CHAPTER SAMPLER

The Art of Mastering in Music

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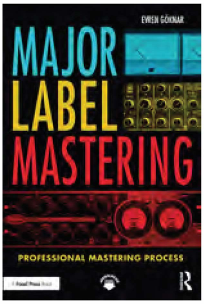
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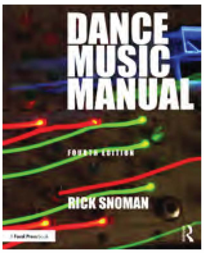
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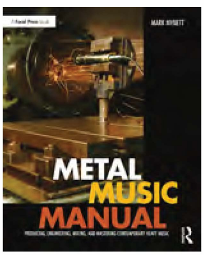
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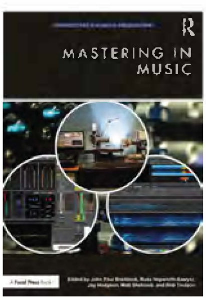
MASTERING IN MUSIC

CHAPTER 4

Fundamental Mastering Tools and The Primary Colors of Mastering

I've reviewed how an effective mastering studio represents a symbiotic component of the entire mastering system. Next, we will focus on the equipment used to make the tonal or surgical adjustments and sonic enhancements. These include the following five *fundamental tools* for audio adjustments: *analog-to-digital* and *digital-to-analog* converters (AD/DAs), *equalizers* (EQs), *compressors/limiters*, *expanders*, *brickwall limiters* (BWLs), and *digital audio workstations* (DAWs). I refer to three items on this list—EQs, *compressors* and BWLs—as The *Primary Colors of Mastering* due to their seminal importance in audio mastering. In this chapter, I will review the function and desirable attributes of these *fundamental mastering tools*.

ANALOG-TO-DIGITAL AND DIGITAL-TO-ANALOG CONVERTERS (AD AND DA)

As their names suggest, these devices convert audio stored in the analog (signal-based) domain to the digital (sample-based) domain, and vice versa.[1] Digital audio must first be converted to analog signal to be heard or used with analog equipment. At minimum, a professional mastering studio requires two or three *DA converters* and one *AD converter*. Two *DA converters* are needed for the PBDAW—one for monitoring flat or unprocessed source audio, and another for the channel path to make adjustments and enhancements to the audio. A third DA is needed to monitor the final processed audio from the RDAW. If you use a modern mastering console that features a *mult* (duplicate) of the input signal, then you only need two *DA converters* for the mastering system, as the PBDAW signal can be accessed after the *mult* for both the flat audio monitor position and the channel path. Additionally, the system requires one *AD converter* to convert the signal from the analog processing chain to digital and then into the RDAW.

Ideally, a mastering-quality *DA converter* will present audio that is focused and accurate without adding coloration, distortion or enhancements. It should be able to convert all common digital audio formats[2] and resolutions to analog transparently. A mastering-quality *AD converter* will also accurately sample and digitize the analog signal into the digital binary code—free from clock jitter and aliasing filter artifacts—at all common sampling frequencies and bit depths. It should preserve transients and dynamic range exactly as they are presented by the analog signal.

Mastering Reference Levels

DA and *AD converters* (Figures 4.1–4.5) usually include onboard level adjustments for calibrating the chosen operating *reference level* for the mastering studio. This calibration correlates digital *decibels full scale* (dBFS) levels with analog voltage readings and the VU meter

Figure 4.1 The Lavry Gold AD122–96MX is an exemplary mastering AD converter. It is sonically transparent and features low jitter, musical clipping properties near 0dBFS, soft limit and soft saturation functions for loudness enhancement, full scale metering, and onboard test tones.

Source: (courtesy Lavry)

Figure 4.2 The Lavry Quintessence DA converter with onboard monitor control.

Source: (courtesy Lavry)

Figure 4.3 The Lavry Blue 4496 chassis with one AD converter and three DA converters. These converters are also feature-rich and work great in a mastering context.

Source: (author collection)

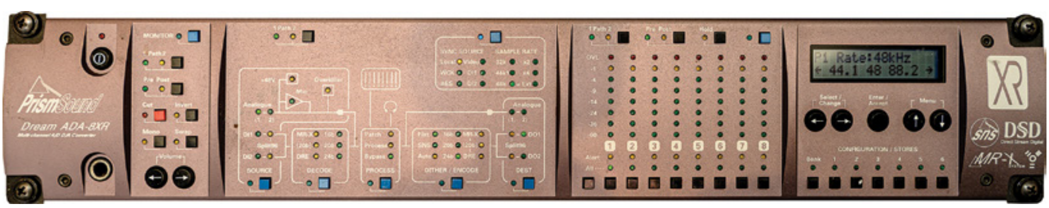

Figure 4.4 The Prism ADA-8XR mastering-grade AD and DA converter with monitoring/headphone functionality. A modular system available in multiple configurations, this unit has the 8AD plus 8DA setup. Also has the 'over-killer' limiting option for creating louder masters.

Source: (courtesy Capitol Mastering)

Figure 4.5 The Crane Song HEDD (Harmonically Enhanced Digital Device) features high-quality AD conversion with DSP-modeled analog/tube coloration (triode tube, pentode tube, and analog tape compression options).

Source: (courtesy Crane Song)

Table 4.1 Shows the relationship between a 1kHz tone dBFS level, its output in voltage at the DA converter, the aligned voltage at the DA converter, and the VU meter alignment. Represents an overview of reference level calibration of the mastering studio. The Dangerous Convert-2 has a preset level calibration selector for these three common options (see Figure 3.7).

Mastering Studio Reference Level Calibration			
dBFS (1kHz at RDAW)	**Volts** (at DA/Adjusted DA)	**VU Cal**	**dBu**
−18	.775/1.23	0	+4
−16	.975/1.23	0	+4
−14	1.23/1.23	0	+4

so that lower dBFS levels (adjusted at the DA converter to 1.23V = 0VU = +4dBu) raise quieter more dynamic source files for a more robust level through the mastering system. The *reference level calibration* is accomplished by playing a 1kHz tone at the PBDAW at the selected dBFS level (usually −14dBFS), verification of voltage level at *DA conversion* (1.23V), then to the AD converter, the RDAW and post RDAW *DA converter* for end-of-chain monitoring. *Reference level calibration* is critically important for establishing *headroom* (region of signal between a nominal level and clipping—often 0dBFS digitally and +24dBu analog) in a mastering system, and is usually calibrated as one of three options (Table 4.1).

I have always used −14dBFS = 1.23V = 0VU = +4dBu as my mastering *reference level*. Most mastering studios are similarly referenced, especially when working in modern pop, rock or hip-hop genres.

EQUALIZERS (EQ)

An EQ is one of the three critical *primary colors of mastering*. EQ allows you to boost or cut selected frequencies and bandwidths. As musical instruments possess their own range of frequencies, this tool allows the Mastering Engineer to either feature—or draw attention away from—an instrument or group of instruments. Larger bandwidth or shelving curves are utilized for tonal balancing across larger areas as needed, and narrower bandwidths are useful for detailed or surgical moves. The power of equalization to enhance the fidelity and presentation of recordings remains a profound component of audio mastering.

Equalizer Configurations

There are various types of EQ—*filters*, *shelving*, *graphic*, *fixed parametric* and *parametric* are most of the EQ types you will encounter as hardware devices or plug-ins.

Filtering EQs either remove frequencies above a selected frequency—known as a low-pass filter (LPF), or below a selected frequency—known as a high-pass filter (HPF). The slope of the filter can be adjusted in a *dB/octave* ratio, with 6dB/octave representing the most gradual slope and 24dB/octave the sharpest. The filtering slopes are in multiples of 6dB, with each of four options referred to as an *order*: first order is 6dB/octave, second order is 12dB/octave, third order is 18dB/octave, and fourth order is 24dB/octave. These are excellent for managing high-frequency issues or low-frequency rumbles (necessary for vinyl cutting), and are also useful in conjunction with *shelving EQs* to keep boosts from becoming excessive and causing a tonal imbalance in the high or low frequencies.

A *shelving EQ* boosts or cuts all frequencies above or below a selected or fixed frequency, making it suitable for tonal balancing of highs or lows. *Baxandall EQs* are a shelving EQ curve adopted and mass-marketed as the treble and bass controls on consumer stereo receivers. They have seen a resurgence of interest from Mastering Engineers and professional audio manufacturers alike (i.e. the Dangerous Bax EQ) for a natural and familiar tonality. A *resonant shelf* is a unique shelving curve attributed to Michael Gerzon[3] that dips at the shelving frequency and never plateaus—making it excellent for adding high or low-frequency *extension*, especially in M/S to bring vitality to a lifeless vocal (see Chapter 15—Mid-Side). This type of EQ can be simulated with a wide band on a parametric set for a frequency below 20Hz or above 20kHz, resulting in the same effect, otherwise known as the left or right half of a wide bell curve. A *tilt EQ* will shift along a selected frequency 'fulcrum' and gradually boost or cut to either side of it in a very gradual manner—best applied for tonal shaping.

It is worth mentioning one of the holy grail EQs, the Pultec EQP-1A. The Pultec is a *passive EQ* (using passive electronic components—resistors, capacitors, and inductors—that are not powered until output stage amplification) designed and produced beginning in 1951 by Ollie Summerland and Gene Shank. They can be set for rich low-frequency enhancement and extended shimmer in the high frequencies. There are only two selectable frequency bands and a low-pass filter, but the ability to boost and attenuate simultaneously creates unique and coveted EQ curves that many manufacturers have since endeavored to duplicate. Pairing the EQP-1A with the MEQ5—a three mid-band EQ—allows for more flexibility. Figure 4.6 shows the vintage pair of EQP-1As from Capitol's Studio B.

Graphic EQs have a set series of frequencies at a fixed bandwidth that can be cut or boosted via sliders. These are not common in a mastering context, and are often used for room tuning or live sound (PA) reinforcement applications.

Fixed parametric EQs (example: NTI EQ3 [Figure 4.7] or Maag EQ4M with only HF 'air' shelf selectable) do not allow for the adjustment of bandwidths or frequencies—likely contributing to less expensive manufacture and quick implementation for the user. They have practical applications for mastering, but frequency limitations require using additional EQ options in a mastering system.

Figure 4.6 A pair of Pultec EQP-1A EQs from Pulse Techniques in Englewood, NJ. Pultecs are passive EQs—they don't require power in the EQ circuit and use passive components (resistors, capacitors, and inductors) followed by a tube gain makeup amp.

Source: (courtesy Capitol Studios)

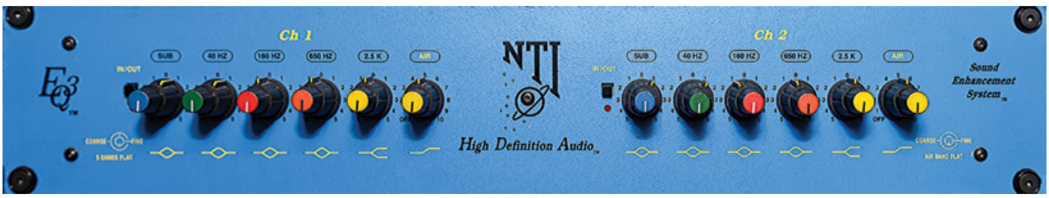

Figure 4.7 An NTI fixed parametric EQ. Stepped attenuators at 3dB clockwise steps, and ¼dB counter-clockwise steps for precise adjustments. 'Air band' is popular in stereo buss/mastering applications.

Source: (courtesy Capitol Studios)

For mastering applications, the *parametric equalizer* remains the primary EQ of choice due to its flexibility in selecting bandwidths (Q)[4] and frequencies. The design and concept is credited to Burgess Macneal and George Massenburg,[5] who designed EQs under the ITI,[6] Sontec, and then later, GML names. This tool allows for precision selection of the independent parameters (bandwidth, boost/cut, and frequency) that will accentuate or attenuate the chosen frequency (or frequencies). To further inform your mastering acumen, it is helpful to familiarize yourself with the following seminal parametric EQs: ITI ME-230 (Figure 4.8), Sontec MEP-250C and MES-432C (Figure 4.9); GML 8200 (Figure 4.10) and 9500; Pultec-style or Enhanced Pultec EQs—Pulse Techniques' EQM-1A3, EQM-1S3, and MEQ-5, Manley Labs' Enhanced

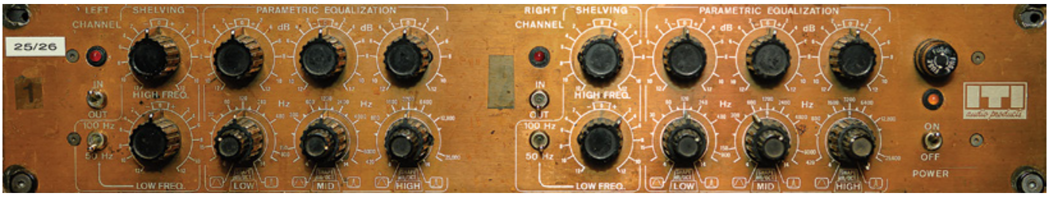

Figure 4.8 The first commercially available parametric EQ—the ITI ME-230 circa 1971.

Source: (courtesy Capitol Studios)

Figure 4.9 Burgess MacNeal started Sontec in 1975 from remnants of defunct ITI, and produced one of the quintessential mastering EQs. Pictured here is one of two Sontec MES-432Cs I use daily at Capitol.

Source: (courtesy Capitol Mastering)

Figure 4.10 George Massenburg's first post-ITI product, the GML 8200 (the GML 9500 is the mastering version with stepped attenuators in 0.5dB steps and precision resistors).

Source: (courtesy Capitol Studios)

Pultec and Mid EQ (Figure 4.11), and the Bettermaker Mastering EQ; Maselec MEA-2; Tube-Tech EQ 1AM; and Weiss EQ1 (Figure 4.12), an acclaimed digital hardware parametric. Notice these mastering-grade parametric EQs generally include the common functionality of a high-pass filter (HPF), low-pass filter (LPF), high shelf (HS), low shelf (LS) and three or four parametric EQ bands for both tonal shaping and surgical EQ approaches. Additionally, they are fitted with stepped attenuators with precision resistors[7] for accurate recall and reliable left/right stereo imaging. Any coloration differences

Figure 4.11 Manley's impressive take on producing mastering-specific Pultecs with stepped attenuators and precision resistors circa 1990–1993. These EQs impart a rich tonality, smooth high-frequency extension and offer the sought-after Pultec EQ curves.

Source: (author collection)

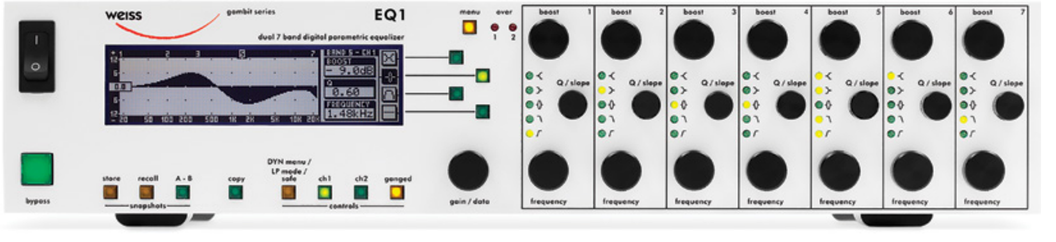

Figure 4.12 The Weiss EQ1 is a seven-band digital hardware parametric EQ. It comes in four configurations (basic, linear phase, dynamic and dynamic-linear phase) and includes M/S functionality.

Source: (courtesy Weiss)

between them are due to discrete components and circuitry design, bandwidth options, the number of EQ bands, frequency options, cut/boost increments, and range.

With the advent of plug-ins, the adjustable settings of a parametric EQ become virtually unlimited, permitting detailed adjustments not feasible in their analog counterparts. These user parameters may include options such as M/S for each band, parallel or series interaction between bands, and phase linear, minimal phase, and analog phase options. Excellent plug-in parametric EQs worth experimenting with are the DMG Equilibrium (Figure 4.13) and the FabFilter Pro-Q3 (Figure 4.14). This precision adjustment of parameters is true for all plug-in audio processing equipment including compressors and limiters—hence my assertion that the best modern mastering system incorporates both digital and analog domains (see Chapter 14—Advanced Mastering Tools).

COMPRESSORS/LIMITERS

These devices manage dynamic range and transient peaks by lowering (compressing) program level above a selected input threshold. They can also be used to add apparent volume or loudness by then adding output gain to the compressed audio. They are used in mastering to keep audio signal from distorting, add sonically pleasing coloration, increase the RMS level of program material (loudness), and create a hyped 'radio' sound. A standard *downward compressor* will compress or lower peaks above a selected input threshold.

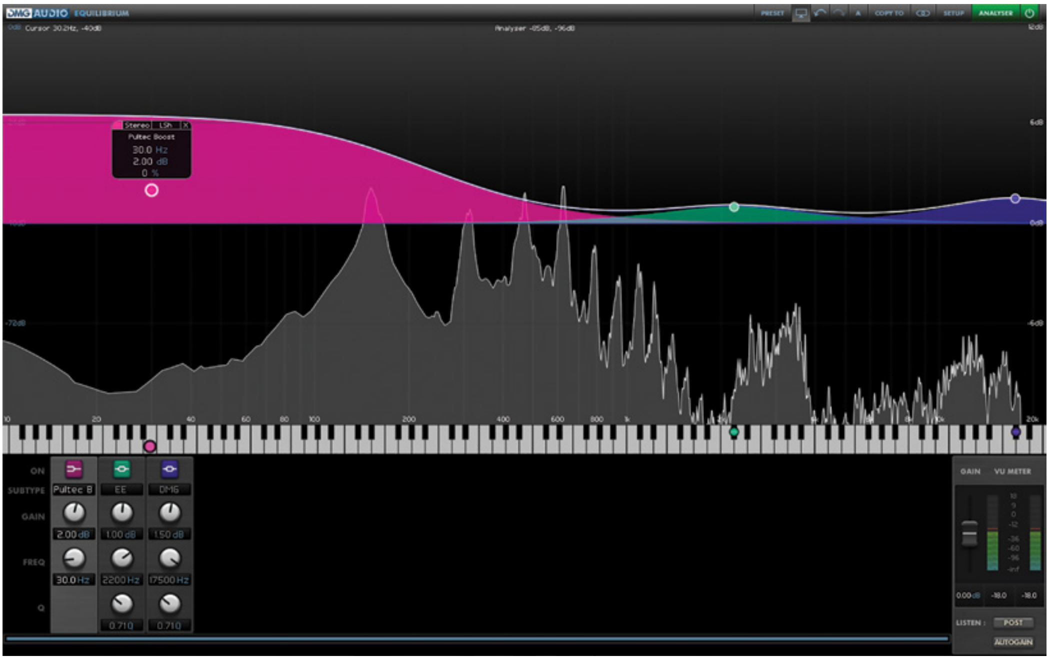

Figure 4.13 The DMG Equilibrium EQ plug-in is feature-rich with multiple filter, shape, phase control, configuration, analysis, and interface display options. It models a hall-of-fame selection of coveted vintage analog EQs (that can be operated linear phase!) including Pultec, SSL E and G series, Sony Oxford, Focusrite ISA 110, API 550, Neve 88, Harrison 32 C, Sontec 250, and GML 8200. Each band can be used in M/S mode. I use it regularly—*the* 'desert island' plug-in EQ.

Source: (courtesy DMG)

Figure 4.14 FabFilter Pro-Q also boasts a great user interface and impressive features such as phase linear operation, dynamic EQ, M/S processing, and spectrum analysis. The EQuilibrium doesn't have dynamic EQ, but the Pro-Q doesn't have vintage EQ modeling.

Source: (courtesy FabFilter)

However, if you use a compressor in parallel with the main program signal, the combination functions as an *upward compressor*, bringing the 'valleys' up. This is handy for adding detail, clarity, and apparent volume while still preserving transient response, thus maintaining vitality and punch (see Chapter 14—Advanced Mastering Chain Tools and Techniques). Perhaps the purest form of *upward compression* is to raise low sections of a recording with level automation or editing, leaving the transient peaks alone.

Compressors typically have adjustable controls for *input*, *threshold*, *ratio*, *attack time*, *release time*, and *output*. A brief description of each follows: *input* controls the input level, *threshold* determines the level after which compression will begin, *ratio* determines the amount of compression (i.e. a *ratio* of 4:1 means that for every 4dB of *input* level above the *threshold*, the unit will *output* merely 1dB), *attack* and *release times* refer to the speed of compression action and its subsequent release above the *threshold*, and *output* refers to the output gain (sometimes indicated as makeup gain) added after compression.

