration provided by the 'Steel' setting, I experimented with further saturation processing using the UAD Ampex ATR-102 Mastering Tape Recorder. Any tape speed lower than 30ips seemed to have too much impact on the mix's HF fidelity, so 30ips remained my default setting, and the quarter-inch 456 tape was the most pleasing setting for the highs. Only a touch of this 'analogue polish' was required, so I kept the tape-emulation's input gain fairly low and adjusted the Reproduce parameter so the plug-in didn't add or remove any gain overall.

Loud & Proud

Last, but not least, came the limiter. Less experienced engineers often exaggerate the role of limiting and loudness maximisation when mastering metal. When over-used, a limiter's fixed high ratio and fast attack take sharp, punchy transient energy and flatten it, leaving a comparatively blunt and lifeless production, often with unpleasant upper-mid distortion. The best approach to loudness (should that be your goal) is a holistic one, in which no single process during mixing or mastering is asked to do too much. In simple terms, gradual accumulative dynamic control through the mix and mastering processes will leave you needing only relatively mild limiting to achieve competitive loudness.

Still, different limiters react to and process audio in different ways. So, ensuring they were loudness-matched, I carefully compared various limiters. For this production, the Sonnox Oxford Inflator delivered the nicest result. After experimentation, I eventually left the effect parameter 100 percent, but reduced the curve parameter to prevent the peaks and highs being overly softened.

So what did this look like in terms of loudness? Well, I actually provided a couple of differently optimised versions for the band. The average integrated loudness of the CD album is roughly -6.5 LUFS, and its PLR (peak-to-loudness — an indication of dynamic range) value around 7.5 — similar figures to those for Metallica's latest album *Hardwired...To Self Destruct* and Jens Bogren's production of Kreator's last album *Gods Of Violence*. I also provided files intended for loudness-normalised playback systems, with a much lower integrated loudness of -16 LUFS, and a PLR of around 12.5.

Signing Off

And here, I have to draw this journey to a close. The band and I were happy, and it seems our satisfaction was justified — as this final instalment was going to print, Damnation's Hammer agreed a three-album contract with a significant independent label, and *Unseen Planets, Deadly Spheres* will be released internationally later this year. I hope you've found the series useful, and gained some insights and ideas to consider in your own productions, be they metal or otherwise. Thanks for reading!

Making Modern Metal: Part 1 Pre-production

Making Modern Metal: Part 2 Recording

Making Modern Metal: Part 3 (of 3) Mixing & Mastering

About The Author

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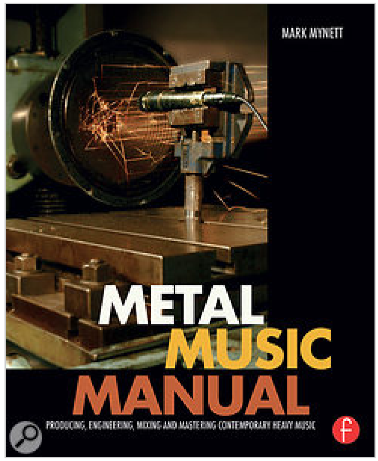

Audio & Video

You can download the album's opening track 'Temple Of The Descending Gods' for free from www.mynetaur.com, and you can find Damnation's Hammer's Bandcamp page at https://damnationshammer.bandcamp.com.

I've also produced a video detailing a number of processing aspects from the mix stage of this album, and you can find this below or on YouTube at https://youtu.be/OY__Tgu9ZMg.

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