By contrast, a *limiter* stops or *limits* transient peaks in program material by using a much higher *ratio* than does a *compressor*. Also, the *attack* and *release* settings are set to more intensely respond to program above the *threshold* setting. For analog *compressors* with *limiting* functionality, such as the Manley Variable-Mu™ or Smart C2, this *limiting* involves selecting from the higher *ratio* settings and adjusting the *threshold* for action on loud peaks, usually with medium to slow *attack* times and fast *release* times. In figurative parlance, a *limiter* is the 'bigger and hairier' version of a *compressor*. A *limiter* can prevent downstream equipment from distorting; or if the engineer raises the overall gain feeding the *limiter*, the average *root mean square* (RMS) level increases, resulting in *louder* sounding masters. For example, the Smart C2 has an 'L' ratio setting for *limiting*, the SSL Stereo Compressor has a ratio setting of 10:1, and the Urei/Universal Audio 1176 has 20:1 as a *limiting* ratio setting. A related note on the 1176—most engineers know of the 'all-buttons' mode that is revered for its renowned 'pumpy' and overdriven sound. I will continue this chapter by primarily discussing hardware *compressor/limiters*, but most of these units have meticulously modeled plug-in counterparts for digital applications.

Varieties of Analog Compressors

Analog compressors are usually designed utilizing one of four different gain reduction-based circuits: electro-optical, field effect transistor (FET), variable-gain, and voltage-controlled amplifier (VCA). A fifth type, a diode-bridge compressor, is less prevalent, but is used in the Neve 2254 and 33609 compressors (see Figures 4.32 and 4.33). Bear in mind that the sound of compressors is also informed by elements beyond the gain-reduction circuit. Transformers or tubes in the circuit design and user-controlled features result in vast sonic differences, so consider all aspects when selecting or auditioning a mastering compressor.

Electro-optical compressors are smooth and musical, and produce warm coloration. They perform gain reduction by virtue of a photocell that reads the brightness of a bulb, light-emitting diode (LED) or electroluminescent panel which is correlated to input level. The classic electro-optical compressor/limiters from the 1960s and 1970s—the Universal Audio Teletronix LA-2A (Figure 4.16),[8] LA-3A (Figure 4.17) and LA-4A—*only* have an input and output adjustment so that *threshold*, *ratio*, *attack*, and *release* are internally set. The time constants for *attack* and *release* remained non-linear meaning program dependent response (fast then slow release, for example) and create a slow or 'spongy' response

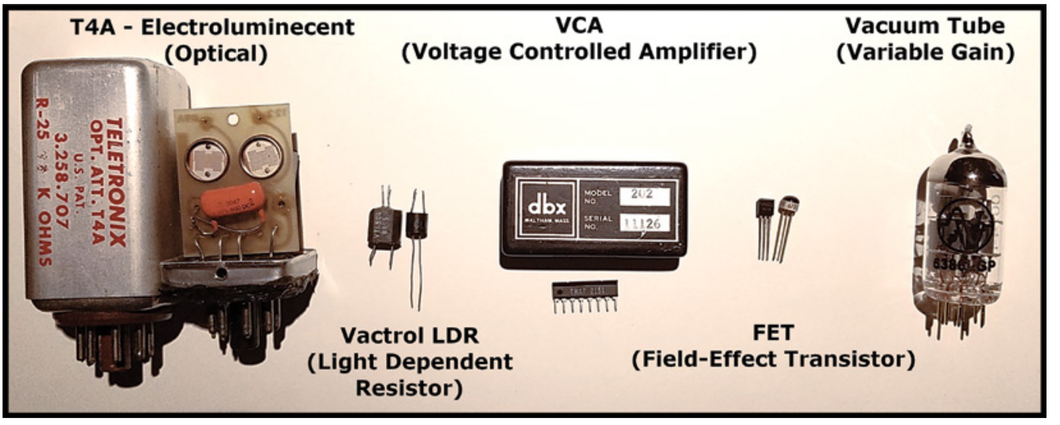

Figure 4.15 Gain reduction circuit component types, from left: the Teletronix LA-2A's renowned T4A optical attenuator with electroluminescent panel and photo resistors, a vactrol light dependent resistor (LDR), voltage-controlled amplifier (VCA), field effect transistor (FET), and vacuum tube.

Source: (courtesy Ian Sefchick)

Figure 4.16 Universal Audio Teletronix LA-2A electro-optical compressor. Originally designed by Jim Lawrence in the early 1950s. Revered for its musical multi-stage release time characteristics, it inspires many modern electro-optical mastering compressor designs.

Source: (courtesy Capitol Studios)

Figure 4.17 Universal Audio/Urie LA-3A electro-optical compressor pair. Uses the same T4 optical attenuator as the LA-2A, but with solid-state electronics (tubes were considered old technology by the late 1960s).

Source: (courtesy Capitol Studios)

characteristic. This makes these compressors well-suited for tonal enhancements and slight peak management. As the demand for stereo buss compression increased, manufacturers added features and functionality to the coveted classic designs to offer stereo electro-optical compressors made specifically for buss compression or mastering such as: the Manley SLAM!, PrismSound Maselec MLA-2, Tube-Tech CL2A, Pendulum OCL-2, and Shadow Hills Mastering Compressor (Figure 4.19).

FET compressors utilize a transistor to perform the gain reduction on the program material. They offer greater versatility and control on transient peaks than an electro-optical compressor since *threshold*, *ratio*, *attack*, and *release* are user-adjustable. Characteristic is a focused or intimate type of coloration increasing with extreme gain

Figure 4.18 A pair of Summit Audio TLA-100A compressors. These are often misconstrued as electro-optical compressors but actually use a proprietary VCA gain reduction circuit.

Source: (courtesy Capitol Studios)

Figure 4.19 The Shadow Hills Mastering Compressor. This unit boasts two compressors in series: an electro-optical compressor first, followed by a discrete VCA compressor. The output transformer is selectable between nickel, iron, and steel for different coloration options.

Source: (courtesy Shadow Hills)

reduction settings. Classic examples are the Urei 1176 (Figure 4.20), 1178, and 2–1176.[9] A FET compressor can be quickly set up to function as a limiter by increasing the *ratio* to the highest setting and then carefully adjusting the *threshold* (generally higher), *attack* and *release* settings for the desired peak reduction response. Modern units suitable for mastering are the Manley SLAM! (Figure 4.21), Crane Song STC-8 (Figure 4.22), and Overstayer 3706 SFE.

Variable-gain compressors utilize the re-biasing of a vacuum tube to handle the gain reduction duties. This topology does not have a user *ratio* control, as the *ratio* increases along with the input amplitude. These compressors are generally known for a warm or smooth coloration characteristic and provide tonality more than the quick control of peaks. Of course, each unit must be carefully listened to for coloration assessment—one of my variable-gain compressors at Capitol imparts a pleasing high-frequency extension along with the expected 'glued-together' component,

Figure 4.20 Universal Audio 1176 compressor pair. Released in 1968, the 1176 implements a FET gain reduction circuit and is known for fast attack and release time settings, and range of tonal characteristics from clean compression at the 4:1 ratio to compelling saturation in 'all-buttons-in' mode.

Source: (courtesy Capitol Studios)

Figure 4.21 Manley SLAM! This is a unique and extremely useful stereo mastering compressor that features both an electro-optical compressor (à la the Teletronix LA-2A) and a FET compressor (à la the Urie 1176). These can be accessed in series, or independently as required. Each compressor has a number of mastering-centric modes. Early models had a digital I/O option that featured mastering quality Anagram Quantum ADDA converters (discontinued in 2009). The output gain has a very even un-hyped sound. It is a clever mastering-centric box and I use it daily.

Source: (courtesy Capitol Mastering)

Figure 4.22 The Crane Song STC-8 is a discrete Class A FET compressor combined with a peak-limiter and an enhancement circuit for introducing analog warmth.

Source: (courtesy Crane Song)

Figure 4.23 The famed Fairchild 670 variable-gain compressor was developed in the early 1950s by Rein Narma for The Fairchild Recording Equipment Corporation. It was used on many Beatles recordings by Sir George Martin, and was a mainstay in vinyl cutting rooms for decades.

Source: (courtesy Capitol Studios)

whereas other versions of the same compressor do not. Attack times are quicker than an electro-optical compressor, but still not at the peak-managing functionality of a FET or VCA compressor.

The most revered and iconic variable-gain compressor is the Fairchild 660 (mono) and 670 (stereo) (Figure 4.23) from the 1950s. The Fairchild has *input*, *threshold*, six different *time constants* (as attack/release times),[10] and *output* available for user adjustment. They have a hallowed status in the pantheon of compressors due in large part to one Sir George Martin, who favored them on Beatles recordings.[11] They utilized military-quality components and had vertical/lateral (M/S) functionality for lacquer disc cutting applications, so they were often installed in mastering studios before the advent of digital audio and the compact disc. They gained favor among newer companies seeking to create a reliable modern version, with the Manley Variable-Mu™ compressor (Figure 4.25) becoming a mastering studio staple. Other excellent modern examples of this topology are the Undertone Audio Unfairchild (Figure 4.24), Pendulum 6386/ES-8, Magic Death Eye Mastering Compressor (Figure 4.26), Thermionic Culture Phoenix, and Capitol Mastering CM5511 (Figure 4.27).[12]

VCA compressors utilize a voltage-controlled amplifier whose control voltage is derived from the audio input signal itself to effect the gain reduction. Classic examples are the dbx 160 (and permutations) (Figure 4.28) and dbx 165. These versatile compressor designs offer a great degree of user control for attack/release settings, making them excellent for stereo bus and mastering applications. Common VCA compressors suitable for mastering are the Alan Smart C2 (Figure 4.30), Neve 33609 (Figure 4.33), SSL G-Series (Figure 4.29), API 2500 (Figure 4.31), Overstayer 3722 SVC, and Vertigo VSC-3.

Figure 4.24 The UnFairchild is a faithful modern-day recreation by UnderTone Audio (UTA).

Source: (courtesy UTA)

Figure 4.25 The Manley Variable-Mu™ Stereo Compressor/Limiter is a mastering studio staple that draws on the design of the Fairchild 670.

Source: (courtesy Capitol Mastering)

Figure 4.26 The Magic Death Eye Stereo Compressor (designed and hand-built by Ian Sefchick) is a variable-gain compressor that boasts impressive details, such as a special EQ feature in the gain reduction circuit and hand wound transformers.

Source: (courtesy Magic Death Eye)

Figure 4.27 The Capitol CM5511 Stereo Compressor. Only four of these variable-gain compressors were built in 2011 by Ian Sefchick, and are used regularly at Capitol Mastering.

Source: (courtesy Capitol Mastering)

Figure 4.28 A pair of dbx 160 compressors (designed by David Blackmer in the early 1970s) that use a VCA gain reduction circuit.

Source: (courtesy Capitol Studios)

Figure 4.29 The SSL G-Series Stereo Compressor is a famed buss compressor, and was also onboard SSL consoles of the era.

Source: (courtesy Capitol Studios)

Figure 4.30 After working with SSL and designing the G-Series compressor pictured in Figure 4.29, Alan Smart released the C2 Stereo Compressor. This one is from my mastering studio at Capitol, and imparts a variety of useful tonal characteristics, depending on the setting. 'Crush' mode yields over-compression with a mid-range boost, and higher distortion.

Source: (courtesy Capitol Mastering)

Figure 4.31 The API 2500 Stereo Compressor is a mainstay of bus compression.

Source: (courtesy Capitol Studios)

Figure 4.32 The Neve 2254/E (released in 1969) uses a diode-bridge topology for its gain reduction circuit. They were included onboard Neve mix and mastering consoles of the era, and later often removed and rack-mounted with a power supply—as pictured here.

Source: (courtesy Capitol Studios)

Figure 4.33 The Neve 33609 evolved from the 2254 and was released in 1985. It remains a favorite stereo buss compressor among mix and Mastering Engineers. It also uses a diode-bridge topology for gain reduction. This is an early incarnation with no Neve logo or front-plate power switch.

Source: (courtesy Capitol Studios)

EXPANDERS/GATES

Expanders increase dynamic range in audio by either lowering level below the *threshold* (*downward expansion*) or increasing the level above the *threshold* (*upward expansion*). *Downward expanders* are more common and function like an 'inverted' compressor by lowering the 'valleys' in the program material. *Expanders* are used far less often than *compressors* in mastering, but can be helpful in working with an over-compressed mix. The extreme version of an *expander* is a *gate*. With a *gate*, very little or no audio is heard when the audio signal passes below the threshold. I mention *gates* here for their relationship to *expanders*, but please note they have no practical application in a mastering context.

BRICKWALL LIMITER (BWL)

A *brickwall limiter* (BWL) is a digital look-ahead peak-limiter that allows for a dBFS setting beyond which no audio signal will pass. Look-ahead refers to the main signal being delayed and the side-chain analyzed so that the limiter can process program peaks with ultra-fast *attack/release* times and a *ratio* of infinity:1. The result is extreme level control with no output signal above a user-set ceiling. The settings are usually *threshold* (with auto-gain makeup in some designs) or *input gain*, *digital output ceiling*, and *release* or *time constant* settings. A BWL is placed in *zone 3* just after the AD converter, and prevents over-levels at the RDAW, allowing for unhindered loudness maximization. The first readily available BWL was the Waves L1 software in the mid-1990s, then in 2000, the Waves L2 hardware (Figure 4.34) was released, quickly becoming ubiquitous in mastering studios. Other early hardware BWL examples are the t.c. electronic M6000 (Figure 4.35) and jünger loudness control devices.

Informed by the intense limiting of radio broadcast compressors, the BWL proved transformative to the mastering world, ushering in a new era of the *loudness wars*. Responsibly and carefully implemented, a BWL remains a valuable and effective tool for a Mastering Engineer to achieve reasonable *target levels*. However, over peak-limiting has very undesirable artifacts including 'hashiness,' inter-sample peak modulation, inverted dynamics (loud sections shrink, and quiet sections overtake), and an uncomfortable *listening experience*. The BWL democratized a critical component of mastering—loudness—but many mix and Mastering Engineers began submitting peak-limited mixes that were afflicted with the artifacts described previously. Indeed, initiatives such as Apple's *Mastered for iTunes* and streaming platforms such as Spotify that post level specifications are a direct response the pitfalls of overly peak-limited music.

Figure 4.34 The Waves L2 (hardware version) is a digital look-ahead peak-limiter released circa 2000 that evolved from the Waves L1 plug-in. It utilizes 48bit fixed-point DSP and allows for a dBFS ceiling to be set, over which no signal will pass. The L2 was widely embraced and ubiquitous in mastering studios (along with the hardware t.c. electronic M6000 Mastering Processor) and demarcated a significant point in time in loudness enhancement, management, and the fabled loudness wars.

Source: (courtesy Capitol Mastering)

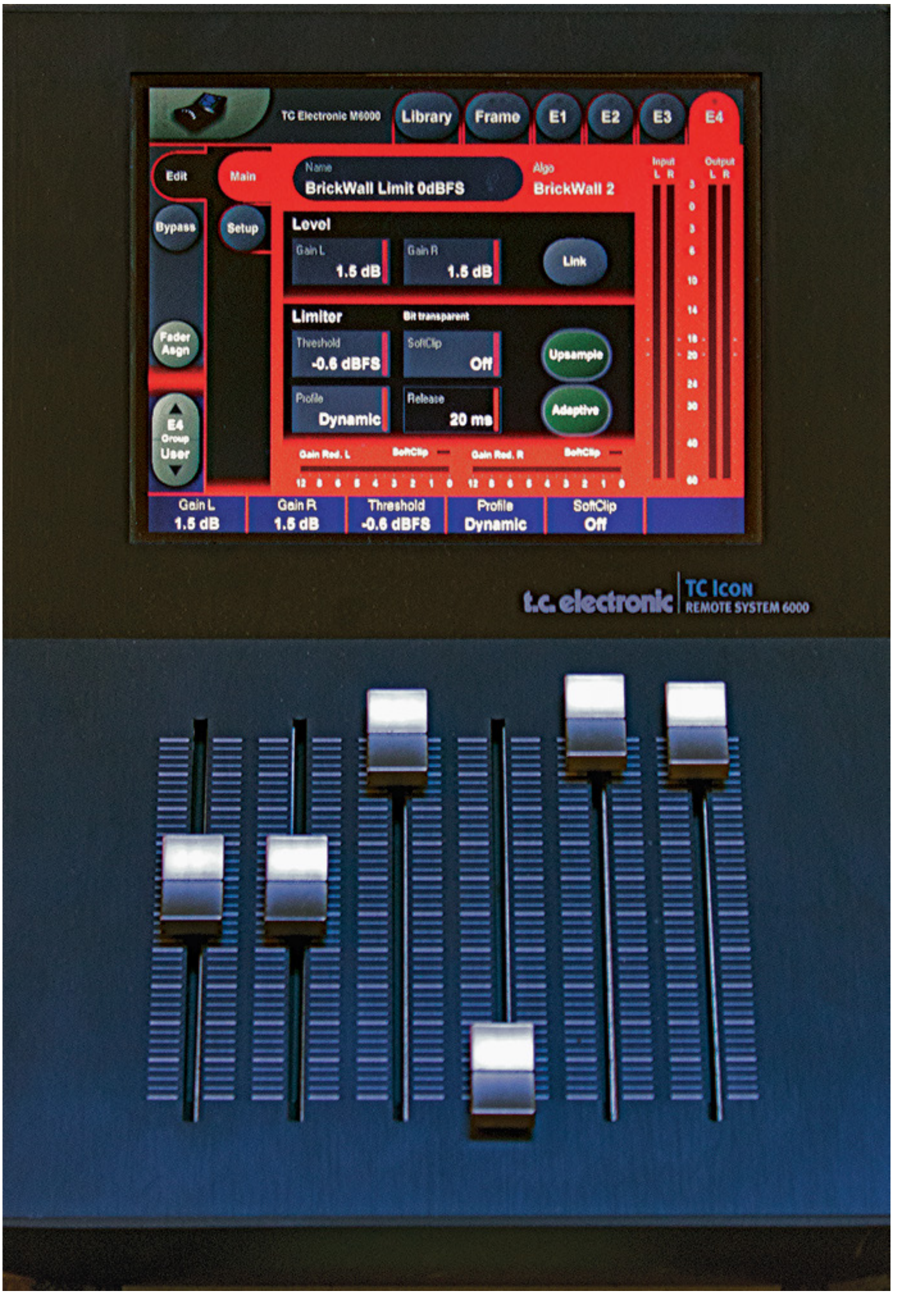

Figure 4.35 The t.c. electronic M6000 Icon Remote, which controls the mainframe M6000 Mastering Processor. The M6000 implements 48bit fixed-point DSP, and in addition to quality peak-limiting, offers vast processing options including EQ, multiband compression, and expansion in both stereo and Mid-Side.

Source: (author collection)

BWL Safety—User Tips

A little peak-limiting goes a long way, and 1–2dB is plenty. One approach I regularly use is to begin with the BWL in bypass, and go for a good genre-appropriate sound and gain structure using the PBDAW and analog chain (*zone 1* and *zone 2*), carefully avoiding over-levels at the RDAW. At this point, you should have a VU level of about +8dBu. Now engage the BWL and make your way to around +10dBu, which may be enough; otherwise, distribute additional gain at appropriate points in the mastering chain (before the AD converter) for a *target level* of around +12dBu. If the BWL is a plug-in, you will capture the relatively dynamic setting of +8dBu at your RDAW, then add a plug-in peak-limiter in the RDAW (*zone 3*). Using a BWL effectively requires judgment, care, and vigilance—the described process may require subtle *macro-dynamic* adjustments at the PBAW to preserve song dynamics. Note that with kick drum/bass-heavy genres such as rap or dance music, the kick will naturally swing beyond the VU meter levels indicated (Figures 4.36–4.38).

Figure 4.36 The Voxengo Elephant is a brickwall limiter plug-in which offers a wide variety of coloration options and customizable parameters.

Source: (courtesy Voxengo)

Figure 4.37 The FabFilter Pro-L2 is another feature-rich brickwall limiter plug-in worth auditioning and implementing for mastering. It includes extensive loudness metering.

Source: (courtesy FabFilter)

Figure 4.38 The DMG Limitless brickwall limiter plug-in implements multiband dual-stage processing that separates dynamics and transients, and generates extremely smooth gain reduction curves.

Source: (courtesy DMG)

DIGITAL AUDIO WORKSTATIONS (DAWS)

A DAW is the computer-hosted software that allows for the recording, editing, processing, and delivery of the final audio master. For professional mastering, I recommend a two-DAW setup for the following reasons: the playback DAW (PBDAW) remains dedicated to playback, preliminary processing, and source level adjustments; the record DAW (RDAW) only captures the processed audio and renders/generates all required master formats; it allows for different playback and record sampling frequencies/bit depths; and it keeps your files organized as flat and mastered on separate drives. My PBDAW is Avid Pro Tools on a Macintosh, and my RDAW is Steinberg WaveLab on a PC. Some Mastering Engineers are devoted to Macintosh computers, others to PCs, or a combination of the two. I have also experimented with a two-PC system, and for that I recommend the excellent PC Audio Labs Rok Box custom PCs configured for audio work. The most common mastering-specific DAWs are Steinberg WaveLab, SADiE, Sonic Soundblade, Magix Sequoia, and Pyramix.

I realize many engineers use a single-DAW setup, especially those newly developing their mastering approaches. In a single-DAW setup, however, usually both playback and capture functions occur in Pro Tools, which is recording/mix rather than mastering-dedicated software. Additionally, playback and record sampling frequencies must be the same, often requiring *sample rate conversion* (SRC) of the source audio to meet any high-resolution requests. Also, final masters must be assembled in other software that can create a DDP or PMCD Master. I acknowledge it as a viable option, but don't prefer it other than for an in-the-box mastering approach (see Chapter 16—In-the-Box Considerations).

CONCLUSION

Proficiency with the fundamental tools described in this chapter provides a solid foundation of skills for a Mastering Engineer. Well-modeled plug-ins represent a simulacrum of the coveted and expensive hardware equipment described in this chapter at a fraction of the cost, and often with expanded functionality. Despite this, desirable aspects of analog processing are hard to replicate in the computer. Conduct your own research and testing regarding the usefulness and fidelity of ADs, DAs, EQs, compressors, and BWLs. Perform A/B listening tests by changing just one piece in the mastering chain and carefully compare. This will help develop your opinion about what sounds better and why. Anyone can buy audio equipment, but informed and accurate knowledge of the tools help define a great Mastering Engineer.

EXERCISES

1. What three devices make up The Primary Colors of Mastering? What specific function does each device provide the Mastering Engineer?
2. If an all-*analog* signal path is selected, what two additional devices are required in the mastering chain (assuming a digital source file)?
3. Define each measurement represented in the relationship equation: −14dBFS = 1.23V = 0VU = +4dBu. Which measurements are digital, and which are analog?

NOTES

1. Analog audio is defined by a signal or wave; digital audio approximates analog via quantized samples at a fixed sampling frequency.
2. Pulse code modulation (PCM) or Delta Sigma binary codes.
3. Michael Anthony Gerzon (1945–1996) is probably best known for his work on Ambisonics and for his work on digital audio. He proposed the resonant shelf EQ shape described—rising shelf with a parametric dip at the shelving frequency.
4. Bandwidth is represented by the letter 'Q' as a single number ratio. The formula is Q = center frequency/bandwidth (cf/bw). It can also be represented in dB/octave.
5. George Massenburg and Burgess Macneal authored a technical paper entitled "Parametric Equalization" which was presented at the 42nd convention of the Audio Engineering Society in 1972. Macneal would design and manufacture the seminal Sontec MEP-250C parametric EQ in 1975. In 1982, Massenburg founded George Massenburg Labs—among GML's most venerable products are the GML8200 Parametric Equalizer and the GML8900 Dynamic Range Controller, which reacts to loudness like our ears do, rather than to voltage levels.
6. Designed by Burgess Macneal and George Massenburg, the ITI ME-230 (Mastering Equalizer 230) was the first commercially available parametric EQ.
7. Utilizing an accurate stepped attenuator controlled by a rotary switch (i.e. Grayhill, Inc.) and a separate precision resistor value for each position of boost/cut, frequency, and bandwidth.
8. Lynn Fuston (2013) *A History of the Teletronix LA-2A Leveling Amplifier*. See www.uaudio.com/blog/la-2a-analog-obsession/
9. Lynn Fuston (2000) *UA'S Classic 1176 Compressor—A History*. See www.uaudio.com/blog/analog-obsession-1176-history/
10. Hannes Bieger (May 2016) *The Fairchild 660 & 670—Sound on Sound*. SOS Publications.

    All positions offer extremely fast attack values, between 0.2 and 0.8 milliseconds. This was considered to be lightning fast at the time, and the figures weren't beaten until a decade later, when solid-state units with FET gain cells (like the Universal Audio 1176LN) brought attack values down to a mere 20 microseconds. Given its primary purpose as a protective device in broadcast or vinyl cutting environments, in which the limiter was required to catch any unwanted signal peaks reliably, these fast attack times were one of the Fairchild's most important features.

11. Hannes Bieger (May 2016).

    Since the "A Hard Day's Night" sessions in 1964, almost all the Beatles' vocals were sent through the Fairchilds, and the American limiters also played a huge role in shaping the sound of Ringo's drums, the guitars, and many more sources. The Beatles pedigree greatly helped to establish the Fairchild as a classic choice in music-production facilities all over the world.

12. Only four of these were made in 2011, one for each mastering studio at Capitol, by then technician Ian Sefchick.

CHAPTER 36

Mastering

Great music poorly recorded will always be great; crappy music wonderfully recorded will always be crap.

Mark Rubel, 'Rubel's Law of Production'

For mastering, I asked Jesse Skeens to contribute this chapter. He runs Medway Studios, a mastering facility in London (http://www.medwaystudios.com) and has been producing electronic music for almost 30 years. His production work includes releases as Medway, appearing on labels such as Armada, Hooj, Universal, Sony, and more, and he has also mastered a wide range of artists such as Oliver Smith, Spooky, and Wideboys. He says: 'This book, way back in the first edition in 2003, was helpful to me during my early days of making music. I hope this section helps you in your pursuit of mastering your music.'

WHAT IS MASTERING?

Mastering is the final step before distribution. Mastering began as a process to transfer recordings from tape to vinyl, the consumer delivery format of the day. It is traditionally known as a process of sequencing and balancing songs across an album. With changes in the industry favoring singles, it now also consists of adjusting a track's frequency and dynamics. By doing so, a record should sound consistent across different playback systems from portable devices through to large club systems.

Over the past ten years mastering focussed on achieving the loudest level possible. This became known as the 'loudness war' where the levels of commercial music increased year on year resulting in the degradation of audio quality. Today, thankfully, this war has mostly subsided, but it can still play an important role in mastering.

Creative mastering improves the impact and listening enjoyment of a piece of music. The amount of difference that a mastering engineer imparts in this regard can vary from little to becoming integral to the overall sound. Depending on the wishes of the artist, of course.

While mastering can be used to fix errors and issues within the mix, it is preferable that these are addressed in the mix so that the mastering engineer can concentrate on the bigger picture. It's here that the experience of a good mastering engineer will become evident, as they will ensure a song is at its best before being presented to the public.

DIY VS. PRO

Mastering, on the pro level, has a well-earned reputation for requiring equipment that is beyond the budget of the average home producer. Alongside hardware equipment such as EQ and compressors, this also extends to the monitor speakers used. In addition to these, however, you must also consider the experience of an established mastering engineer. They have worked on many types of track, each requiring a slightly different set of adjustments and this culminates in a wealth of knowledge at the engineer's disposal.

A mastering engineer who has worked for ten years will have upwards of 20,000 hours' experience. This includes an in-depth understanding of the select hardware they use, that permits them to extract and enhance subtle details with great finesse. A second pair of ears is also useful, especially when one has poured hours into a project … by which point objectivity can be hard to come by.

Naturally, you may wish to try mastering at home to save costs, and there are benefits to learning to master in your studio. Understanding how processors interact on the master buss and how they relate to your mixing can be rewarding. You also have the opportunity to experiment with your music without having the expense of paying a mastering engineer each time.

What follows is a technical overview of the process involved in mastering followed by a 'real-world' example putting this into practice.

GETTING IT RIGHT IN THE MIX

You should always begin with the best mix possible. In a DIY situation, you have the luxury to go back and forth between the mix and master to make adjustments if required. So, before approaching a master here are a few things to pay attention to.

The closer you can get the EQ to being balanced the better. Ideally, mastering will entail only slight broad boosts or cuts and perhaps some surgical application to remove resonance. Resonances are small peaks that stick out in the mix and obstruct adjacent sounds. This leads to a cluttered mix that can sound piercing at high volumes.

If a particular sound is overly bright or resonant, this would need to be corrected with EQ (either static or dynamic), but this action may affect other sounds that may then appear dull or muted.

Compression is a secondary mix process that's easy to overemphasize. This is especially applicable when artists have attempted to mimic what they hear on mastered commercial songs. Once limiting is applied to achieve a reasonable volume level, some dynamics are lost so you must be cautious with any dynamic processors. However, if you do not compress enough or don't balance levels correctly, this may result in more compression required during mastering. This again, like incorrect EQ, can exhibit adverse side effects.

Effects such as delay and reverb may also become pronounced during mastering. The difference is often minimal but is something to consider, mainly if the mix employs these effects excessively. Finally, it is essential to remove any hiss, crackles, or other audible artifacts as these may become more evident while mastering. Subtle imperfections such as uneven frequency balances or wayward dynamics are okay to leave until mastering.

SOFTWARE

What software is required for mastering your music? In many cases, the same DAW you produced on will suffice. If budget allows, dedicated programs such as Steinberg's Wavelab include extra features such as integrated metering and DDP/CD authoring.

Although the full Wavelab package might be out of reach for many, the Elements version still has many of the required features to put the final polish on your premaster.

Many DAWs will include some basic form of metering, a good alternative that's free is the Voxengo SPAN. It provides excellent frequency analysis along with stereo correlation and level metering.

As an all-in-one package, iZotope's Ozone gives a full suite mastering tools including compression, limiting, saturation, imaging, EQ metering, and more.

MetricAB, the successor to MagicAB, now includes a full complement of analyzers along with the ability to check against commercial references, quickly and easily.

FIGURE 36.0
ADPTR MetricAB

THE MECHANICS OF MASTERING

The first step in mastering is to decide on the tools you'll use. While many professional mastering engineers will employ analog hardware for EQ, compression, and tonal coloration, mastering in your studio can be accomplished with various third-party plug-in software.

If you are on a budget, digital will offer a far higher level of quality than comparably priced hardware counterparts. Although there are some affordable pieces of hardware on the market, most of the products suitable for mastering far exceed the cost of digital software tools.

Instead, any additional finances should be put towards monitors and the monitoring environment as these will have a significant impact on mastering, where excellent levels of detail are required.

Part of the challenge in mastering is that often correcting one problem will adversely influence another. Going into any mastering project, you must be conscious of avoiding damaging the program material, but this is not always as easy as it sounds. As most mastering work occurs on a stereo file, care must be taken not to remove more than you enhance. This can be mitigated somewhat with the use of stems, more on this later.

In a mastering situation, less is invariably more, stacking up plug-ins or techniques you've recently learned is a sure fire way to overdo it. The most significant offender for this is compression.

COMPRESSION

Compression has been discussed in earlier chapters, so I won't reiterate what has already been discussed, but the difference lies in the application.

During mastering, you should avoid heavy-handed application with a stereo single band compressor. At this point, compression should be used as a subtle glue and gentle shaping of dynamics to provide a final level of polish and detail so that it 'sounds like a record.'

The inclusion of a side-chain filter is essential for getting the most out of a compressor in dance music applications, particularly during mastering. A side-chain filter will remove some of the low-end energy of the internal key signal so that it doesn't react to this region. The signal that passes audibly through the unit remains unaltered. This means the compressor doesn't 'see' the low end, so it only compresses using the rest of the signal. If a strong bassline or kick is present in the mix, this will avoid the compressor continually being triggered by those types of element.

The amount of compression to apply will vary greatly depending on how it was implemented during the mix stage. Everyone has their preference for how they use compression during the mixing process, so this is up to you. A typical situation is where the mix-down compression applied 2–3 dB of gain reduction to gel the mix and then while mastering a slight amount, say 0.5–1 dB is used as a final polish.

A proper technique is to stack multiple compressors each compressing by small amounts. A favorite plug-in compressor, the UAD Shadow Hills, combines two separate processes into one unit for this purpose. Here, there is an optical and a VCA stage. The two exhibit different responses: the opto (optical compressor) is considered smooth while the VCA is deemed to be punchy. These two separate processes permit you to dial in different dynamic responses and tonal coloration that wouldn't be possible with a standard compressor.

In my use of compression, the difference between compressed and an unmolested signal will always be subtle. I aim to add a delicate touch of warmth and glue, but not change the dynamics. Here, running a mix through a compressor such as the API-2500 – without any modification to the dynamics of the signal – can add a smooth tonal character.

Any more than 1 dB of gain reduction on the stereo bus should be handled in the mix stage so you can adjust levels as the music interacts with the compressor. Typical settings for master bus compression involve low ratios such as 1.15:1 up to 2:1. Depending on the unit, the attack is set to one of the slower settings, but in some cases, faster settings may work. For the release time, it should be set to the groove of the music. This provides a tightening effect that's musically in time with the rhythm. A rule of thumb is to set the compressor so that it snaps back before the next kick hits. The timing of this varies depending on each compressor but somewhere between 150–500ms is a good place to start. A release that's too fast can result in distortion as it tries to 'ride' the individual cycles of a low bass waveform. Set too slow, the release can make the mix pump and suck out all of its energy.

One of the most infamous mix bus compressors, the SSL (in its multiple incarnations) features an 'auto' release mode. Here, the release will slow down when longer periods of compression occur and will react faster to short peaks. This can prevent the mix from pumping and is a good place to start if you're not sure what release to use.

Notable compressors for mastering include:

- API-2500. This can be daunting to use at first because it offers a multitude of options. These include as feedback, feedforward, knee, and a patented 'thrust'' control (essentially a filtered side-chain). The API is revered in mastering circles for the thick warm sound it imparts, even when no compression is taking place. This sound can be attributed to its op-amps and output transformers.

FIGURE 36.1
API-2500

FIGURE 36.2
UAD Shadow Hills' mastering compressor

- SSL bus compressor. This compressor is probably the most famous glue compressor ever made and is considered by many engineers to make most mixes 'sound like a record.' The operation is straightforward, so it's easy to get a great sound from one. For dance music it must feature a side-chain to avoid pumping; otherwise, it may cause the stereo field to collapse, making the mix appear more narrow, and a loss of dynamics.
- Cytomics' the Glue is an excellent choice for SSL style compression if you want to use a plug-in.
- If you own a UAD DSP card, the Shadow Hills is an all in one package with two compressor models and three settings to impart transformer coloration to glue a mix further.

You can even experiment with parallel compression during the mix stage. Used sparingly (a trend that will continue throughout this chapter), parallel compression during mastering can reap benefits as well. There are two distinct applications for this process: to add density or to add punch. Which to choose depends on the premaster.

A song that lacks punch and dynamics can be rescued with parallel compression. Employing a slow attack and quick release set to the groove of the music will accentuate the attack portion of the material. Mixed in low underneath the dry signal this can add some extra transient information. If a song is lacking density, try a fast attack and quick release. This will increase the sustain and body of the material, which will thicken up the master. Experiment with ratios, but typically lower ones works better.

MULTIBAND COMPRESSION

An alternative to single band compression is multiband. A multiband compressor divides the signal up to four or more frequency bands so that each one may operate independently.

This enables you to focus on specific frequency areas of the mix without affecting others. This permits you to tame the low end of a kick drum and bass line without changing the sound of the midrange and high-frequency instruments. It can also add density and glue to specific frequency bands while leaving others to breathe.

However, while this is an incredible tool for specific applications it also has the potential to mess up a mix severely if not applied with caution. A multiband can upset the tonality of a mix and cause certain sounds to become disjointed. For instance, if the lower mid and upper mid bands are not set evenly, a sound that spans across both of them can become unbalanced.

iZotope produces what is perhaps the most commonly employed multiband software compressor within their Ozone suite of mastering plug-ins. It features plenty of control and provides a solid punchy sound. Alternatively, the Fab Filter Multiband makes an excellent choice too.

A typical application for multiband compression would be to tighten the very low end of a mix so that the kick and bass become consistent. This can be accomplished by soloing the low-frequency compression band and adjusting the crossover to the required frequency. If you want to even out the subs, then choosing a low frequency such as 80 Hz can work. The action of removing a few peaks with a ratio of 10:1 can maintain a stable low end, but be careful not to flatten out the impact entirely. A couple of dB is all that is required; any more and you should return to the mix. As with standard compression, parallel can work here too to add density to some regions of the audio without completely changing the dynamics.

It's important to have a clear idea of what you expect to achieve when applying multiband compression. During a mix, a novice may be tempted to employ a preset because it provides an immediate boost to the overall mix level. The problem with this approach is that it can negatively influence how you mix. If the preset is adding lots of gain to specific bands, you might add enough EQ on the track level. Likewise, if certain bands are compressed too much, they might encourage you to push mix levels up higher than they need to be to compensate. Later on, if the multiband is taken off to perform mastering, the mix falls apart as it relied too heavily on these settings to sound somewhat balanced.

Until you become comfortable with multiband compression, it's best to apply it sparingly during mastering.

MASTERING LIMITERS

Limiting has become synonymous with producing a competitively loud mix so that it stands up in volume against the competition. While increasing levels of a commercial standard is a high priority for mastering, it's less of a competition than it used to be the loudest due to changes in streaming services such as Spotify and YouTube.

The quest for loudness in commercial music dates back many decades, but it hit its stride in the late 1990s. Louder is invariably better to a listener, so engineers would push limiters to the max to produce a loud master. This became intertwined with the sound of the music itself with genres such as dubstep pushing levels to their limits as an artistic choice.

Thankfully, for the 21st century, we seem to be heading towards a positive trend to produce masters at a more conservative level. This is due to changes in online services that now auto-level music to a specific LuFS output.

LuFS is short for **L**oudness **u**nits relative to **F**ull **S**cale. It's based on the k-weighting system and offers a more accurate way to represent the perception of sound levels compared to RMS.

At the time of writing, the standard level for YouTube and Spotify is −14LuFS while iTunes sits a little lower at −16LuFS. How does auto-leveling work?

For decades FM radio stations used special limiters to control the dynamics of music to both protect their transmitters but also even out the level differences from one song to the next. These streaming platforms apply a form of *auto-leveling* that adjusts the music volume to their standard LuFS measurement. For example, if you were to release a track mastered at −12LuFS onto YouTube, their automated software will reduce the level by −2 dB to bring it in line with their standard broadcast level.

There is a useful feature on YouTube that will show you this happening. Right-click any video and choose 'Stats for nerds' menu item. Here you will see a section named 'Volume – Normalization.' This shows whether YouTube raised or lowered the volume of the audio.

This doesn't necessarily mean the loudness war is over, however. Not everyone has adapted her mastering practices to suit the new standards of streaming services, and the mastered volume is often dependent on the genre and the target for consumption. DJs playing tracks in a club are not affected by these changes, so many home producers still produce loud masters to compete with other mixes, despite the fact that a DJ has access to a volume control.

A heavily limited mix will cause some problems when played back on a club system. Without small breaks in the level, speakers never have time to reset resulting in a loss of punch and impact. Most clubs will also have some protection in the form of limiters (similar to an FM station), but overcompressed mixes will trigger these processors more heavily with another loss of dynamics.

The good news is that if you don't want to remove all the dynamics from your music, many streaming platforms are in your corner. Have a listen to producers in your genre and try to find a middle ground of not too quiet but not too heavily limited either. This should allow your music to be appreciated without having to compromise on sonic integrity.

Even with these standards in place, you'll probably still want to push your music up to an acceptable level. There are many ways to achieve loudness, but a limiter is undoubtedly the most straightforward approach.

Limiting can be described as compression with a high ratio of 10:1 or more with a fast attack and release. Today's software limiters, however, often operate with much higher ratios, up to 1000:1.

Before digital limiting became commonplace, many engineers would hit the inputs of their A/D converters to clip on purpose. By shaving off small peaks, they were able to produce a louder level without audibly affecting the mix.

Waves' L1 was one of the first plug-in limiters to hit the market over 25 years ago, but today there is a wide assortment to choose from.

As the loudness war raged, software developers began to devise more complex techniques to deliver loudness with minimal sonic artifacts.

Paradoxically some of the latest generations of limiters returned to the more straightforward approach of clipping but with control over wave-shaping and other processes to mimic the technique some engineers had employed by clipping their hardware A/D converters.

There is plenty of talk on forums about the best limiter, but it doesn't exist. Every limiter will impart a particular characteristic on to the program material. It's up to you to choose which complements your music while providing the amount of level you require.

The character imparted by any limiter will be proportional to the amount of processing employed. At just a couple of dB at most, it will remain neutral. However, a limiters attack and release times will also influence the sound. As a general rule, faster attack times will capture the transients and prevent clipping, but this sudden action will reduce the dynamics of the transient. Similarly, a quicker release time will produce a louder result but may introduce distortion as the limiter tracks and reshapes the lower frequencies. The key to successful, transparent limiting it to adjust both carefully, so there are minimal audible artifacts.

While it's helpful to have a handful of limiters available to test against program material you are mastering, not everyone can afford all of the latest limiters. In this case, demoing a few and picking what you have the budget for is the best course of action.

At the time of writing, there are a few limiters that prove popular:

- Fabfilter's Pro-L is one of the leading limiters and offers plenty of control with four modes optimized for particular material. Characteristics of this limiter include a smooth and warm sound that is perfect for material that might be edgy. Downsides are a lack of punch and clarity when pushed hard.
- iZotope's Ozone mastering suite contains a collection of mastering tools, including a limiter. Ozone has been available for almost 20 years,

and the limiter includes some submodes. These allow the limiter to react differently depending on where a punchier or smoother outcome is desired. It's an excellent complement to Fabfilter because it's sonically tighter and crisper, but it can become sharp and edgy if pushed hard.

- Sonnox's Limiter is also a mainstay of many engineers. It's a little tricky to set up at first due to some nonstandard controls, but with practice, it produces excellent results. It includes an enhance feature taken from its Inflator standalone plug-in. Adding in this process gives more density to the material without limiting it. This is an excellent way to achieve additional loudness without excessive limiting.
- Tonebooster's Barricade is perfect for those on a tight budget. It's capable of smooth limiting that keeps lower frequencies intact and rarely produces harsh distortion. The rounded nature of its action may not be the best fit if you want a crisp sound, but for the price, it cannot be beaten.
- L2 is a true classic and is available in both software and hardware. This has probably appeared on more masters than any other limiter. It's not used as much today but certainly worth checking out as an alternative especially as it can be found on sale often. It's not as punchy as some current limiters, but it can work well on the right material.

MIXING WITH LIMITERS

Some artists and engineers suggest that you should produce and mix with a limiter on the master bus. This is a hotly contested debate. My advice would be to insert one but only engage it once the mix is coming together to hear how it responds to limiting. Any problems such as the kick or percussion pushing the limiter hard will show up. If you do find the limiter is behaving aggressively on any elements you should correct them. A good aim is 3–4 dB (max) limiting. With this, the sound is entering commercial loudness territory. Never mix long term with a limiter on, however. If the limiter remains active on the master bus, every time you increase the mix level, the limiter will push it back down. When the limiter is eventually removed for mastering, the mix will likely be overdriven and will require mixing again.

Most limiters include an output level. I would recommend you leave this slightly below 0 dB, around –0.3 dB. You will, however, notice that many commercial masters reach a maximum of at 0 dB. In fact, many might go above 0 dB when accounting for intersample peaks.

What are intersample peaks? Simply put, when digital audio is reconstructed with a D/A the filtering involved can create a level that is above 0 dB on a digital scale. This can mean distortion happens even if think your level is safe. This is where a limiter with intersample peak detection can be useful to help reduce the likelihood of these peaks cropping up at the consumer's playback device.

DE-ESSERS

Although de-essers are considered the mainstay of mixing when dealing with high-frequency vocal sibilance, they are sometimes employed in mastering.

The obvious application is removing sibilance in vocals where a change in the mix-down is not possible. This is only a case where a mastering engineer is performing the master. If you have access to the mix, the best approach is to return to the mix and solve the sibilance there.

Sibilant peaks do not occur only with vocals. Synthesizer patches that employ high filter resonance values or flanging effects can also produce forms of sibilance. Similarly, overly bright snares or hats in the 6–10kHz area can be treated with de-essers. They can also be useful where the overall track lacks high-frequency energy except for a small range of frequencies. By employing a de-esser to tame that problem, you can then boost the whole top end.

MASTERING EQ

Most hardware mastering EQs are designed with broader curves to offer broad adjustments. They will also usually provide a tight enough Q to take care of more surgical problems, and this is one area where digital EQs shine. These can offer countless bands with extremely quick Qs to notch out particular parts of the audio spectrum.

The amount of boost or gain applied during mastering is far subtler than in mixing. As a general rule, think of only using about 2–3 dB of gain, anything more, and it's time to return to the mix to solve the issue.

Digital plug-in EQs also have the advantage of recall in a project, something that hardware lacks. To offer the ability to recall in hardware, mastering EQs employ stepped parameters. These are generally in the range of 0.5 to 1 dB per step. This, however, not only increases the cost of the hardware itself but also makes finer adjustments than these impossible.

Ideally, mastering EQ will gently balance the overall spectrum of a track in a way that would be less practical to accomplish a channel-by-channel basis. What follows are a few examples of EQs suitable for mastering:

- Sonnox EQ. A classic EQ that includes a good array of different filter types and curves. The interface is a little dated by current standards but nonetheless remains a popular option.
- DMG Equilibrium is possibly the ultimate mastering EQ with a host of curves taken from notable hardware such as API, Pultec, SSL, and more. It also lets you dive into different quality settings via linear and minimum phase modes to balance CPU and sound. It can be daunting for newcomers, however.
- Fabfilter Q2 is equally useful in mastering as mixing. A match setting also lets you impart the curve of one reference to another.

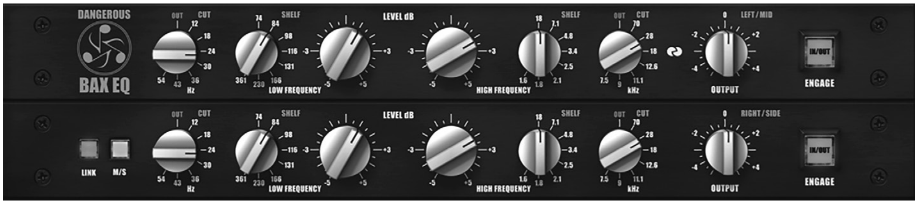

FIGURE 36.3
UAD BAX EQ

- Ozone's EQ offers both digital and analog style filters and curves with a matching EQ in a similar function to Fabfilter.

A useful option in some mastering EQs is M/S mode. This allows separate adjustments for the mid and side channels. A stereo channel can be encoded into separate mono and stereo channel information, processed and then decoded back to the original stereo signal. This makes it easier to fix problems such as too much low-end detail in the stereo image, but it can also be employed creatively, such as widening the mix by pushing out the sides at key frequencies.

One of the most useful applications will be the high and low shelf. A small boost in the top and bottom produces the familiar 'smile curve' we are all accustomed to hearing in commercial releases. An excellent variant on this shape is the Baxandall EQ. It's similar to a shelf, but instead of leveling off it curves back down for a more natural tone.

The Dangerous BAX EQ is a beautiful tool that offers this curve and is available in software form, both natively and on the UAD platform. Possibly the most famous example of Baxandall curves is found in the Pultec EQ. Available both natively and on UAD, it provides an easy way to massage the frequency balance of a mix gently. One positive side effect of these units is that they impart a pleasing phase shift. Although phase shift is usually avoided when processing audio, with the above units it widens the perceived image of a mix and adds warmth by simultaneously boosting and cutting the same frequencies. This can be so fundamental to the sound of a mix that many mastering engineers will use both a transparent, clean EQ and one with more color to add extra life beyond just clinical EQ adjustments.

DYNAMIC EQ

Alongside multiband compression, mastering engineers may also employ a dynamic EQ.

While multiband compression operates on some frequency bands within the audio, a dynamic EQ will cut or boost only when audio in a specific frequency range exceeds a user-defined threshold. This is very similar to a compression threshold, hence the term *dynamic* EQ. This has many uses during mastering. For example, if a note of a bass line is overbearing in individual sections of the program material, a dynamic EQ can be set to focus on the problematic frequency. With a suitable threshold setting, when the overbearing notes exceed the dynamic threshold at the EQ's frequency, they are reduced in gain.

It is not necessary to own a dynamic EQ to do this, and there are a variety of ways in which it can be accomplished with some lateral thinking. For instance, you could insert an EQ onto the master channel and follow that with a compressor. By focusing on and boosting the offending frequency while cutting those either side, the compressor will be triggered by these frequencies. This produces a frequency-dependent compressor similar to the action of a dynamic EQ. Alternatively, you could automate a standard EQ, but this may not be practical if the trouble spot occurs frequently.

Nevertheless, for small peaks and resonances, automating an EQ to cut out specific troublesome areas in the mix can result in a more transparent result than if it had been left on for the entire mix as is the case with a static EQ.

High or low passing the higher and lower frequencies of a mix is commonplace in mastering. Here a mastering engineer will employ filters with names such as Butterworth and Elliptic etc. While this is good practice, care should be taken to not harm the audio by applying them too strictly. Always listen comparatively to ensure they do improve the mix. Many EQs provide a parameter to adjust the gradient of the curve. This is measured in dB per octave. The higher the number, the steeper the curve becomes. For high or low passing, you probably want to employ a brick wall effect so that all frequencies are removed directly below (or above) the selected frequency.

The problem with this approach is the resulting phase shift that these curves impart on the audio above or below the cut-off point. Tightening the low end with a 96 dbB filter at 30 Hz results in phase shifting above this frequency that could make the low-end detail appear less focused.

Alternatively, employ gentle slopes of 6 to 18 dB and don't be afraid to keep the cut-off conservative. This is one area where the monitoring environment of a mastering engineer can help decide the optimal settings.

Finally, you should also consider the EQ algorithm you employ. Many EQs offer a choice between minimum phase or linear phase. The former is typical of a hardware EQ where phase shifts occur based on the shape and cut or boost of the band.

Linear phase doesn't suffer from this but instead introduces pre-echo. This is where EQd material occurs slightly before the rest of the audio, creating a small echo. This effect is not always audible, however, and depends on the program material. It's down to experimentation and comparing both minimum and linear phase units to determine which produces the most appropriate results. Generally speaking, the linear phase will sound more transparent, whereas minimum phase produces a forward effect that is preferable for when impact or punch is desired.

You may often come across references about 'magic curves' or EQs that miraculously boost without any adverse effect or harshness. While these can be used for mastering, in most cases, this is accomplished by employing a broad curve that is exaggerated by the interface. An EQ like this may appear to be heavily boosted, but in reality, it's only boosting by a few dB. Similarly, there are some EQs that boost in extremely high ranges such as 40 kHz. Here, the slopes of these bands are so broad they start at 10 kHz or lower but end up reaching far beyond the audible range.

Spend some time analyzing these EQs with a tool like Christian Budde's VST Analyzer or DDMF's Plugin Doctor to see what's going on. Seeing these curves can teach you a lot about how classic hardware units earned their reputation.

IMAGING

A common aim of mastering is to create an increased sense of space and depth to the music. This can be accomplished through various means, but is typically handled via a stereo imaging device. In the analog realm, these are commonly wideband units that cover the entire frequency spectrum.

Again, on a digital platform, there are more parameters available than on hardware. Multiband imagers are popular as they allow enhancement, or even tightening of different parts of the frequency spectrum.

Using this, you can mono out the low frequencies, while simultaneously widening the mid and high frequencies. Although with these processors it can be tempting to go overboard to make a mix sound wide, restraint is critical. Incorrect use of widening will result in a mix that sounds unnatural and is fatiguing to listen to.

Although we mentioned using M/S earlier with a capable EQ, a simple encoding and decoding plug-in can be used to allow the extra gain to be applied to the S (side) channel. This will raise up any stereo information to create a more extensive mix.

Most of these methods only work if there is sufficient enough information contained in the stereo channel. If there is none or very little, then the mix should be revised.

SATURATION

Although you might be wary of introducing any distortion at the mastering stage, it can be useful to achieve current levels without the side effects of limiting. Because saturation has a natural limiting effect, used sparingly, it can help contain peaks in a more musically satisfying manner than limiting. There are plenty of options for applying saturation in both wideband and multiband formats. Some notable examples are:

- Plugin Alliance VSM3, a model of a costly hardware unit that represents one of the most elegant saturation devices available. With multiple modes of distortion and the ability to process M/S, it produces a wide range of effects.

- Sonnox Inflator is a unique process that aims to emulate valve behavior by increasing the density of the material, including mixes. It's controlled by two sliders that allow more or less effect and a brighter or full tone overall for quick and easy results.
- Ozone's Multiband saturator is more complicated but also permits more control when required, especially handy during the mastering stage. You can target specific areas of the mix to increase density in the midrange while leaving the rest alone. Multiple saturation types are available with a mix knob on each band.

Other processors in this area include tape and console emulations. UAD's Studer 800 and Ampex ATR-102 add some subtle glue and harmonics. Waves NLS and Slate's VCC emulate the sound of a console's channel and mix bus sections. Appropriately driven they can also help give a more compact and robust mix.

The danger of all of these processes is that they are easy to overuse and can compensate for areas in the mix that might best be addressed there. However, used sparingly they can give a master that final level of warmth.

METERING

Having some quality metering while mastering will make the process much smoother. This is especially true when trying to master in a room and monitor system that may not be perfect. There are many metering packages available and some of the tools already covered will include some form of metering. But standalone packages such as Spectrafoo, Flux, and Spektre are also worth looking into.

One confusing area for producers is the choice of level metering. As the audio industry evolves, new ways to measure and standardize metering have emerged. Currently, the standards you will come across are peak, RMS, K-system (now superseded by R128), LuFS, and PLR.

For dance music production, it used to be enough to monitor your peak and RMS levels but with today's streaming standards monitoring LuFS can be helpful too.

As mentioned elsewhere, YouTube, Spotify, and other services auto-level music to a specific LuFS output. Therefore, if you aim to get your final master close to these standards, you can produce a loud master without some of the side effects from exceeding that range.

Referencing a frequency analyzer can help with avoiding room modal problems when mastering. If you are experiencing trouble monitoring and processing the lower end frequencies of a mix, a frequency analyzer can help you keep it under control. Issues with phase coherence in the stereo field can also be evident with these. There are multiple ways of displaying this information; iZotope's Vectorscope has three, for instance. Making sure low-end information is centred, and the rest of the material is not overly broad is the primary importance here.

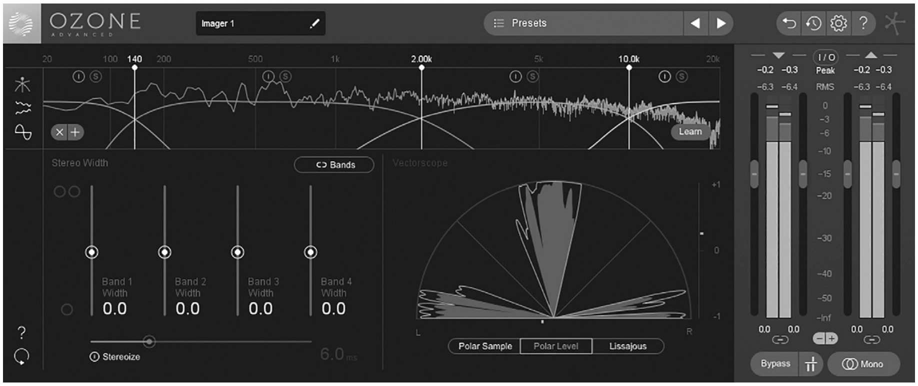

FIGURE 36.4
Ozone 8 Imager's Vectorscope

RESTORATION

Sometimes a master may reveal noise, crackles, and pops in a recording. Although these are best treated at the source sometimes, it is necessary to operate at the master level.

There is a variety of software to tackle these problems, available in both plug-ins and standalone software. Sonnox's suite of restoration tools allows for comprehensive work inside your DAW, while iZotope's RX suite can be used in standalone.

Noise is one of the most common problems encountered in the mastering stage. Using the software as mentioned above, you can select a section of only noise to create a 'footprint.' This footprint is then applied to the program material as a filter to remove the same noise across the entirety of the music. It's not an ideal solution, however. Even with the best algorithms, some artifacts will be introduced, proportional to the amount of processing applied. Consequently, only select parts of the master should be processed. Ambient intros and quiet breakdown sections are apparent choices and won't incur the same level of artifacts as more complex areas of the song. These are usually the only places where noise would be noticeable since noise is often masked in complex sections. Clicks, pops, and crackles can also be reduced significantly with the appropriate software module (most of these packages will include specialized modules for each problem type). If possible, zooming in and redrawing errors manually can work as a more targeted approach.

STEM MASTERING

Mastering may also occur in a *semi-mix* situation known as stem mastering. Here, instead of working with a stereo file, the song is subdivided into channel groups such as the kick, bass line, percussion, synths, vocals, and so forth. This approach offers the advantage of more in-depth control, allowing for adjustments on say the vocal without affecting the rest of the track.

The downside is that you don't fully commit to a final mix until the mastering stage. It can serve as an excellent middle ground, however. With the main elements rendered it prevents tinkering on the micro level and focuses your attention on more substantial moves.

In some genres editing at this level can result in creative processing that is not as easily accomplished during mixing like reversing a section of synths just before the break. Many mastering engineers don't consider stems as true mastering but don't let that stop you from experimenting with this method to see how it works for you.

OUTPUT FORMATS

When mastering is complete, you will need to export your master into some formats. The format will vary depending on your distribution channels. The standard format is 16-bit and 44.1 kHz, while for uploading Mp3s, 320 kHz CBR is a good choice.

Example mastering session: vocal house track

Below is an example of how a hypothetical mastering session might go when working on your material. Of course, settings will vary on your material so only uses these as a starting point.

As always, export your song after the mix is completed, keeping an eye on the master peak level. Set your master fader, or the output of the last mix bus process so that the highest peaks register around –3–6 dB. Select 24-bit and whatever sample rate you were working at.

You're happy with the mix overall, but the low end could use some tightening and the mid and top range lack clarity? The mix is compressed already, but it's missing some of the glue you hear on professional tracks? First, take an EQ set to a sharp Q, about 10, and then boost 8–10 dB. Start at 500 Hz and sweep down into just above the sub-region.

At around 160 Hz, you find a strong ringing resonance in the kick. Reduce the gain to 0 dB and set the Q to 3. Give your ears a few seconds to adjust to the unprocessed sound as the exaggerated response you just heard might tempt you to drastically cutting here.

After this short break, reduce the level to around 2–3 dB. Raise and lower it until the main resonance is gone but without losing punch in the kick. Toggle the setting off and on to verify it's fixed the issue without loss of valuable information.

The low end is smoother now without this resonance masking the bits we want to hear, but it's still not as solid as we'd like.

Next, open up a multiband compressor and solo the lowest band in the bass range. Move the cut-off until you hear only the kick and bass. Set the ratio to 2:1, attack at 20ms and release to 100ms. Lower the threshold until about 4–5 dB of gain reduction.

Adjust the attack until the amount of punch is desired, avoiding distortion of the transient at faster attack settings. Set the release until it matches the groove and tempo of the song. Too fast can incur distortion, too slow and it can drag the energy down.

Now, back off the gain reduction so that only 1–2 dB is being reduced. We used more reduction earlier just to focus our adjustments.

Depending on how often the reduction is being applied, you might need to then add some overall gain to compensate. Something around 1 dB should work as the peaks at 2 dB will only happen every so often.

Using an EQ with M/S capability, set a low cut filter to the S channel and cut around 100 Hz, sweeping up until any unwanted low-frequency information is removed on the sides. This helps focus the low end on only the mono channel for a tighter low end.

Then add a band of EQ set to a wide Q, 0.7. With about 4 dB of gain sweep around the midrange until the synths and vocals become more intelligible but without any harshness. Reduce the gain to 2 dB, again toggle off and on to make sure it's made a positive impact on the sound.

The track is sounding clearer, but it could use a little more sparkle? We add in a shelf at 18 kHz which extends down to around where our mid-range boost was. Slowly raise the gain until the track becomes crisper but without upsetting the overall balance.

There's a small amount of low-end rumble, so another high pass filter is added but this time to the mono channel. Sweeping up to 30 Hz with a slope of 12 dB manages to tighten up the very sub-region without losing any needed warmth.

A complementary 18 dB low-pass filter is set to 17.5 kHz across the entire stereo channel to filter out any sharp information that might come through from the previous high shelf boost.

Next, we add in an instance of the Glue SSL compressor and set the side-chain filter to 100 Hz to avoid pumping on the kick.

The attack is set to 30ms, release to auto, and the ratio to 2:1. Lower the threshold over the loudest section of music until the meter just barely moves. This doesn't change the overall dynamics, but now the track is more robust and together.

Finally, we wrap it up by adding an instance of Barricade set to 0.020 attack and 0.100 release. The threshold is lowered so that 3 dB of reduction is gained.

We check against a couple of references and find out track has enough level to compete with other records in the genre, job done.

The mix is bounced, put into an editor where the front and tail end is cut so that any extra silence is removed. After letting it sit for a day, we're happy with the result, time to send off to DJs and post to social media.

CHAPTER 17
Mastering

The preproduction, engineering, and mixing stages aim to capture, combine, and optimize the best possible performances and sounds to create the most effective stereo picture. Mastering examines this final picture and enhances the image for maximum impact, framing the music in a way that allows for the best possible reproduction—regardless of format, playback system, and listening environment.

For this book to completely cover mastering, the following sections would be required: delivery formats; DDP (Disc Description Protocol); metadata such as ISRC codes; PQ sheets; sample rate conversion; and dither (word length reduction). These technical tasks are covered by other publications, and involve the same procedures regardless of genre. Consequently, this chapter mainly focuses on the key corrective and creative processes involved in mastering metal music. A separate chapter then looks at mastering for loudness-normalized environments.

DIY AND THE "FOUR Es" OF MASTERING

Mastering is the final creative phase of the production process prior to the music's distribution, and therefore the very last opportunity for any deficiencies to be corrected. So there are numerous reasons to avoid "do it yourself" mastering, particularly if the end product is to be properly marketed. As just one example: if a project is mastered within the same environment and through the same monitors used for mixing, any deficiencies this failed to expose are less likely to get revealed and corrected during mastering. By handing a project over to a professional mastering engineer, it benefits from an impartial objective perspective formed within a finely tuned, acoustically "flat" room with high-resolution monitoring. Crucially, it also benefits from the engineer's critical listening skills and experience. This tends to result in a far higher production standard than a DIY approach. The "four Es" of mastering are therefore: "*ears*"—critical listening skills to inform the processing applied; "*experience*"—through which these skills were developed; "*environment*"—the acoustic space; and "*equipment*"—monitoring and signal processing focused on the specific requirements of mastering.

All the same, professional mastering engineers are far from cheap—especially those with substantial experience and a strong CV—and at the other end of the spectrum, many cheaper online mastering services comprise generic loudness processing, and little else. So it makes sense to develop your own mastering skills. By doing so, you also gain an understanding of what makes a great mix from a mastering perspective, enabling your subsequent mixes to contain fewer weaknesses than otherwise.

SIGNAL CHAIN

The first mastering task is to duplicate the relevant tracks so there are two versions. By leaving one of them entirely unprocessed, easy access to the original unmastered mix is enabled, without the need to bypass the mastering processing. This is vital; to prevent the intended aesthetics of the original mix being compromised, you should carry out level-matched (original and processed) A/B listening throughout mastering.

From here, when required, it is first good practice to remove or minimize any inherent noise or hum issues. This often reflects the capture of air conditioner/computer fan airflow noise, or electrical hum in a flawed recording environment. Your speaker monitoring and mix-room acoustics may not be precise enough to reveal these problems, whereas good-quality headphone monitoring usually will. Noise reduction with EQ can degrade valuable spectral content, so when possible use dedicated audio repair software (e.g. Waves Z/X-Noise, iZotope's RX 5, or Wave Arts MR Hum/MR Click, to deal with these obstacles).

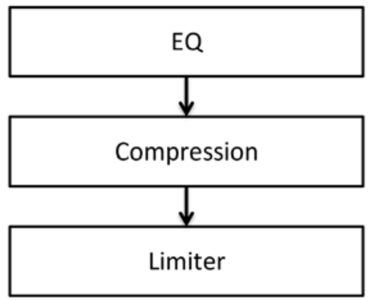

FIGURE 17.1

> With so much music now accessed via earbuds/headphones, it is vital these playback mediums are represented during mastering. Good-quality headphone monitoring can also help reveal flaws less apparent through loudspeakers.

Although there is no single correct sequence to follow, the mastering signal-chain order can dramatically influence the final resulting production. Mastering normally involves relatively moderate EQ modifications, which can therefore be applied pre-compression. But if significant *additive* EQ is required, a post-compression application tends to be preferable. To act as a final level cap on the earlier processing, limiting/loudness maximization needs to be placed last in the chain. This means that if only modest EQ treatment is required, a simplistic processing order starting point could be as shown in Figure 17.1.

If harmonic excitement, stereo widening, further (additive) EQ, and soft clipping are required, this processing order could be amended as shown in Figure 17.2.

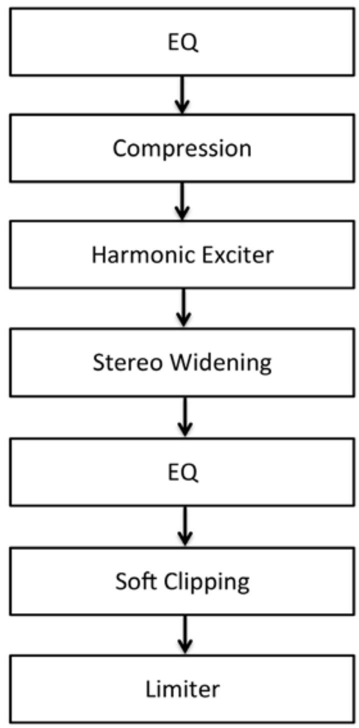

FIGURE 17.2

These figures are not meant to suggest a *correct* signal chain order, but could form an appropriate starting point. Also worth considering is that you don't necessarily have to move through the signal processing chain in the relevant sequence. An initial application of limiting can sometimes enable more valid (pre-limiting) EQ and compression gestures.

> "I tend to EQ before compression when mastering. However, if I need to add some low end to the production, I usually compress first."
>
> Jens Bogren

> "Every project comes in with its own mastering requirements. Some need more enhancing then fixing, some need more fixing, and some combine the two. I have a few different signal chains to address those situations. Applying EQ before compression helps clean up problematic frequencies before they reach compression. But in some cases, the compression overreacts to the lows, so you have to set the side chain of the compressor to ignore those frequencies. EQ after compression can help you shape the tone after it has been dynamically modified, which sometimes helps you bring back what was lost during the compression stage."
>
> Maor Appelbaum

MASTERING EQ

This section is presented before compression, but bear in mind that significant additive EQ gestures tend to more effective post-compression.

High-Pass Filters and Low-End Control

A tactically set HPF makes a suitable corrective EQ starting point. Even when a mix's low-frequency energy is controlled and naturally rolls away, it is good practice to attenuate sub-40 Hz sonic sludge frequencies with an HPF. To put 40 Hz into perspective, anything below this frequency has a single wave cycle over 28 feet/8.5 meters in length. Wavelengths this long/slow consume valuable headroom and without advancing the production's low-end weight in any way. You might not detect the influence of this energy due to monitoring that is unable to reproduce it. So a spectrum analyzer can prove valuable (see Figure 17.3), especially as this allows you to make comparisons with the spectral content of reference productions. You shouldn't be aiming at entirely *removing* frequencies within this sub-40 Hz region, as this would likely require an HPF cutoff setting inappropriately high up the spectrum. Nevertheless, attenuating this energy redirects the available headroom toward the music's more essential and effective low-end regions.

For mixes that display pulsing, sluggish lows, the HPF may need to be lifted as high as 55 Hz—or even higher in extreme circumstances. This setting may sound excessive, but the 55–125 Hz sonic weight/low-end foundation region translates a more effective sense of heaviness, impact, and perceived power than the 20–55 Hz sonic sludge range that can obscure these qualities. However, bear in mind that higher HPF cutoff and pole position settings can result in phase shift that becomes splayed across the pass band, sometimes with an audibly detrimental impact. Linear phase EQ

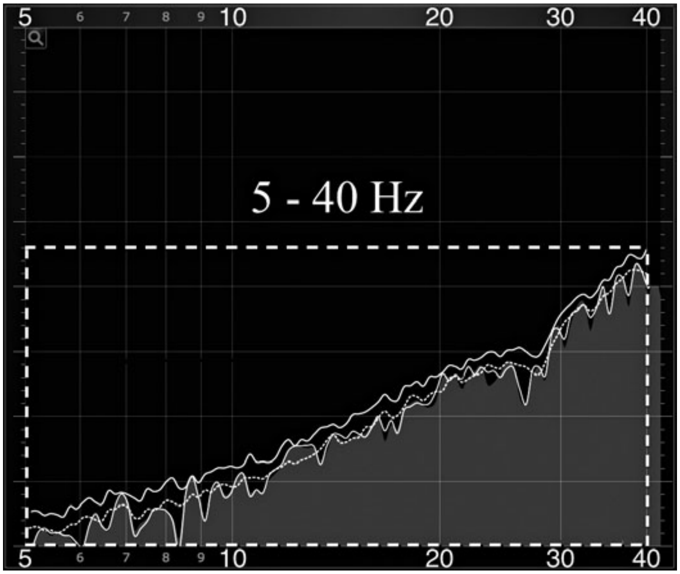

FIGURE 17.3
Figure 17.3 shows the HOFA IQ-Analyzer zoomed-in to display the 5–40 Hz content of a pre-mastered project. Despite firm HPF use throughout the mix stage, there is still significant energy present. Due to the lowest section falling below the threshold of human audibility (lower than 20 kHz, referred to as infrasonic), and due to monitoring that is unable to reproduce them, the influence of these wavelengths might not be detected. Without benefitting a production's low-end weight or impact, this content demands valuable headroom.

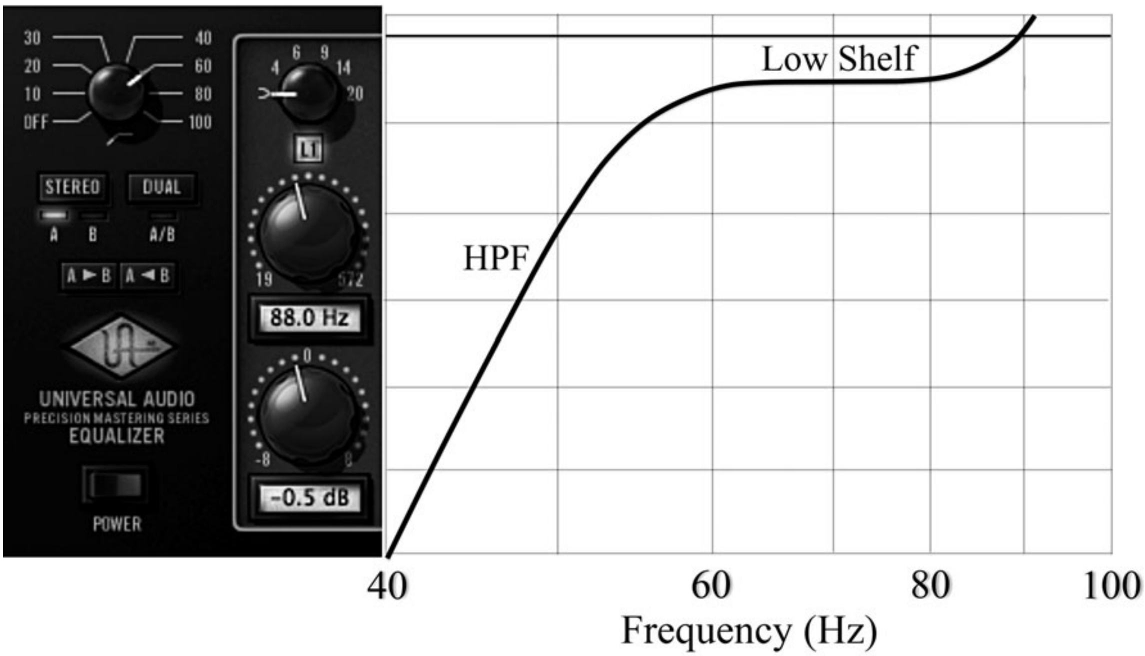

FIGURE 17.4 Figure 17.4 shows the combination of an HPF and low shelf being used to sculpt the disproportionate, uncontrolled low-end energy of a problematic mix. Whereas the HPF, with a cutoff setting at 60 Hz, provides aggressive attenuation to the very lowest frequency content, a further more subtle attenuation of the lower frequencies is required, so a low shelf, set at 88 Hz, is introduced, providing a moderate 0.5 dB of attenuation. An HPF alone would have been an overly heavy-handed approach.

largely avoids these artifacts (although minimum phase EQ might offer a comparatively more aggressive sound).

Be careful not to overstate the role of an HPF, though. If you have implemented an HPF, and only subtle further attenuation of the lows is required, adding a subtractive shelf can be preferable to elevating the HPF, while affording improved fidelity (see Figure 17.4).

> "A lot of metal productions have a low end that extends lower then 40 Hz, which can be taxing on the speakers and can easily clog the system. Cleaning this up with a high-pass filter can help gain clarity. Filters can be your friend or your enemy, so listen carefully to what you lose from their use, but also what is gained. In some cases, shelving EQ can also help clean up the low end, as they touch a wider spectrum, and can be combined with an HPF to achieve the desired results."
>
> Maor Appelbaum

Low-Mids and Mids

Having dealt with any nonessential or counterproductive low-frequency energy, it would be tempting to move directly to the sonic weight region somewhat above this. However, it can be preferable to consider the low-mids first, providing an improved context in which to judge the lows.

A mix that contains muddy qualities is often the result of excessive content within 200–550 Hz (regularly around 230 Hz, but occasionally as low as 150 Hz)—with energy closer to 550 Hz tending to have a nasal quality. But take into consideration that perceived muddiness can also be the result of a deficient or insufficient top end. Compared to the low-mids, the 550 Hz–1 kHz region tends to require less attention when mastering. But an overemphasis within this range usually results in a mix containing what can only be described as "honky" qualities. Bear in mind that too much subtractive EQ to the mids quickly undermines the power of a mix. And in circumstances where there is an apparent midrange spectral "gap"—usually reflecting excessive attenuation during the mix—a thicker/fuller sound is afforded through additive EQ to the appropriate region.

Wider bandwidth settings, less than one, are required for broad spectral modifications, while tighter "Q" settings—greater than five but sometimes much higher—are required when narrower regions of detrimental energy are present. Either way, very small incremental adjustments should be experimented with. Remember that each EQ gesture carried out in mastering impacts every sound and effect that has energy in the relevant spectral region.

A common mistake—particularly among mix rather than mastering engineers—is to try to focus EQ moves toward a single instrument, while overlooking the impact this has on other sounds. For instance, attempting to accentuate the kick impacts the overlapping frequencies of the bass, and brightening the snare affects the brightness of the vocal. Mastering EQ is equivalent to going through an entire mix applying the same EQ curve to every channel. With this in mind, half dB steps or less can have a profound impact. Occasionally, you may need to go further, perhaps 1–2 dB, but anything more should only be required for mixes with specific flaws or deficiencies.

Low-End Foundation

> "Low-end boosts are a tricky area, as they can add a lot of body and power but at the same time can clog the sound in a way that loses punch and clarity. When adding low end, always check if it's needed or you just enjoy hearing it bass-heavy; it's very easy to go overboard. Shelving EQ can help get a full-bodied sound, but they can also fill the lower spectrum too much, causing muddiness. Bell-shaped peaking EQ can be easier to work with for achieving a tighter, more controlled sound."
>
> Maor Appelbaum

Having potentially modified the mids, additive EQ can be applied to fill out the low end if required. Defaulting to the use of a precise bell-shaped curve for emphasizing the lows tends to be preferable, with bandwidth settings in the region of 0.6 (very wide) to 0.9 (slightly tighter) often appropriate. Boosting within the 65–90 Hz region, occasionally lower, can be suitable for performances with slower subdivisions, moving through to a higher 85–130 Hz range for faster performances that have likely involved more aggressive HPF settings during the mix.

When a broader emphasis of the lows is required, the use of a bell-shaped curve may be too focused, so a shelving filter is preferable. Bear in mind that all frequencies below the shelf setting are impacted—meaning that unwanted sluggish content can easily get emphasized. Rather than automatically elevating the HPF to deal with this, a sometimes effective approach is to use two low-frequency shelves—one with a boost and one with a cut—so that a small "plateau filter" is created. This tends to provide a more natural-sounding result than the use of two bell curves, or the combination of shelf and HPF for this purpose. However, a plateau filter can introduce phase issues, especially when shelving filters with steep slopes are involved, so employ linear phase EQ when possible.

If possible, retain the phase relationship between the center and left/right of the mix by using stereo EQ. But in circumstances where there is a detrimental spectral imbalance between the center image and left/right image, mid/side EQ may be required (discussed later).

Upper-Mids and High Frequencies

The various frequency ranges are heavily interactive, so additive or subtractive EQ to the lower regions can result in the highs being perceived as duller or brighter. For example, attenuation to the 200–550 Hz mudrange can have a similar psychoacoustic effect to amplifying the 4–7.5 kHz low-highs—and vice versa. Before automatically dialing-in the highs to brighten up a slightly dull mix, then, first consider attenuation of the lows, or low-mids. If this is inappropriate, or has already been carried out, experiment with broad but moderate additive EQ within the 6–12 kHz region using a bell-shaped curve with a broad bandwidth, perhaps 0.6–0.9. This should introduce brilliance and sheen to the production, but hopefully without overly accentuating vocal sibilance, usually found within the 6–9 kHz region.

For a perceptually lighter, more transparent brightness that can subtly enhance a production's sense of space and air, a frequency center between 10 and 14 kHz can be used. This is a processing area where high-quality mastering grade EQ (e.g. Manley Massive Passive) can prove most valuable, allowing a mix's top-end energy and detail to be highlighted in a smooth, musical (non-grainy) way. If a focused boost fails to provide the right impact, a Baxendall curve commencing from around 4 to 10 kHz may be preferable. But take into consideration that significant additive EQ to the 10–17 kHz upper-highs can result in an artificial "fizziness" to the cymbals and overall production, heightening ear fatigue in the listener. Also, be aware of high-frequency boosts unintentionally accentuating hiss/noise residing upward of 14 kHz, potentially requiring an LPF to correct this.

> "Shelving EQs are great for brightening up a mix as they cover a wide area. However, it's very easy to go overboard with them and make a mix sound brittle and harsh. Using a bell-shaped EQ, you can hone in on the frequencies that sound pleasant to the ear, or smooth the ones that obstruct this content. You can brighten a mix by cutting some low-end or low-mid frequencies as well."
>
> Maor Appelbaum

Throughout the process of emphasizing the highs, it may become apparent that there are areas of harsh, abrasive content that obscure how smooth the mix's top end is. This sometimes reflects the uppermost 1.5–5 kHz energy of the bass interacting with the same region of rhythm guitar content, or occasionally due to bright vocal content merging with the cymbals and uppermost 7–8 kHz content of the rhythm guitars. But of course, less focused top-end additive mix EQ may have left narrow pockets of resonance in the collective instruments, which become further highlighted with mastering treatment. After establishing the center frequency, and bandwidth over which the detrimental energy is dominant, the level of cut often needs to be negotiated with any high-frequency boosts applied, with each informing the other. Whereas with subtractive and additive EQ in quite different areas of the highs, tighter surgical EQ cuts may need to be compensated with moderate broad bandwidth boosts at the same frequency.

Stereo EQ vs. Mid/Side EQ

Excessive or unnecessary processing is a common novice error when mastering this style of music. In many cases, this isn't disproportionate EQ, compression, or limiting, but needless or inappropriate multiband and/or mid-side processing. These processing tactics can be powerful and effective when employed in a discerning and measured way, but perilous tools in the wrong hands.

The ability to rebalance the levels and separately modify the spectral qualities of the mid and sides of a mix affords significantly increased control. However, partly due to an impaired phase relationship, this can result in the mid and sides becoming "incoherent/disconnected." Unless you are mastering a project with specific shortcomings, then, it is preferable to start with the application of stereo EQ, rather than mid/side EQ. Corrective and creative EQ to the stereo signal should determine whether there is anything apparent in the mid or side components that need to be separately treated. For example, if the kick and bass contain detrimental low-mid frequencies but attenuating this region damages the thickness and prominence of the rhythm guitars, mid/side EQ could be preferable. Similarly, if the rhythm guitars have an excessive, uncontrolled low end, or simply aren't bright enough, compensating for this with stereo EQ could be detrimental to the kick, bass, snare, or vocals.

> "I'll sometimes use mid/side EQ during mastering, perhaps if I need to bring the vocal out in the middle. It can also sound a little wider if you put some more top end on the sides, and can give the cymbals that width, and a bit more edge to the guitar. Sometimes I'll push the low end in the middle of the mix as well."
>
> Andy Sneap

As ever during mastering, though, restraint and moderation are required. Separately amplifying the low frequencies of the centrally panned kick and bass can enhance their weight and impact. But due to the reduced impression of the sides, this emphasis can quickly result in a narrower stereo image. Conversely, boosting the highs of the side signals where guitars and cymbals dominate can enhance a production's sense of "air," width, and size. But due to the kick/bass/snare/vocals mid-image potentially becoming slightly overpowered by the bright aggressive energy from the sides, this can lead to an effective "chasm" in the middle of the mix. Additive EQ to the sides can also result in the left/right of the stereo reverb and delay treatment getting detrimentally exaggerated. You simply need to evaluate what is lost through the processing, and offset this against what is gained.

LOW-END LOCALIZATION

Although human hearing is less able to localize lower-frequency energy, insufficient lows in the side signals—primarily the guitars—result in a natural emphasis toward the "weight" of the center image, with the low end of the sides thereby overpowered. Consequently, regardless of how this might be localized, separately amplifying the side signal's lows can be appropriate and effective.

When a production requires it, an almost contrary approach can be equally valid. Especially with radically down-tuned guitars, "low-frequency mono summing" can tighten up the low end of a production, enhancing its impact across different playback systems. Sometimes referred to as elliptical EQ, this involves folding the deepest bass frequencies from the left/right of a mix into mono/the center. Not all mid/side mastering plug-ins provide this capability; one that can is the Brainworx Digital V2, which is able to mono-out the bass frequencies of a stereo mix via its "mono-maker" function. To avoid excessive loss of stereo width, it is generally preferable to restrict any mono-out bass frequencies to the sub-70 Hz region of content.

Mid/side processing can also be a useful educational tool. By isolating the center image of a high-quality reference production, the rhythm guitars and effects processing get attenuated in a way that helps reveal the spectral/dynamic relationship and relative levels of separation between the kick/bass/snare/vocals. Equally, by isolating just the sides, the extent and decay time of the reverb use becomes clearer, as does the pan positions and resulting image of the rhythm guitars, and widest drum shells and cymbals.

REVERB

Mastering stage reverb processing can introduce an increased sense of depth and space into a production, as well as some audio "glue" that gels the ensemble together. But unless a project features stems (discussed later), there is no way of treating certain instruments without this simultaneously influencing other sounds. So at the same time as reverb potentially enhancing a production's sense of dimension and unity, this can quickly soften the power and intensity of the guitars and bass, while creating an impression that the production is disordered. All the same, for spatially deficient mixes, or those lacking a sense of cohesion, very short reverb times less than 0.2 seconds can be constructive, but set at such a low level that—when alternating between muted and active—the processing is barely perceptible.

UNIFIED MASTERING—BRIDGING THE DIVIDE

> "I often master simultaneously during the mix . . . I took this approach with the Gojira album *The Way of All Flesh*."
>
> Logan Mader

Mixing and mastering are two very different art forms, and keeping these phases separate enables greater objectivity. For many years, CPU/memory limitations meant there was no option, but recent computer-processing advances now enable dedicated mastering processing to be active during the mix.

Given the constant alterations made as a mix progresses, unified mastering presents a quandary over which aspect to focus on. If inappropriate mastering processing is applied early on in the process, critical mix decisions can become entirely misguided, with these flaws only revealed when the mastering parameters are corrected. Despite this, and regardless of the potential increase in option anxiety, if a valid mastering blueprint is used from a relatively early stage of a mix's development (possibly the final processing from a previous mastering project of a similar style), this can be an effective way of removing the traditional divide between the mix stage and final product.

COMPRESSION

Effective loudness, punch, and power is achieved through the combined impact of numerous dynamic and spectral processing instances, most of which applied during the mix. As highlighted in Chapter 13, "Compression," the cumulative approach to compression is vital. When single compressors are worked hard (i.e. providing heavy gain reduction), the processing tends to become both increasingly unnatural and increasingly obvious.

When mastering a project not mixed by yourself, the cumulative compression principle is largely out of your control. All the same, if your mastering compression becomes audibly obvious, a change of parameters or approach is required. Particularly worth listening for is unnatural cymbal dynamics, or the audio becoming thick and congested compared to the unmastered version.

Broadband Compression vs. Multiband Compression

> "I rarely use multiband compression unless it's a rescue operation, and I hardly use limiting. Limiting and multiband compression are the two processing approaches that people tend to overuse when it comes to mastering, as both can really suck the life and punch out of a mix."
>
> Jens Bogren

Broadband compression processes the entire spectral range of an audio signal, whereas multiband compression involves filters that separate the audio signal into two or more frequency bands, with each of these bands assigned an individual compressor. As well as enabling different compression settings to be used for each band—with more gain reduction applied to the spectral regions that require it most—the key advantage of multiband compression is that loud events in one frequency band don't instigate gain reduction in other bands.

Despite the flexibility provided, multiband compression in the wrong hands is a fast-track route to wrecking a strong mix. The different gain reduction in each band can produce an unnatural sound, while unavoidably changing the mix's original frequency equilibrium, which then has to be reconstructed/restored. With a mix that is instrumentally and spectrally well balanced, with controlled effective dynamics, particularly in the lows, the use of broadband compression is more likely to retain these qualities than multiband compression.

On the other hand, if you are dealing with a mix that is dynamically unstable, or perhaps lacks solidity in specific frequency areas—particularly the lows, the most challenging region to effectively stabilize—attempts to control this with broadband compression can result in unnatural gain reduction applied elsewhere in the spectrum. This is especially the case as the lows tend to dictate how much gain reduction a broadband compressor applies and when. Due to the high frequencies being compressed in unison, the upper regions can become dulled and unnatural modulations can occur, often evident in cymbal sounds with unnatural sustain qualities. A side-chain HPF can help compensate, but only to a certain extent, as the lows are benefitted by gain reduction applied in direct response to its energy fluctuations. These issues and challenges are avoided with multiband compression. Likewise, if EQ has proved ineffective at reshaping the lows/mids/highs frequency distribution of a problematic mix, multiband compression proves invaluable, with the two to four separate bands providing a simple EQ unit.

> "Another option is to use multiband compression to shape the overall sound in a different way then it came in, and adjust the levels of the different bands to fit with the sonic balance you are seeking."
>
> Maor Appelbaum

Regardless of the compression approach adopted, it is vital you assess the processing's impact on the instrument balance of the unmastered mix. This differs according to circumstance, but, as highlighted in Chapter 16, "Master Buss Processing," the kick is usually the first element pushed down in level when mastering compression is applied. This reflects its peaks, usually being the first and loudest signal to exceed the compressor threshold. Depending on their mix balance and frequency attributes, the snare, vocals, and sometimes even the bass can also be pushed down in level, but usually to a lesser extent than the kick. Conversely, the sustain-based, dynamically stable nature of the rhythm guitars tends to increase in level—as do the quieter elements of the mix, including the natural ambience captured in the recordings, and the reverb and delay tails. To restore the original mix balance, or indeed enable a preferable balance, the interaction between mastering EQ and compression needs negotiating.

Broadband Compression

Generally speaking, there are two common approaches to broadband mastering compression, largely reflecting how low the threshold is set and how high the ratio. The first, typically more transparent approach tends to be appropriate for dynamically well-controlled mixes. With a very low threshold, perhaps between –35 and –45 dBFS, the compressor is responsive to the body, rather than peaks of the signal. While this means that gain reduction is permanently provided, the levels are kept relatively moderate due to a low ratio such as 1.3:1–2.5:1, and smoothly applied due to the use of a soft-knee setting.

Despite being less effective at controlling peaks, this approach still has the potential of softening transient energy, so an appropriate attack time is vital. Although 20–30 ms makes an appropriate starting point, for productions involving very fast subdivisions with short, sharp drum transients, this setting could be shortened to adapt to the compacted, transient energy. Likewise, with "bigger" collective drum and bass sounds, the attack setting could be somewhat lengthened.

For the release time, a relatively fast 100 ms makes a suitable initial setting, but adjusted according to how fast the performance subdivisions are. With slower performances, a longer release time can increase the stability of the low end, enabling a heavier overall production. But if the release parameter is too long, the continued gain reduction has a detrimental impact on subsequent transient detail (see Figure 13.6 in Chapter 13, "Compression"), thereby requiring a faster release for faster performance subdivisions. All the same, if the release time is *too* fast, the compressor does not influence the audio as much as it should, while potentially resulting in unnatural gain reduction. And, to further complicate matters, fast release settings can result in signal distortion. For many mixes, particularly when there are complex or frequent changes in performance tempi, engaging the auto-release control can be effective, allowing the recovery to adapt to the changing dynamic detail. But, of course, the success of this tactic is dependent on the relevant auto-release characteristic itself, which might range between clean/transparent through to fast/distorted.

Initially aim for around 4–5 dB of gain reduction, and as always apply appropriate make-up gain. Partly dependent on how much compression was applied during the mix, some tracks immediately sound squashed and display unnatural gain changes with this level of gain reduction, so a lower 2–3 dB level is required. However, others can easily withstand several dB of gain reduction, sometimes more, and without any "pumping" artifacts. In either event, the amount of gain reduction from your compression likely needs revisiting later in the mastering process, and adjusted according to how this combines with limiting. In combination with the production's spectral content, this interaction has a significant influence on the inherent loudness of the final mastered product.

> "Longer release times can tame and round the sound more; in some cases, this makes the music feel heavier, with less high-end intensity."
>
> Maor Appelbaum

This first, usually more subtle broadband compression approach brings all the audio content up to a more consistent level with comparatively consistent gain reduction. A stronger, more influential application—better suited to mixes with less well-controlled dynamics—involves fluctuations between higher and lower levels of gain reduction. This requires a higher ratio—perhaps 3:1, but usually no higher than 4:1—a hard-knee characteristic, and a threshold setting between the program material's peaks and valleys. The gain reduction meter should therefore reflect the compressor reacting to the beats and peaks of the music, rather than the entire dynamic range. A similar 20–30 ms attack time starting point discussed for the first approach is relevant, as are the release considerations and levels of gain reduction.

Of course, there are no rules to follow with each of these two approaches. Either tactic may require higher ratios or lower threshold settings to enable the control required. Similarly, a two-stage serial dynamic approach can be valuable, potentially exploiting the use of different compressor design types (see Figure 17.5).

MID/SIDE COMPRESSION

Mid/side compression becomes necessary when the dynamics of a mix's stereo image is defective, and the application of compression fails to correct this, or even exacerbates the situation. For example, excessively distorted rhythm guitars could become even more brittle with stereo compression, and in a way that EQ fails to adequately correct. Similarly, a mix with a weak center image might benefit from this aspect being more heavily compressed than the sides, with the resulting level compensated via make-up gain. As with mid/side EQ, though, mid/side compression can subtly yet appreciably disconnect a mix's center and side components, while damaging the stereo effects processing's natural decay properties. This can quickly result in an artificial-sounding production.

Side-Chain Filtering

One of the common problems with mastering applications of broadband compression—and particularly with this style of mix—is the low end having a disproportionate influence on the gain reduction applied. By applying an HPF to the compressor's side-chain (see Figure 13.16 in Chapter 13, "Compression," and Figure 17.5 overleaf), this makes the unit's level detector less sensitive to the lows, which can help to reduce pumping artifacts and help retain more natural dynamics across the spectrum.

The potential disadvantage of side-chain filtering is the reduction in dynamic control to the low end. In a situation where side-chain filtering is required to prevent unnatural modulation to the highs, but this results in the lows being insufficiently contained, multiband compression might offer the appropriate solution.

FIGURE 17.5 The Shadow Hills Mastering Compressor features separate optical and VCA dynamics stages. With both engaged, the smooth, musical qualities of the optical stage is followed by a more forceful, yet more controllable VCA stage. Figure 17.5 shows the side-chain HPF option engaged (circled), which prevents frequencies lower than 90 Hz affecting the gain reduction in both stages.

Multiband Compression

Before considering the parameters, the first step with multiband compression is to determine how many bands are required, and where the band crossover point(s) should be set. Unless there are particular mix deficiencies that require four bands of compression, it is preferable to default to three, or (often overlooked) the use of two bands. In any instance, the crossover point(s) need to be set in a musically appropriate way that engages with the challenges you are trying to solve. This process is made easier by auditioning each band in isolation.

Two-Band

When mastering a well-constructed mix that doesn't react well to the application of broadband compression, a two-band approach is often more effective than a three-band. In order to avoid unnecessary modulations in the upper part of the spectrum, this simply involves a low-frequency band and a low-mid upwards band. A single crossover point just above the most prominent combined sonic weight of the kick and bass, perhaps somewhere between 140 and 220 Hz, tends to be effective, with higher crossover points appropriate when radical mix-stage HPF settings are evident.

Three-Band

When the additional flexibility and control of three separate bands is required—usually as a result of the mid and high frequencies reacting defectively to the same gain reduction—an additional crossover point that roughly divides the high-mids from the low-highs is required. A useful starting point is 2 kHz. The middle band below this represents the warmth and midrange character of the mix, and above this is the area we generally associate with treble. For some mixes, this results in a less focused bass sound, as the wiry note definition and attack from the 1–3.5 kHz spectral region gets impacted by both sides of the crossover point.

An alternate starting point is therefore the 4 kHz region—the point at which the bass is normally starting to roll away, and just below the presence region of the rhythm guitars. In the event that heavier gain reduction is required for the highs—perhaps to provide increased high-end solidity—experiment with moving the high crossover point up to around 6 kHz. This helps to avoid the presence range of the guitars becoming overly sharp and dominant as a result of greater compression.

Four-Band

In the less fortunate event of the mid-highs and extreme highs needing to be compressed separately, a crossover band can be set at the point where the rhythm guitars have largely rolled away, usually 8–8.5 kHz. This enables separate control of the lower-highs and mid-through-to upper-highs. But when a fourth band is required, the crossover setting needs to be informed by the lower crossover. In the instance this was set to, for example, 6 kHz, the intermediate 6–8.5 kHz band would serve little purpose.

Multiband Parameters

The following parameter guidelines discuss the frequency bands from the approximate perspective of lows, mids, and highs. This needs to be modified according to how many compression bands you are using. With two bands, the following mid- and high-band discussion needs to be considered collectively, whereas if four bands are required, the high-band section needs relating to two separate upper bands.

Low Band

The low band tends to be the region that benefits from the highest overall level of gain reduction, affording stability and solidity the other frequency bands can build upon. If (despite the requirement for multiband compression) the low end of the mix is relatively powerful and controlled, a low threshold combined with low ratio tends to be effective. Depending on the signal strength in question, this could be in the region of –40 dB, and 1.5:1–2:1, respectively. Initially, aim for 6–8 dB of gain reduction during peaks, with equivalent make-up gain applied (avoid using auto make-up gain), and revisit these settings and resulting level of gain reduction once the low band is heard in context.

If the low end is somewhat unstable, weak, or simply needs enhancing, a higher threshold—somewhere between the peaks and valleys of the band's dynamic content—and a slightly higher ratio, perhaps 2.5:1–3:1, should lift the low end into focus. For firmer low-end control, increase the ratio. But be aware that, after you have applied suitable make-up gain, heavier compression brings a spectral region "forward" in the mix. So make sure the lows don't overpower the mids and highs after they have also been separately compressed.

There isn't any upper transient energy that can be compromised in the low band, so a fast attack time can be both appropriate and effective. For some mixes, a setting within several milliseconds, sometimes as fast as possible, is suitable. However, you need to ensure this doesn't result in obvious distortion artifacts, which is also informed by the release setting (see Figure 13.10 in Chapter 13, "Compression"). As a starting point, try dialing-in a fast release, and then slowly lengthen the setting until the distortion is minimized. This often requires the attack setting to also be somewhat lengthened.

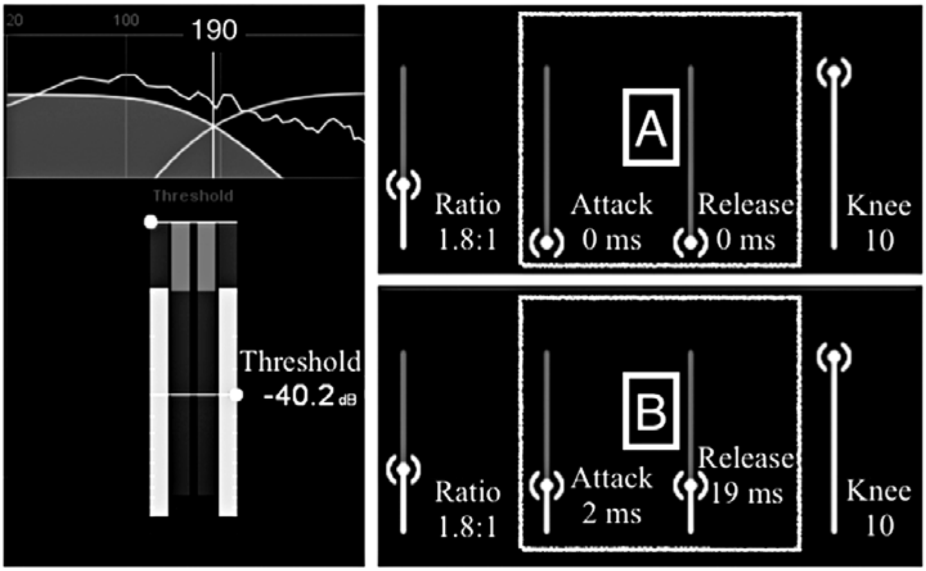

FIGURE 17.6 Figure 17.6 shows that 190 Hz has been established as being above the region where the combined sonic weight of the kick and bass is most prominent. A very low threshold has been combined with a low ratio and soft-knee setting. The boxed region marked "A" illustrates the attack and release parameters initially being set as fast as possible, which—due to the gain reduction acting within each half-cycle of the low-frequency wavelengths—inevitably results in distortion. "B" demonstrates that both parameters have been adjusted until the distortion is far less evident, while still allowing the gain reduction to clamp down quickly and aggressively on the low band of energy. The release setting could be further lengthened to provide increased low-end density to a mix featuring slower performance subdivisions.

Now analyze the low band at an appropriate level within the context of the other bands. If audibly unnatural gain changes are evident in the lows, or this band still lacks sufficient stability, continue to lengthen the release parameter to correct this. Especially with a production featuring slower subdivisions, a longer release setting can be effective for enhancing low-end density and weight.

Mid Band

This varies with circumstance, especially when mastering a challenging project; however, the mids tend to require less gain reduction than the lows and the highs. This band contains the central "mass" of information we perceive in a production—and to retain a clear, punchy sound, it is vital this energy has sufficient breathing space.

As a starting point, set the threshold in the central range of the band's dynamic movement, above the valleys/below the peaks, and use a moderate 1.5:1–2:1 ratio enabling gain reduction with wide fluctuations. To retain the transient energy contained here, especially in the upper-mids, fast attack times generally need to be avoided. A medium attack setting in the range of 30–40 ms offers a suitable initial guideline.

Similar to the low band, fast release times are also preferable, but importantly without resulting in unnatural gain reduction or distortion, and likewise potentially lengthened in response to slower performance subdivisions. Some mixes require a more heavy-handed mid-band ratio to acquire a smoother result, especially when trying to contain the snare or vocals. However, a heavily compressed midrange is one of the fastest routes to a flat, lifeless guitar sound, and a final product that sounds over-processed. By analyzing the impact of bypassing the mid-band compression, as well as referencing the original mix, an overly congested midrange "mush" can hopefully be avoided.

High Band

Applying significant gain reduction to the high frequencies can enhance the perceived proximity and density of a production. But an excessive application typically results in unnatural modulations in the highs, accentuated hiss and noise artifacts, and an ear-fatiguing final master. There is a broad tendency for the high band to require less gain reduction than applied to the low band, but at least as much, though usually more than that applied to the mid band.

For a relatively transparent application, start with a low-to-medium threshold, and a relatively subtle 1.4:1–2:1 low ratio setting, or for increased high-frequency stability or density, opt for a medium range threshold, with a higher 2.5:1–3.5:1 ratio. A 20–30 ms starting point attack time is appropriate, but shortened if the transient energy in the highs is overly pronounced, or perhaps slightly lengthened to help retain the attack/punch of a bigger snare sound.

The smaller wavelengths in this region often mean that a very fast release time—faster than that of the mid band—can be effective, but this may need to be lengthened to negate any unnatural gain changes, usually most evident in the cymbals. Listening to the frequency band in isolation should highlight when unnatural gain changes are an issue.

Achieving the required tonal balance after an application of multiband compression can be a distinct challenge. Additional compression, once compensated for, results in the relevant frequency band effectively being brought "forward" in the mix—and instruments that extend across both sides of a crossover point can lose their natural focus. It therefore makes sense to try to achieve the desired tonal balance via the make-up gain/output of each band *before* any significant corrective or creative EQ gestures are applied. This is important; the fewer signal processing moves to achieve the desired result, the better, affording fewer artifacts, thereby enhanced sonic quality.

MID/SIDE MULTIBAND COMPRESSION

Mid/side multiband compression further increases the already highlighted risks involved with mid/side compression. This processing tactic may be required when attempting to correct specific problems solely in the sides or center of the mix; for example, attempting to reduce the dynamic range of just the vocal, or de-essing. Importantly, though, mixes that call for mid/side multiband compression are seldom ready for mastering, and a remix should be carried out to address the relevant issues.

Parallel/Upward Compression

The mastering compression approaches discussed so far involve downward compression. In other words, the signal peaks are bought down closer to the valleys, allowing the overall level to be raised in line with the level of reduction provided. An alternative approach is parallel compression, whereby the quieter elements of the dynamic range are effectively raised upward to be closer to the peaks.

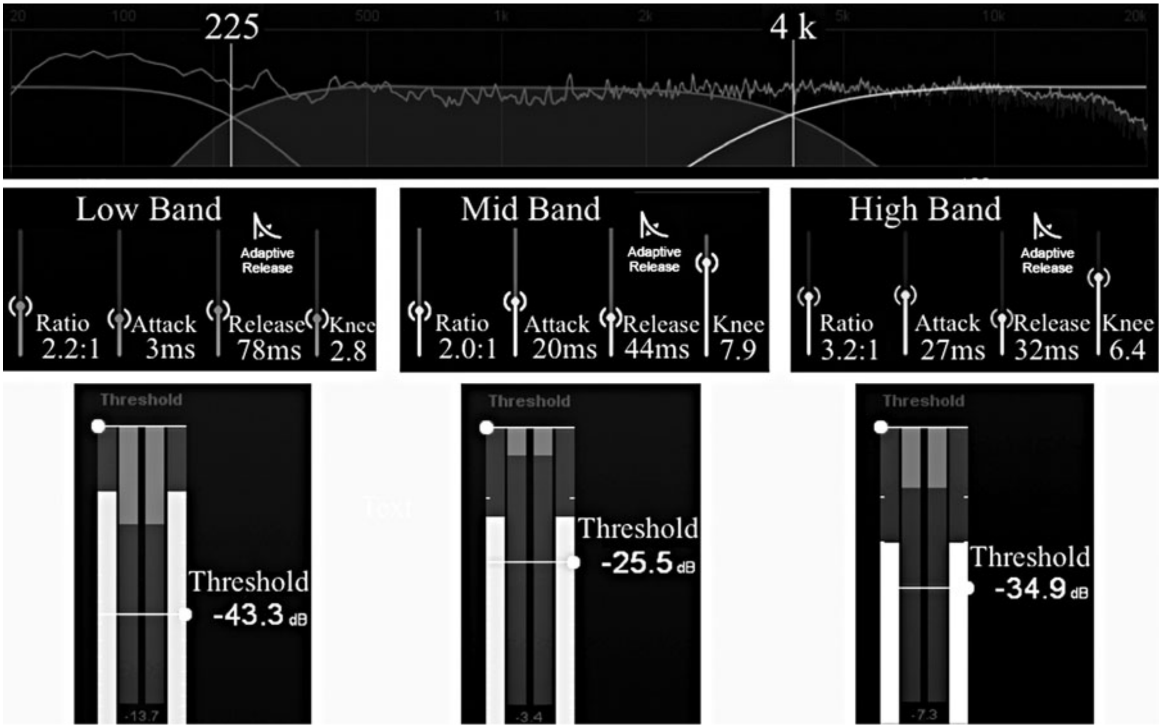

FIGURE 17.7 Figure 17.7 illustrates a multiband compression approach for a problematic mix, where the band in question was unable to provide a remix. The use of just two bands of multiband compression resulted in the midrange frequencies reacting unfavorably to the heavier gain reduction required for the highs—largely due to a "chewed up/murky" guitar sound. A further crossover point to divide the high-mids from the low-highs was thereby required, and positioned at around 4 kHz where the bass had started to roll away, and just below the presence/brightness region of the rhythm guitars.

After setting both the low band's attack and release parameters quite fast, but adjusted to minimize the resulting distortion, an initially moderate 1.5:1 ratio was increased, and threshold decreased until a radical level of gain reduction sufficiently controlled the lows. This was in response to the mix's problematic/overly dynamic low frequencies. A very different approach was required for the mid band, where, to retain clarity and punch, a longer attack setting with comparatively moderate gain reduction was required.

As the high end of the mix lacked aggression and density, a firm approach with a relatively high ratio was needed, yet with less gain reduction than required for the low band. The initial attack setting was slightly lengthened to retain the punch of the snare, and the release setting modified to avoid unnatural cymbal swells. To further help avoid unnatural gain reduction (i.e. pumping) as well as distortion, iZotope Ozone's program-dependent adaptive release setting has been engaged (represented by the icon in each of the bands). This doesn't override the release value set by the user, but is scaled in relation to this.

This can be considered as upward compression. It may sound like the two might provide similar results, but this is far from the case. For reasons that will be explained, the use of parallel mastering compression is better suited to projects featuring slower subdivisions, where the thickness/body of the production needs to be enhanced.

To experiment with parallel mastering compression, first take an aux send from the final mix and route this to a stereo aux input track. Apply compression to this channel using a particularly low threshold (perhaps –40 to –50 dBFS), a modest ratio such as 2:1 or 3:1 collectively enabling roughly 15–20 dB of gain reduction during peaks. The attack time can initially be set to zero, but if this results in the mix's transient energy becoming excessively flattened and dull when the parallel signal is introduced, a slightly lengthened attack time can prove preferable. With a relatively fast release time dialed-in—around 100 ms can be an appropriate starting point—the gain reduction heavily reduces the transient peaks of the aux sent version, thereby emphasizing the quieter dynamic aspects. When introduced in parallel, but at a far lower level than the main stereo mix and with the release time subsequently adjusted to avoid audible pumping artifacts, the "body" of the production is thickened.

Due to the parallel track lifting the quieter elements upward toward the peaks, there is a reduced requirement for the main stereo signal to have its peaks reduced toward its valleys. This means the production's transient energy remains sharper/cleaner than with heavier gain reduction otherwise required. In theory, the introduction of parallel compression therefore provides a more natural, more transparent result than the sole use of downwards compression. In practice, this is not always the case.

The sound qualities of the relevant compressor delivering heavy fast attack gain reduction are fundamental to upward compression being successful. Similarly vital is accurate plug-in delay compensation. It only takes several samples of latency to ruin the phase alignment with the original mix, resulting in comb filtering that "hollows out" the production. Also, due to the parallel compression signal inevitably introducing a level increase, it can be hard to determine whether you are actually enhancing the mix, or simply making it louder. Nevertheless, with the right compressor and with these challenges effectively resolved, parallel compression can be a highly effective tactic for mastering certain projects.

The reason parallel mastering compression tends to be less suited to productions with faster performances is that the aggressive gain reduction accentuates the ambient qualities of the recording and reverb/delay treatment applied in the mix. As there is less space available for these attributes, the production's punch, clarity, and impact can quickly get softened. Additionally, the combination of heavy gain reduction with fast release (set to be empathic with the faster subdivisions) tends to generate audibly detrimental distortion.

"Parallel compression can be a useful approach for bringing more power and level to the sound without making it feel more compressed. But in some cases it can work quite the opposite by adding more artifacts from the actual compression while raising the noise floor, making a mix sound dirtier and less punchy."

Maor Appelbaum

HARMONIC ENHANCEMENT

Whereas additive EQ can only amplify *existing* frequencies, a harmonic enhancer generates *additional* frequencies, which affords significantly different results. Sometimes modeled on analogue tape or tube saturation, harmonic enhancement involves the introduction of subtle harmonic distortion, as well as varying degrees of phase shift and musically related harmonic synthesis of the frequencies present.

When mastering EQ is not providing the required tonal "lift," this style of treatment can be used for adding "presence" qualities and a degree of upper-frequency density. Care is required, though; despite the initial appeal of the enhanced sheen and brilliance, a few further listens can reveal this to be a somewhat "synthetic" brightness, with anything other than a subtle application leading to a brittle and ear-fatiguing production.

STEREO WIDTH ENHANCEMENT

Stereo widening is a relatively simple psychoacoustic effect, whereby the differences between the left and right side channels are increased. The standard way this is achieved is by making a duplicate of the left channel, reversing its polarity, and mixing this into the right channel at a much lower level, and vice versa with the right channel that is mixed into the left channel. This can create the impression of a sound field that is actually outside of/wider than the relevant stereo speakers. Some stereo image enhancers offer the option of taking this a stage further by introducing a small delay (usually within 20 ms) to the polarity-reversed signals.

As with harmonic enhancement, this is a form of signal processing that needs to be handled with extreme care. In addition to the potential introduction of phase anomalies, the enhanced width is often detrimental to the center of the mix, so anything other than a very subtle amount of stereo width enhancement can result in the perception that the middle of the mix has been somewhat "hollowed out." Even when applied in moderation, though, avoid treating the low frequencies; look to the mids or highs, where we perceive directional information far more efficiently.

STEM MASTERING

Rather than involving a conventional final stereo mix, stem mastering involves separately exported mix elements with all relevant treatment in place. With these files imported and summed at unity, the desired final mix is recreated. The kick and bass might be individually or collectively exported, representing the low-end content for the mix center, then the rest of the drums, guitars, and then vocals separately exported, with all files rendered from the exact same start point. As mentioned in the previous chapter, master buss compression cannot be used when exporting stems.

The advantage of stem mastering is that the mastering engineer is able to provide corrections and enhancements to the separate elements, without the potential for this to damage other instruments. For instance, when mastering a stereo mix with vocal sound or vocal level problems, attempts to correct this inevitably impacts other signals in the center of the mix. With stem mastering, the vocal sound can be freely manipulated without such concerns.

Stem mastering can therefore prove valuable when a novice producer feels uncertain about committing to particular mix decisions, and can effectively communicate these doubts, as well as the desired final result to the mastering engineer. However, when professional-standard mixes are involved, the additional control provided to the mastering engineer is unlikely to be of value—and if the producer/band doesn't attend the mastering session, their vision for the project can get misrepresented. Most mastering engineers therefore prefer to work with conventional stereo mixes.

> "To me, stem mastering is kind of entering the realms of mixing, so I personally don't go there. If there is a problem and certain elements in the mix need fixing, I will contact the mixing engineer and ask for a different version with the specific issues addressed."
>
> Maor Appelbaum

SOFT CLIPPING

The maximum peak amplitude/highest digital level within the digital audio environment is 0 dBFS. Attempting to raise the amplitude higher than 0 dBFS results in hard clipping, whereby all the bits in the digital word have been consumed, meaning this information cannot be represented. Hard clipping leads to the top segment of an audio waveform being squared off, generating a hard angle where the waveform meets the clipping point. This is highly destructive to the audio information exceeding the limit, resulting in unnatural and harsh high-frequency distortion.

Soft clipping minimizes these unwanted, unmusical qualities by gently smoothing the transition between the waveform's unclipped and clipped sections, with a gradual rounded edge. This enables the transients to remain far more intact, while enabling a reduced reliance on traditional hard limiting to provide the final master's inherent loudness.

> "Limiting and multiband compression are the two processing approaches that people tend to overuse when it comes to mastering, as both can really suck the life and punch out of a mix. I use A/D clipping instead—that's the secret for achieving good volume without destroying the transients. I'll hit a premium A/D convertor really hard on the return. When you go too loud it will distort, and then you reduce to a level where it doesn't distort, and you will get it much louder, and with less impact on the sound than any limiter will be able to provide."
>
> Jens Bogren

FIGURE 17.8

There are various soft clipping peak limiter software plug-ins now available, for example Stillwell's "Event Horizon" (see Figure 17.8), which additionally offers look-ahead limiting. However, many mastering engineers and producers, including Jens Bogren, prefer to convert the stereo mix from digital to analogue, then clip the returned inputs of an analogue/digital convertor. This is widely seen as providing more musical, cleaner, and more transparent clipping than provided by the software route. A-D convertors well suited to this task include the (D-A and A-D) B2 Burl Bomber, and the Lavry Gold AD122-96 MkIII Mastering AD Convertor.

LIMITING

Although an unmastered final mix and a mastered version could both peak at 0 dBFS, the latter is invariably a lot louder. This is largely as a result of the average (RMS) amplitude of the mastered version being closer to the peak 0 dBFS level. Limiters are not intended to act on average (RMS) levels; they are designed to engage with peak/transient content. Nevertheless, when a limiter attenuates a mixed transient energy by, for example, 3 dB, the non-transient (RMS) elements are effectively raised by 3 dB when the audio's output level is restored, thereby increasing average apparent loudness.

Less experienced engineers often exaggerate the role of limiting/loudness maximization when mastering metal music. When overused, the fixed high ratio/fast attack takes sharp, punchy transient energy, and flattens it into a comparatively blunt/lifeless production, often with unmusical distortion in the upper-mids. The best approach to loudness is therefore a multifaceted, holistic one, whereby no single processing instance during mixing or mastering is provided with this responsibility. In simple terms, gradual accumulative dynamic control—which subsequently requires only relatively mild limiting in order to achieve competitive final inherent loudness—is likely to result in a higher production standard than the use of heavy limiting. Although relatively mild limiting partly defines the overall loudness of a finished master, it changes the mix's energy, aggression, and frequency response in a negligible way.

This concept is important; limiters provide the user with few controllable parameters and hence minimum control over the final outcome. With most designs, it is simply a case of pushing the input level/lowering the threshold to raise the loudness of the source against the "ceiling" (the limiter's highest maximum output level) while avoiding perceptibly obvious artifacts such as pumping and distortion (which sometimes requires a lengthened release parameter).

> Limiting is seldom, if ever, a successful replacement for effective compression use.

All the same: different limiters react to and process audio in different ways, and increasingly so with higher-gain reduction levels. And whether a limiter remains transparent, or initiates coloration and audible distortion can also depend on the qualities of the relevant mix. It is therefore valuable to compare the limiter models you have access to when loudness matched and providing the same levels of gain reduction. As an overarching principle, the processing should have as transparent an influence as possible on the mix's transient energy.

It is therefore preferable, but not always possible, to keep the maximum amount of limiter gain reduction to within 3–4 dB at any point. In the less fortunate event that more significant limiting is required, it is worth assessing how your more favorable units combine when splitting the gain reduction load between them. Two separate limiters, each providing, for example, 4 dB of gain reduction, can enable a more natural result than a single 8 dB limiting stage. An alternative approach that sometimes provides better-quality results than single or dual limiting is applying an additional stage of fast attack compression just prior to the final limiter.

FIGURE 17.9

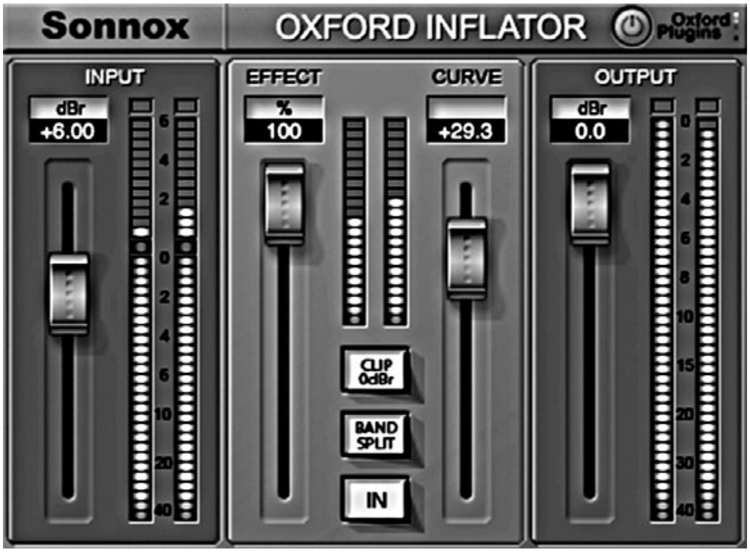

FIGURE 17.10

FIGURE 17.11

FIGURE 17.12

AUTOMATION

Although novice engineers often process an entire project through a single array of EQ, compression, and limiting parameters, the foundation of successful mastering is responding to the separate challenges of each individual track. This tends to require individualized parameters for certain songs, and even distinct parameters for different sections within the same song. Also take into account that with combined compression and limiting, the dynamic variations between quieter and louder song sections can easily be lost, merging into the same apparent level. Compression threshold or level automation should be used to retain/restore these dynamic variations. As each subsequent track is being worked on, take into account the fundamental mastering task of delivering a "cohesive" project, by comparing each track to the previously completed master(s) and making adjustments accordingly.

FADES

Brief fades at the start or end of a track can be inconsequential to the mastering processing applied across the fade region. However, with greater fade lengths, the audio suddenly and audibly overshoots the various processing thresholds, resulting in drastic, unnatural level changes. To avoid these issues, export the mastered version without the required fades, and then apply the fades to the imported file.

MASTERED OUTPUT PEAK LEVELS—CD

Final mastered peak levels of 0 dBFS exceed the reproduction capabilities of certain CD players, as the cheaper D/A conversion within these units can result in the analogue output being slightly higher than 0 dBFS. It is therefore preferable to limit your final mastered peak levels to anywhere between –0.5 and –0.1 dBFS, with –0.3 dBFS being a favorable option. However, when loudly mastered WAVs are converted to MP3, this can—depending on the encoder—result in clips and "overs." It therefore tends to be preferable to carry out MP3 conversions from a WAV mastered to –0.5 dBFS.

Having closed this chapter discussing mastered output peak levels for CD, the next and final chapter looks at loudness normalization technology and metering. This involves very different inherent loudness levels.

FIGURES 17.9–17.12 (*facing page*) Often most effective for mastering metal music are limiters designed to enhance perceived loudness in a more rounded way, by minimizing the detrimental transient reduction effects of conventional limiting. Figure 17.9 shows the Sonnox Inflator, Figure 17.10 Slate Digital's FG-Level, Figure 17.11 the Universal Audio Precision Maximizer, and Figure 17.12 the Waves L3-16 multiband peak limiter. These models feature very different detection algorithms, meaning that each can prove suited to different source material.

10 Mastering success

A ROUNDTABLE DISCUSSION WITH EARLY TO MID-CAREER MASTERING ENGINEERS

Russ Hepworth-Sawyer

INTRODUCTION

The predominate interest for many readers of music production in the past 20 years or so has, rightly, been those interviews with engineers and producers who hands have touched the microphones and consoles that managed the signals that recorded the classics. We're still coming to grips with the phenomenon that was the boom in popular music in the latter part of the twentieth century. Academically speaking, within the music production field, we're still processing!

To read a Ken Scott interview, is now quite commonplace, such as ours in 2012 for the *Journal of the Art of Record Production* [1]. Since Howard Massey's success in raising considerable interest in the views of engineers and producers, the likes of Ken Scott and several other seminal colleagues, too numerous to list here, have been interviewed frequently and deeply, providing the researcher with so much material to discuss and interpret.

During the preparations for Producing Music, another book in the Perspectives on Music Production series (POMP), I considered the voices so far neglected in the discourse from producers in a 'Behind The Glass' style. There are few interviews with Ken Scott and his peers at the time they were doing their seminal work, or even before their first crowning achievements. I considered what it might be like to be able to research an engineer's view perhaps before they'd done their most popular and defining work. In Producing Music, I interviewed three colleagues and friends that are what I considered 'mid career'. This can be read with interest. Since that chapter, British engineer and producer Adrian Breakspear, now working out of Sydney, Australia, for example, has won more awards and done more defining work. Equally, Mike Cave has since worked on the hugely successful album launching Lewis Capaldi's career. Their interview contributions to that chapter, which stand before further, recent successes, will stand as a useful resource for future researchers looking into their work at a defining period.

Once again, in preparations for this POMP series book, Mastering In Music, I felt it important to continue the theme. This time looking at early to mid-career engineers to capture their views on their education.

THE QUESTIONS

In the preparation for our roundtable discussion, held mainly over email, but also direct Skype interview for some with a later transcription, I prepared nine questions relating to their start in audio mastering, their training journey and views on current situations within the field. This is a current field of interest for me presently and it is interesting to see the views and consideration of the engineers included here who have not received a 'standard' apprenticeship route in audio mastering.

The questions are listed below.

1. How did you get into audio, and was mastering your first work in the area?
2. If not, how did you come to know mastering existed and how did you fall into it?
3. Would you describe yourself as self-taught, or has some form of 'teaching/mentoring/apprenticeship' been at the heart of some of your training?
4. What are you views of the routes into mastering?
5. If you're self-taught, how have you trained yourself. Has it been organic, figuring stuff out, or Internet or other methods I've not listed?
6. What do you think you've missed out on within a apprenticeship or higher education context? One example might be the passing down of vinyl cutting for example.
7. What do you think are the advantages of being self-taught, if there are any?
8. If you were asked to write a higher education course based on your experience, what might you include?
9. What do you see as the future of audio mastering and how you might advise those coming up in the field?

Not all questions were answered sequentially and, as perhaps a poor interviewer, conversations naturally ventured elsewhere. I have tried to edit the work back to its original path where necessary.

PARTICIPANTS

The participants are all known to the author personally and have been chosen due to their early to mid-career stage in their careers.

Nick Cooke attended Leeds College of Music and from there toured both as a musician and soon landed a job with a production music company in their mastering department. Nick is fascinating as his experience was being dropped in the field without preparation. He's since gone on to set up his own mastering business and exceptionally renowned White Mark designed studio.

Katie Tavini, known for her contributions to Audio Media Europe and as a keen supporter of the AES UK Mastering Group that is co chaired by JP Braddock and the author. Katie's business is growing exponentially and it is wonderful to capture her thinking at this pivotal growth in her career.

Jay Hodgson started life as a performer in several bands in his native Canada. Later in life he became one of the first academics to study popular music production and be awarded the Governor General's Award for his contribution. Jay came to mastering from a purely creative angle and has enjoyed several successes and Juno nominations for his work internationally.

Jeremy Graham started working in the live audio industry in his native New Zealand. Later he came to the UK to study audio mastering and became a valued assistant in MOTTOsound with the author. Successfully Graham now runs his own mastering business, Transfer Lounge AMS and continues to gain success as a music producer. Jeremy's insight here is exceptionally valuable, as we predict his career to be one to watch.

QUESTIONS

How did you get into audio, and was mastering your first work in the field?

Nick Cooke – Prior to my degree I was already interested in audio, as I have been playing in bands since I was 13 and I continue to work as a musician. My parents and grandparents had a huge influence on me from a young age with their interest in music and Hi-fi. My dad also had a PA system for his band, so I started off by doing live sound for them and other local events. I was lucky to go to a school with a strong music department and I was involved in a range of audio related activities – both in and outside school – which naturally led to me doing music technology at A-level and then to continue on to higher education. Throughout and after completing my Music Production degree, I continued to gain work experience in different areas in the audio world.

Mastering was definitely not my first work in audio. I initially wanted to work in foley for TV and film and managed to gain some time at both Aardman Animations and the BBC in Bristol as well as North One TV and Nickelodeon in London. I also assisted on various location recordings for a Classical music label to gain as much experience as possible, whilst working in my local pub to fund it all. Eventually this all led to paid work in Outside Broadcasting for televised tennis, horse racing, and football. This covered rigging, maintenance, comms, and simple mixing tasks. I also continued to obtain work as a session musician, both live and recording, as well as some live sound work at a few small festivals around the country.

Katie Tavini – I got into audio because I was super into music, played the violin since I was a kid, and was fascinated by computer music. I used to borrow books from the library on midi sequencing and super nerd out. I got a copy of Cubasis when I was about 16 and started making MIDI music. I studied both music and music technology in college but didn't really know what career options there were. A career officer told me the only job in music was to become a performer but I knew that I never wanted to do that. So I applied for an acoustics based course at university, but then changed to a BA Music course when I realised I was the only girl on the course. There was a tiny 'recording studies' module which I loved, and so I spent all my time in the studio – I used to buy my friends studio 'credits' off them so I could have longer in the studio. One day my tutor asked if I'd like a job working in a studio which was ace. I really wanted to be a studio engineer at that point. The job was amazing and the producer I was working for taught me so much. Eventually I went freelance as a recording engineer and started doing a bit of production work too. It wasn't really my thing because I was expected to mix, and hated it. Too many choices.

Jay Hodgson – I got into audio by participating in recording sessions as far back as 1992 or 1993. I was lucky enough that the first recording session I ever participated in was for John McDermott's platinum selling 'Love Is A Voyage' album, so I got to see an elite operation for my very first session. McDermott was somewhat mean to me, though. He would often introduce the band, at live shows, and tell everyone that I was only there because my father was a very expensive lawyer. So needless to say, I didn't stick around for very long. I soon travelled down to Boston, and worked as a session guitarist a fair bit while attending Berklee. I also wrote my own songs and had a band. Slowly I became increasingly interested in how some producers were able to get particular sounds and translate my songs into particular arrangements that I found super compelling. Once I put out my first record, in 1997 or so, I was completely hooked on recording, and utterly bored with performing and touring. But it wasn't until 2002 or so, when I bought a laptop with Garageband on it, that I got really interested in audio. I didn't start mastering in earnest until a few years after that.

Jeremy Graham – I started working professionally in the music industry towards the end of 2007 where I began a career as a live sound engineer, initially starting out as part of an internship programme during my bachelors degree. Although my strengths leaned more towards music production, there weren't too many opportunities to be involved in the recording/mixing side of things, and I didn't view myself as a studio mix engineer at the time.

Live sound and working in the events side of the industry became a real solid foundation that would provide financial security over the following decade and really trained my ears to be able to listen critically, both sonically as well as socially, in terms of communicating and working alongside a diverse range of people, events and a variety of spaces.

If not, how did you come to know mastering existed and how did you fall into it?

Nick Cooke – My first mastering job was at a small family run music production library, De Wolfe Music, that have been going since 1909 when they produced music scores for silent film before sound recording was available. I didn't know I was applying to be a mastering engineer: the role had been advertised as 'Music Editor' and my interview went so well chatting about mainly Glastonbury Festival that I forgot to ask about the job!

On my first day I was put in a small room with a tape machine, a pile of tapes, a couple of Weiss units and a SADiE system. I was asked to compile a CD out of a few CDRs from various composers and to make any slight EQ adjustments. I later realised this was mastering. Although I had learnt a bit about mastering at university, and had the opportunity to meet mastering engineer and audio education guru Bob Katz at a University conference, I hadn't really thought about mastering until accidentally getting that job.

De Wolfe have a vast and varied back catalogue on tape and a big part of my job there was remastering past vinyl releases from the original master tapes. This was an amazing opportunity to hear the music from a time when budgets were big and large studios still existed. The audio and production quality was eye-opening. I really enjoyed this work and started to consider mastering as a career.

Katie Tavini – Someone on Gearslutz (I think) told me that if I wanted to improve my mixing I should learn how to master. So I started stealing tracks from studios, trying to 'master' them on crappy speakers, and then when the pro masters came back, I used to compare and try and work out what the mastering engineer had done. I didn't know anything about the technical side or formats, I just enjoyed having a go.

Jeremy Graham – I was exposed to post-production and mastering during the final year of my bachelors degree. Although I excelled at the post-production side of things (in terms of syncing audio & video), Mastering was honestly still kind of a mystery to me.

It wasn't until a few years after I completed my honours degree, alongside continuing to produce music that I slowly began to understand what mastering was all about sonically, and out of necessity more than anything else (in terms of releasing music and getting tracks ready to perform live). I decided to move to the UK at the start of 2017 and consciously stepped away from live sound, which is when I really started to hone in and understand the full breadth and craft of audio mastering from a more professional context; building on the framework I had already developed.

Would you describe yourself as self-taught, or has some form of 'teaching/mentoring/apprenticeship' been at the heart of some of your training?

Nick Cooke – I haven't had direct training in mastering like an apprenticeship or a higher education course focussed on mastering, so I could be described as essentially self-taught. However, in learning on the job, I have always been picking up bits from engineers, both mastering and other types of audio engineer, and other people around me. You could also say that I have had indirect mentoring through keeping bosses and clients happy. They would tell me what they wanted changing and I would work out how do it. They feel like mentors to me.

When first starting at De Wolfe I was shown the basics of SADiE and how to use a tape machine, and then left to my own devices. Luckily, as most of the time I was working with amazing sounding tape recordings, I was a bit scared to alter things too much and was encouraged not to. I quickly learnt that a little can go a long way. Later moving to Extreme Music at Sony/ATV I was working with of all sorts of producers/engineers of various different calibres, some just starting out in their careers and some iconic big names in the business. These were often all on the same album that I needed to make sound coherent as well as creating a specific sound to meet a given brief. Working with producers, engineers and artists with very clear ideas of the way they want their material to sound was a quick learning curve and showed me a very different side to mastering. I gained a lot of experience from working to briefs and meeting their goals.

I think I was lucky to have these experiences of such different styles of mastering so early on in my career. Throughout all of this, however, it has always been really important to me to read a variety of audio and music related books not just aimed at mastering, and of course a huge amount of listening to masses of music and recordings.

Katie Tavini – I'm totally self taught and still learning!

Jay Hodgson – I am completely self-taught. I mastered for friends and family for years, and figured everything out by myself – Totally self-taught.

Jeremy Graham – I'd consider myself an observant person, willing to put in the time to research varied opinions and processes in order to draw my own conclusions, so yes, fundamentally self taught.

Having said that, I've been fortunate enough to have had some enlightening and extremely educational conversations with some well respected mastering engineers in the industry very early on in my career, and that advice, along with my own investigations, trials, errors, and overall persistence, has really paid off more than I could have hoped for.

It's probably not the most traditional route, but I'm not convinced there really is one.

What are you views of the perceived routes to becoming a mastering engineer?

Nick Cooke – It is hard for me to comment on apprenticeships as I have technically not been through one but I certainly believe they are valuable. Probably the biggest challenge with that route is finding an opportunity in the first place. It seems to be the traditional way to move into mastering as a career, and may well be a more straight forward route into it. Certainly assisting and observing a notably great engineer is a wonderful opportunity that I would have jumped at the chance if one came along. My route into mastering however, feels like I have snuck in through the fire exit: it couldn't really have been planned and was in a large part down to chance. And even after 12 years it still feels like I'm only just getting started.

I am pleased to have gone through higher education and although I wasn't focussing on mastering at the time it certainly helped open my mind to possibilities and cemented the idea that I wanted to work in audio. I personally feel that it is better to keep a broader focus at degree level, rather than specialising too soon, then you make the most of whatever opportunities arise. Plus I don't see mastering as a 'black art': it is part of music production as a whole, perhaps one from a different perspective, but it is necessary to understand all the parts to know how mastering fits in.

In my opinion building a career in mastering requires a lot of self-teaching even with apprenticeships or higher education. I would also be bold enough to say that a mastering engineer cannot be fully self-taught. It relies on working with and learning from other people throughout the whole career. That is certainly what I have found so far anyway. Rather than three distinct routes, I would argue they can often intertwine.

Katie Tavini – Self taught is cool because it's the only way I know, and it's enabled me to develop a very clear style and tastes. It's also given me the freedom to work on the types of music I enjoy the most. Apprenticeship I guess gets your sense of the actual business side of things good fast because you're watching someone deal with stuff on a daily basis, but I would struggle to sit there and shut up when someone else is mastering. I can't learn that way. Higher education seems like a great way of learning the technical skills, but you won't really learn the business side of things and honestly I'm really not an academic person so it's not an option I would personally consider.

Jay Hodgson – I can't speak to anything other than being self-taught. I imagine it is a tonne of extra work, though the results are very personalised. I would have liked an apprenticeship, and even higher education, but unfortunately I couldn't afford either, as I worked my way through to a PhD in music which left little time for anything else professionally speaking. Being self-taught does allow you to have a certain confidence in your own tastes and artistry, I think, that other routes perhaps wouldn't emphasise. I truly don't believe in a right or wrong, or a good or bad, beyond client approval, when it comes to mastering. When I hear people slagging other people's work I just roll my eyes, and move on. We're all just getting client approval how we see fit. It's that simple. Anything more is just reassuring ourselves that our job matters. It's so loud its distorting? Cool. That sounds good sometimes. Any more words on the matter, beyond 'the client approved', is just pretentious, insecure bullshit to my mind.

Jeremy Graham – I suppose I'm biased in that I really enjoyed and valued the five years of full time music study I completed straight out of high school, but I'm not sure what else I would have done; it was the only logical option given the nature of opportunities that existed for me at the time, or that I was aware of.

Tertiary study is definitely not for everyone (and I suppose neither is an apprenticeship), but regardless of whichever path you might choose, only you know the right path for you, and whatever that path is, there is no evading the abundance of hours that are required to figure out and develop your skills and overall craft.

If you're self-taught, how have you trained yourself? Has it been organic figuring stuff out, or internet or other methods I've not listed?

Nick Cooke – In terms of self-training, it's been mainly listening, over and over, and trial and error. I have read a lot of books on mastering, which has been essential to learning the technical bits, but also wider reading around audio in general, as well as how to run a business etc. I have learnt bits at university, conferences, networking events, YouTube, podcasts and internet forums. I also learn from reading magazines and equipment manuals, even old ones from the 1950s, 1960s, and 1970s, I have a brilliant set of manuals from Neumann and Scully cutting lathes for example. But where I've learnt most is through talking to and collaborating with people. Plus having the opportunity to work in an acoustically accurate room from pretty much the beginning of my career has definitely made a huge difference to my understanding of mastering processes.

Being a musician and knowing the music business from other angles has been key for me for working with artists and producers and understanding their needs and processes. This has really helped me with building relationships with clients and having clear communication in both directions. I know that some of the top engineers have also come from a musical background so it obviously helps.

Katie Tavini – Figuring stuff out, having a go, comparing work to pros work (like I said before, I used to nick tracks and try and master them, and then compare them with professional masters that came back). My background in classical music gave me really good critical listening skills I think.

Jay Hodgson – I trained myself by working. I just worked, worked, worked. I must have cleared 1,000 masters before I had any sense of certainty about what I was doing. I read a book to figure out what meter numbers to peg records to so what I produced would sound roughly the same on my monitors as they did once distributed.

Jeremy Graham – In terms of my own mastering development, I'd say it's been a fairly organic process.

It's really been a matter of first figuring out the specific tools I wanted to work with, knowing their strengths and weaknesses (including my room/monitors), picking up bits of advice here and there (whether directly from other engineers or via the Internet, books, interviews, etc.), and then developing my own approach to problem solving issues/problems (where they exist) and tuning in my ears/mind-set somewhere between the expectations of the client and my own intuition based on my personal tastes and references.

Aside from that, I'd say trying to remain consistently self-critical over the long term has been valuable for me personally, but not being too hard on myself either; understanding my own shortcomings, expectations, and seeking to improve and refine my abilities.

What do you think you've missed out on within a apprenticeship or higher education context? One example might be the passing down of vinyl cutting for example

Nick Cooke – I wouldn't say missing out as I wasn't aiming to be a mastering engineer, but aspects of an apprenticeship such as on-the-job guidance and reassurance of my work is something that I have had to manage without. The main things I feel I missed out on in my degree was working within a professional studio and a work-placement scheme within the degree. I am aware of some courses that do offer these, which I would say is definitely beneficial. As work placement opportunities were not available on the course I was on, I had to find my own way to build experience.

Yes, vinyl cutting. I would love to learn to cut vinyl, it's still a big dream of mine but unless you are already working in a facility with cutting engineers it is a very daunting and expensive skill to obtain, although it is not impossible. Saying that I have learnt a lot from various conversations with cutting engineers either casually or as part of a project. And there is still time!

Katie Tavini – Cutting vinyl is the main one, but no one's ever complained that I don't cut. I just give them the details of some good cutting engineers I know. But I had a good background in analogue formats from the first studio job I worked at.

Jay Hodgson – Well I've mastered a lot for vinyl, but I don't cut. The whole 'cutting vinyl' thing is a stodgy last ditch effort to gatekeep, IMHO, anyway. If you master for vinyl, you're usually preparing masters for cutting now. And that's fine. It's kind of like complaining about how kids nowadays have always had a screen to stare at while working, so everything sounds too much 'on the grid'. So what? Time moves on. I don't feel like I missed out on anything, really.

Jeremy Graham – If I were starting all over again from scratch, I'd definitely consider going back into higher education, but it's certainly not the only route to take, and as mentioned earlier, it's such an individual thing, down to your own determination, available opportunities, and what you're willing to sacrifice in order to follow that specific path.

Aside from the obvious musical/technical knowledge I gained during my years studying, I would say that I also had instilled in me a sort of lifelong camaraderie for those involved in the music industry, right from the get go. I don't believe this has anything to do with something you're necessarily taught per se, but more to do with an attitude that develops over time while you're immersed in that kind of environment. I'd like to think the same applies in the traditional apprenticeship route but I'm not really in a position to speculate in that area, as it just wasn't the path I went down.

From an individual standpoint, I've consistently been put in positions throughout my life where others around me are either older, wiser, and more experienced than myself, and it's forced me to raise my own bar in order to keep up so to speak, which I think has a lot to do with the route I went down, but that's just me, and I don't feel like I've missed out on anything really, it's just a different path.

What do you think are the advantages of being self-taught, if there are any?

Nick Cooke – Possibly not inheriting bad habits, although of course it's absolutely just as likely to develop your own. Higher education offers a good starting point, particularly in discovering and learning about the basics of the industry, but if someone knows they absolutely want to do mastering, perhaps getting stuck in is more efficient than completing a degree course first. As far as I'm aware, the basic aspects of mastering haven't changed much so can be self-taught through the many mediums available. Hopefully the continued exploration, experimentation and creative side of mastering will also develop naturally though engineers that are self-taught.

Katie Tavini – Freedom to experiment, make mistakes, get a good solid network of people who I can trust to skill share with, the ability to work on the types of music I'm most interested in (particularly the DIY scene and classical music, I'm not really interested in big pop artists for example).

Jay Hodgson – It is 100% your creative art, then. You were led to it, or drawn to the craft, and you have created a process from a void.

Thus, your work is entirely creative. You are not beholden to notions of right or wrong, or good sounding vs. bad sounding. That means you can just do the work. That's an advantage, to my mind. But then, I wouldn't know the obverse. So who can say?

Jeremy Graham – There are certainly pros and cons with either route. If you're able to critically assess yourself alongside the information you absorb, I think being self-taught can be a real advantage. However just like higher education, it's not a path for everyone. We each learn and pick up skills differently so it really depends on the type of approach you want to utilise along with your own learning style.

There's always value in learning first hand from others while you're in the same room, where you might have those light bulb moments far earlier on than you might have otherwise, and that can apply to both higher education and apprenticeship routes, but that also doesn't necessarily mean you'll gain more or less, it depends on the environment, your own situation and the attitudes of the person/people you're learning from or with.

Either way, you have to be able to communicate effectively with your peers, colleagues, and ultimately your clients; the right attitude has a big part to play in my opinion and experience.

If you were asked to write a higher education course in mastering what would it include?

Nick Cooke – Conferences and meeting mastering engineers would be my top priority. Plus chances to really listen to music in a proper mastering facility. I don't think teaching specific methods are that important, as mentioned earlier, these can be pretty much self-taught. More focus should be on the overall goals of mastering and creating differences in feel. Therefore, providing space to experiment with different methods would be important.

The biggest skills required I think are people skills. Mastering is often not just doing what you think sounds best but making something how someone else wants it to be, which can often be difficult to pinpoint. My main advice would be to not to over complicate your approach. I find simple is usually best, and its ok to go back to the beginning and start again.

Katie Tavini – Focus on critical listening (musical dictation although not enjoyable has really helped my work, as it's enabled me to pick up on areas which can be improved in the mix super quickly). Ditto for music theory – it's great to be able to talk to clients in their language and makes communication so much quicker and easier. The importance of metadata! Developing your own tastes and learning how to enjoy music you wouldn't normally listen to (because I don't know any mastering engineers who only work in one genre). Business skills – good communication, being nice to people, tax returns, how to find work. Understanding formats such as disc description protocol (DDP) and MFiT, now Apple Digital Masters and so on.

Jay Hodgson – I've actually done this. I teach them my process, from track arriving to the final emailed goodbyes to the client at the end of the process. I take them through what the goals are, and how I achieve those goals. And then I really push them to develop their own processes. In this way, I create the same conditions I learned under, but in a controlled 'safe' way. I honestly don't believe there's any other way to learn this craft.

Jeremy Graham – If I really had to, and in a very broad sense, I'd likely focus the majority of the course time on ear training, whether that was musically/sonically, micro/macro, attentive/passive, space/depth, height/width, etc. in order to help engrain more readily usable listening habits earlier on, despite the intended role or application; I think it would benefit all involved across the music industry to have a fundamental foundation of critical listening skills.

I'd also factor in some kind of critical thinking element as well (specifically tailored to the music industry), as this can really open up your mind to other possibilities and allow yourself to have more unbiased opinions, especially while researching new topics or information; it was certainly valuable for me during my studies and something I still use everyday.

What do you see as the future of audio mastering and how you might advise those coming up in the field?

Nick Cooke – I think there is a big future for mastering although it is constantly evolving as it always has done. I'd advise people to keep up with the different formats and industries developing: we already have streaming, gaming, apps, film, and TV as well as the standard music business and the syncing and production music areas are stronger than ever. There will always be new formats and mastering applications arising.

Advice that I have been given that I would pass on is to not undersell your services. As an engineer, look at how you fit in with other engineers, keep prices reasonable but don't undercut people or you'll be undercutting the industry as a whole. We need to keep quality high and value that quality.

Katie Tavini – Over the past year I've seen more artists and mix engineers wanting detailed feedback in order to perfect a mix. I've not really mastered long enough to see trends come and go, but it really seems like clients are valuing mastering engineers as a fresh set of ears this year rather than 'just making stuff louder'. I've also had to learn as much about the rest of the music industry as possible too, as clients often ask me which distributor they think would be the best for them, can I put them in touch with any music blogs, what different types of royalties can they claim etc. So even though I'm just doing mastering, I get asked a lot of other stuff too which is becoming more of a thing lately.

Jay Hodgson – AI-based algorithms doing the transferring work, and the rest of us doing the creative work. I recommend being as much of an artist as possible, and shaping what you receive to suit your ears. Everything else is being done by machines now. And will continue to be for cheaper and cheaper… my creativity is the only aspect of my mastering that can't be automated. So it's the only thing I really care to develop and invest in. Of course, that means I'll lose some clients. But they didn't want my mastering anyway; they just wanted their work to be mastered. Big difference.

Jeremy Graham – As technology improves for everyone who's involved with the creation or process of making music, I'd like to think more emphasis would be focussed on the core of the song or production i.e. how to best translate the intended feeling or emotion across to the audience, whether that's at the recording, mixing, production, or final mastering stages, even how the visual elements are presented from cover art, through to artist presentation. Although this concept is far from new, I feel like technology is becoming less of a focus and more of a tool or foundation to extend expression, however one may choose to use it in order to do so, musically or otherwise.

In terms of mastering specifically, and from what I can infer from at least my own experience, is that the right mastering engineer can offer a lot more than just the final creative decisions and format deliverables for a project, and can be a real asset to the artist and their projects over the long term. As budgets appear to becoming smaller (or perhaps just consciously tighter), I personally think it's more vital than ever before to retain real lasting connections throughout your immediate team and become invested with a mastering engineer who can work alongside you as a source for not only technical knowledge, but who can also assist the artist in collating and expressing their vision from a trusted and committed outside perspective.

Some advice I try to remind myself of from time to time is as follows; listen to and develop your intuition, practice patience with yourself and others, be prepared to play the long game (not just the innings), take your time (but hurry), and you're only as good as your last gig. Finally, talk with as many professionals and peers in the industry that you can; those interactions are invaluable, however fleeting they may appear at the time.

CONCLUSION

The pool of early to mid-career mastering engineers selected here have all launched their careers in a somewhat self-taught paradigm. This is not to promote this method, or route to professional mastering is by any means the norm, but there does seem, to the author in his research, to be more acceptance in this route. As many of the interviewers note, they do not cut to vinyl, which is often seen as Jay Hodgson notes as the 'gatekeeper' to the dark art of the field. To many in other spheres or generations would uphold this mantle. To others this is a fairly insignificant feature of the tonal adaptation and improvement that occurs before the cut is made, or the ‘bounce’ is done.

Had time permitted, further questions beckon around what each engineer has learnt and how it has been achieved. There have been discussions of ‘trial and error’ but also Jeremy Graham being brutally realistic about being honest at the same time as not being too hard on oneself. These appear to lead to further discussion at a later point around the soft-skills and personnel factors in audio mastering, at a time when non-attended sessions are most definitely the norm.

Of particular note is Jay Hodgson’s striking comment that he believes mastering engineers are potentially all too critical, and that the approach of ‘what the client approves is what the client gets’ is his rule no. 1. This suggests a question for further research that is ‘when do we know when the mastering is done’? There also seems to be a sophisticated psychological dynamic at play here, which is sadly not addressed in these interviews, but warrants further research.

A concurrent theme through the interviews is the clear professionalism demonstrated. Each interviewee highlights the importance of being able to listen and to adjust music for the improvement that can be offered by a positive experience in mastering.

As a snapshot of early to mid-career mastering engineers in the new decade of the 2020s, the aim of this interview has been to provide interesting insights into their training and their opinions – just at the point when audio mastering may change its identity forever.

REFERENCE

1. Golding, C. R. Rosen, D. Hodgson, J. Hepworth-Sawyer R. 2012 *Interview With Ken Scott. Journal of The Art of Record Production*, Issue 7.